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30-Channel p.c.m

Summary

Thirty-channel p.c.m to CCITT recommendations offers a significant improvement in transmission performance over the now familiar and well-described 24-channel systems, particularly for operation in trunk networks.

The following article briefly defines the specification changes leading to the improved transmission performance. It then describes the units and circuit functions of a complete p.c.m system with its multiplexing, signalling, line and ancillary equipment.

Finally, the British Post Office plans for the network are briefly described in order to show the make-up and development of the proposed digital hierarchy from the first service using 30-channel systems.

Introduction

For nearly a decade now the British Post Office has been buying 24-channel p.c.m equipment for installation in the junction network and, to a limited extent, in the trunk network as well.

By the end of 1975 the value of the Post Office purchases of this generation of equipment and its ancillaries approached £60m, of which, Marconi Communication Systems have supplied over one-third.

The years 1977/8 should see the introduction of the second generation of Marconi equipment into the UK network, its 30-channel capability giving a basic building-block having a capacity 2.5 times greater than the present 12-channel group used in the frequency division multiplex (f.d.m) network. It is in accord with the current CCITT recommendations, and meets the requirements of the relevant BPO specifications.

All items closely follow established practices for 24-channel systems in terms of equipment engineering, though the circuit design uses more sophisticated techniques, as will be shown later on in the article.

Improved Specification

The early 24-channel systems were designed primarily to service the needs of telephone traffic expansion in heavily congested city and urban areas. This immediately placed a limitation on the number of systems likely to be connected in tandem, being typically a maximum of four, i.e, a particular speech circuit could be encoded and decoded up to four times. As a result, acceptable speech quality could be realized with 7-bit coding for the speech samples, leaving an eighth bit available for channel signalling and synchronization information interleaved over a four-frame multiframe.

In the 30-channel system, signalling and synchronization information is concentrated into two separate time slots, making a 32 time-slot frame. This, coupled with a 16-frame multiframe, gives a much greater signalling capability than in 24-channel. This arrangement also enables 8-bit coding to be used for speech samples, so providing improved definition and signal/quantizing noise ratio. Coupled with this, and important for tandem connection of more than four systems, the gain/level linearity is now tightly specified for inputs as low as −60dBm0 as compared with −30dBm0 for 24-channel systems.

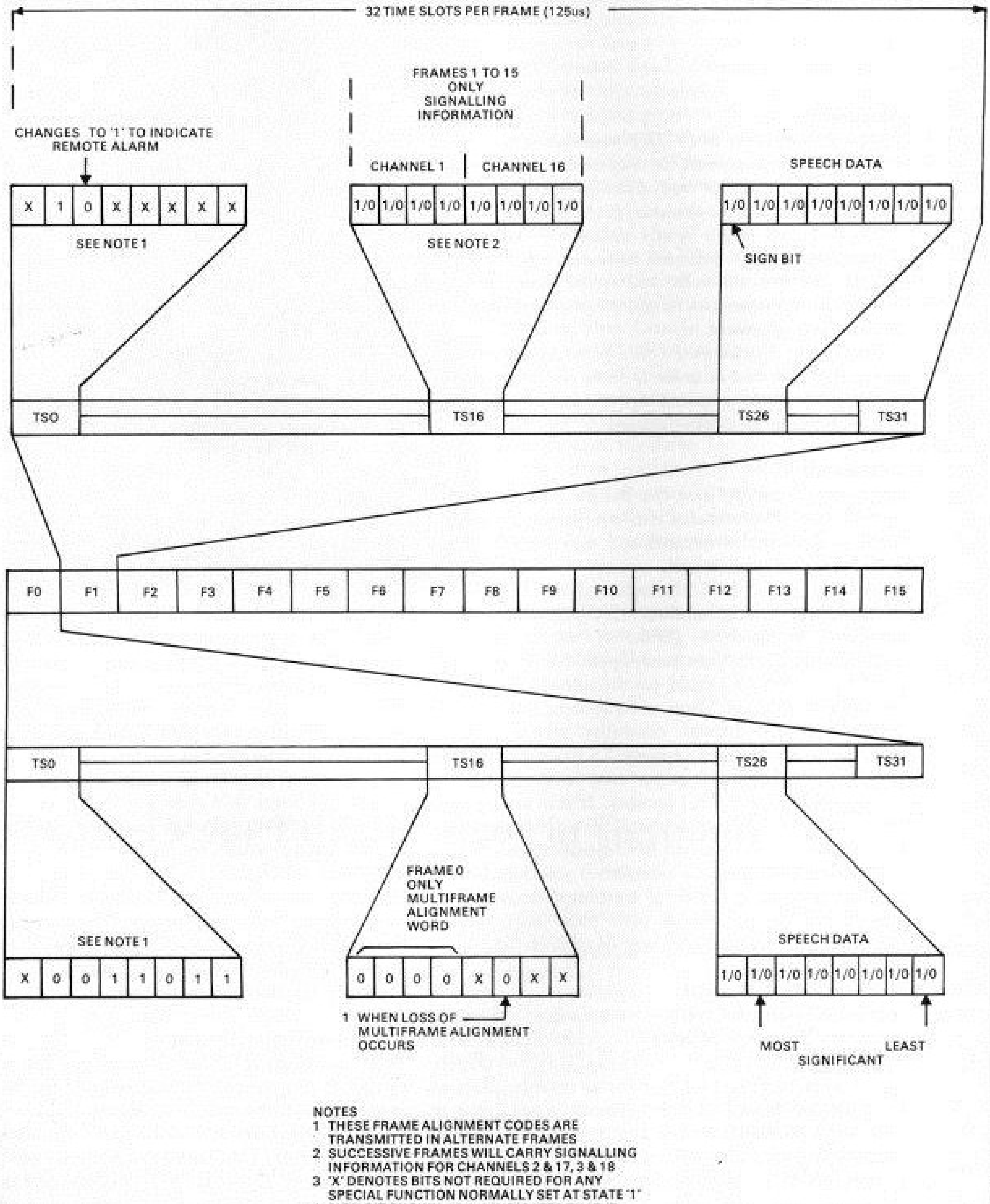
Additionally, there is a reduction of 3dB in the allowable quiescent channel noise to −65dBm0p and 6dB in the intelligible inter-channel crosstalk to −65dBm0, together with reduced roll-off in channel amplitude/frequency response. The supervisory and alarm system is much more comprehensive than in the 24-channel arrangement, as will be seen later in the list of facilities provided for in the alarm unit.

A further significant improvement in overall system performance is obtained by the use of the HDB3 line code.[1] This is characterized by the insertion of 'ones' into the binary signal stream whenever more than three consecutive zeros are encountered. The increased mark density so produced greatly increases the amount of timing information available for the line regenerators, which reduces the effect of line jitter.

In the case of the line equipment, the dependent regenerator units will generally be co-sited with 24-channel, 1.536Mbit/s units and will be installed in similar waterproofed, pressurized housings. The two significant differences in the new design are first, a reduction in unit depth which allows 50 per cent more units to be accommodated in any of the three commonly-used housings, these now taking 6, 18 and 36 units, and second, automatic equalization to match line characteristics over the insertion-loss range of 4–37dB, at 1 MHz.

The transmission format

Before considering the basic elements of the multiplex, the p.c.m transmission format adopted for 30-channel systems should be noted, see figure 1. This shows 32 time-slots per frame and a 16-frame signalling multiframe structure. One encoded amplitude sample from each of the 30-channels (30 time-slots) and two time-slots containing signalling information and alignment signals comprise one frame. The 32 time-slots forming one frame are designated time-slots 0–31, and the 16 frames in a multiframe are designated frames 0–15. In each frame, time-slots 1–15 and 17–31 are allocated to the 30 speech channels. Time-slot 0 is occupied in alternate frames by either the frame alignment words or the time-slot 0 data word. Time-slot 16 carries the signalling information on a channel-associated basis in frames 1–15, the time-slot 16 in frame 0 being allocated to the multiframe alignement signal. The eight signalling bits in time-slot 16 consist of two separate groups of 4 bits representing the signalling codes for each of two



NOTES
1 THESE FRAME ALIGNMENT CODES ARE TRANSMITTED IN ALTERNATE FRAMES
2 SUCCESSIVE FRAMES WILL CARRY SIGNALLING INFORMATION FOR CHANNELS 2 & 17, 3 & 18
3 'X' DENOTES BITS NOT REQUIRED FOR ANY SPECIAL FUNCTION NORMALLY SET AT STATE '1'
4 SIGN BIT SHOWN AS 1/0 INDICATES THAT IT CAN BE 1 OR 0

Figure 1. Transmission format

channels, the channels being separated by 16 time-slots. The signalling information appertaining to a particular channel is thus updated once in every 16 frames, corresponding to a sampling rate of 500Hz. With 8 bits per time-slot, 32 time-slots per frame, and a sampling-rate of 8kHz, the gross bit-rate is $8 \times 32 \times 8000 =$ 2048kbit/s. The frame period is thus 125μs, the time-slot period is 3 906ns and the bit period is 488ns.

System arrangement

Typically, a p.c.m system comprises two sets of rack-mounted terminal equipment, located within the exchange environment and interconnected by line equipment and junction cable, see figure 2.

Within each set of terminal equipment there are further equipments, namely:

(1) PCM multiplex equipment, including signalling units and alarm extension facilities

(2) Line-terminating equipments and ancilliary equipment providing alarm, fuse and speaker facilities.

SERVICES AVAILABLE

The multiplex equipment provides for the transmission, in time-division multiplex, of 30 speech channels and the necessary supervisory and system alignment (frame and multiframe) information over two cable pairs (one for each direction of transmission) at a gross digit-rate for the aggregate signal of 2.048Mbit/s. The spare digits in the frame and multiframe alignment signals can be used for data transmission at bit-rates up to 28kbit/s and 1.5kbit/s respectively. These bits are marked X in figure 1. The performance complies with CCITT Recommendations G711, G712 and G732, and satisfies the requirements of BPO Specification RC5549.

Description of units

Functionally the multiplex equipment divides conveniently into transmit and receive terminals and a signalling sub-multiplex, see figures 3, 4 and 5.

THE TRANSMIT TERMINAL

This basically consists of five transmit channel cards each handling six speech channels (fed from the associated signalling units), an encoding unit common to all channels, plus a transmit clock, transmit signalling sub-multiplex, shared power supply and alarm unit.

TRANSMIT SIGNALLING SUB-MULTIPLEX

Commencing at a 'typical' signalling unit (one of 30, figure 4) the signalling conditions and speech analogue signals on the exchange pair are applied to the 2W input of the signalling unit. The speech signals are fed via a hybrid-connected transformer to the 4W input terminals of the appropriate transmit channel unit. The d.c signalling conditions are applied to the signalling encoder which is activated at the appropriate time by one of 32 secondary transmit signalling select pulses derived from the time-slot 16 transmit unit. The resulting 4-bit signalling code is applied to a common signalling transmit highway and, because the select pulses activate the signalling encoders sequentially, the signalling codes for all channels (plus the multiframe alignment word and data) are interleaved, i.e time division multiplexed on the highway to form a 64kbit/s 'signalling multiplex', which is fed to a transmission-rate converter in the transmit clock unit.

With a transmission-rate of 64kbit/s, eight 'signalling bits' will be fed into the transmission-rate converter in a time equal to one frame period. These digits, which could comprise the multiframe alignment word (4 bits) and multiframe data (4 bits) or signalling codes for each of two channels, are stored as they are received and then inserted by a combining circuit into the outgoing bit stream at the nominal transmitted bit-rate of 2048 kbit/s during the next time-slot 16 period.

Reference was made earlier to the generation by the time-slot 16 transmit unit of the 32 sequential secondary transmission signalling select pulses. Thirty of these are applied to the signalling units to control the production at the appropriate time of the signalling codes. One functions as the multiframe alignment word select pulse and is applied to the multiframe alignment insertion circuit to control the generation of the alignment word (0000) and its insertion during the first half of time-slot 16 of frame 0 onto the common transmit signalling highway. The remaining pulse is a multiframe data select pulse which controls the insertion of the multiframe data word XOXX onto the highway during the second half of time-slot 16 of frame 0.

The unit also incorporates buffering for the 64kHz clock which is then fed to all signalling units on the 64kHz highway.

SPEECH SAMPLING AND ENCODING

Each channel circuit is essentially a lowpass filter whose output is time sampled at a rate of 8000Hz by an electronic switch controlled by channel timing pulses from the transmit clock unit. The continuously varying speech analogue signals are reduced to a train of pulse amplitude modulated (p.a.m) speech samples. Each speech sample has a discrete value, the voltage amplitude being proportional to the speech level at the moment of switching, and is therefore suitable for subsequent encoding.

The outputs of the odd-numbered channel circuits are applied in sequence to the common p.a.m highway A and those of the even channel circuits to p.a.m highway B. The pulses on both highways together with signals from the Codec Test Unit, are fed to the encoder unit.

These samples are processed in turn by the encoder. This is a non-linear circuit of the feedback type in which the amplitude compression and encoding of each sample is effected simultaneously. Companding, the process of compressing the amplitude of the signal at the encoder and expanding the signal at the decoder, follows the logarithmic A law (see figure 6 and note below).

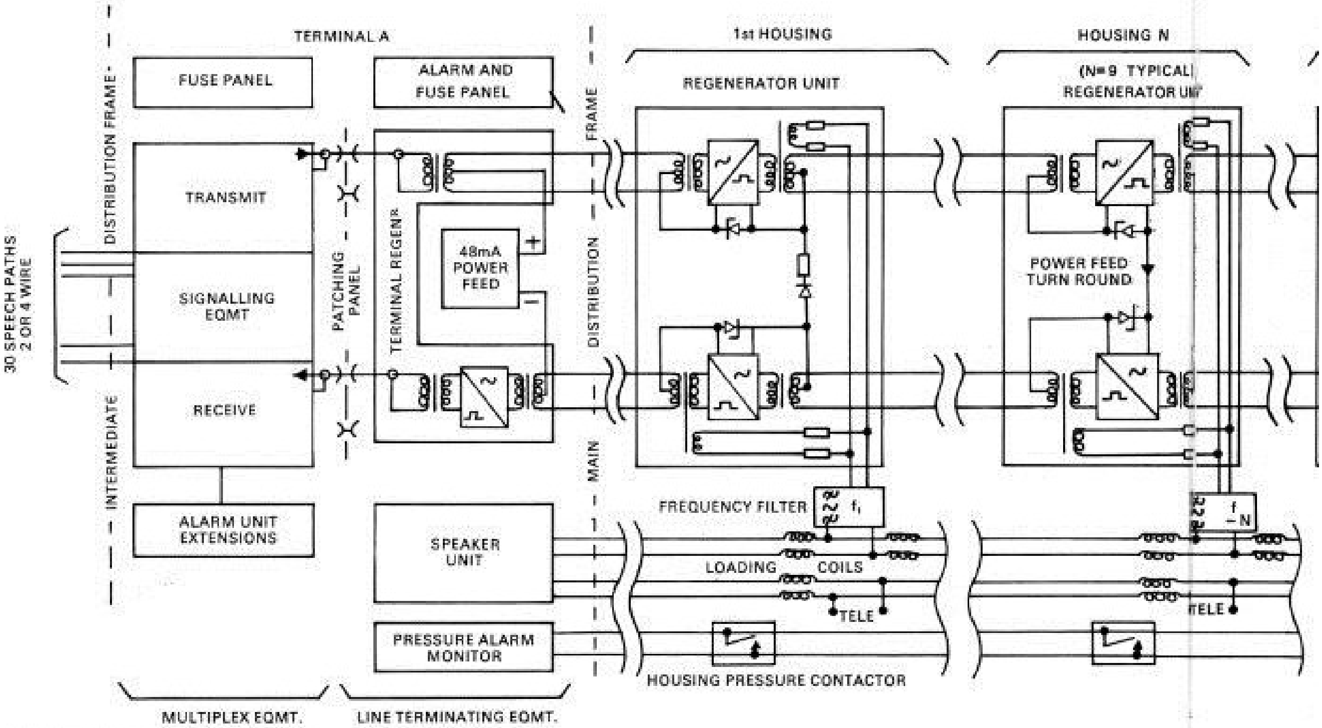


Figure 2. A typical p.c.m system

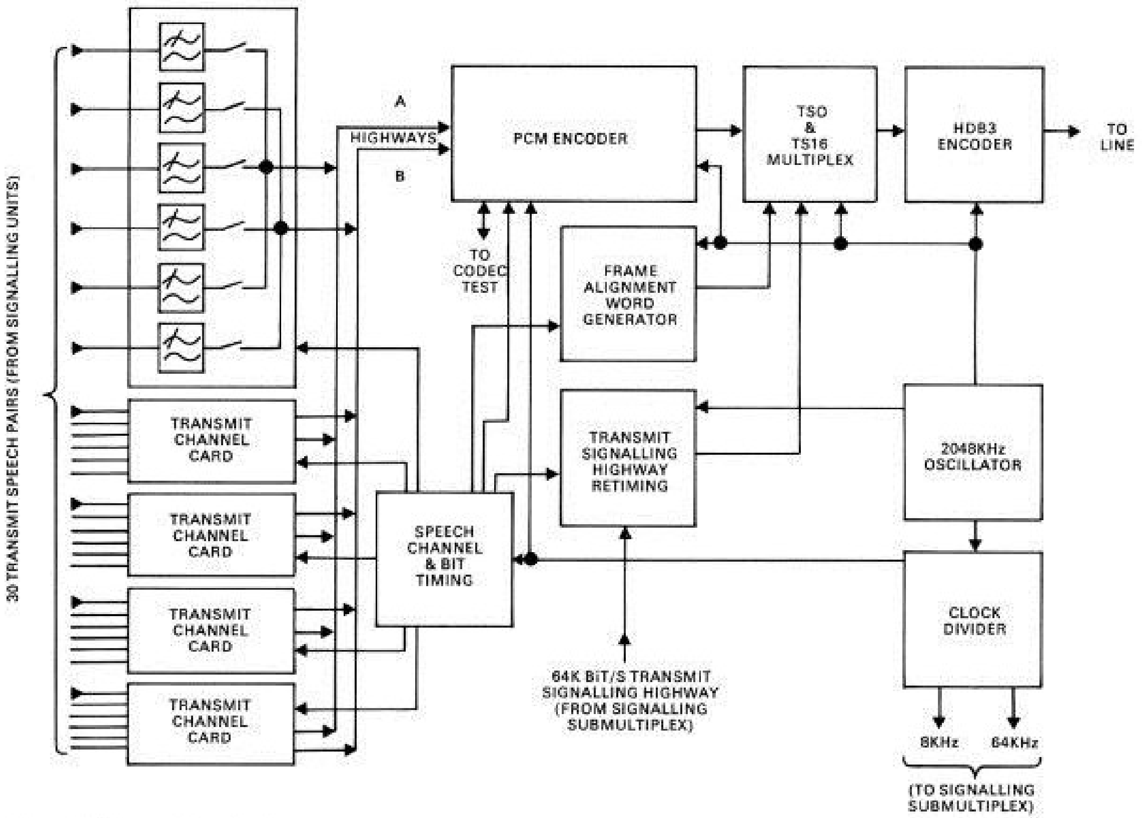
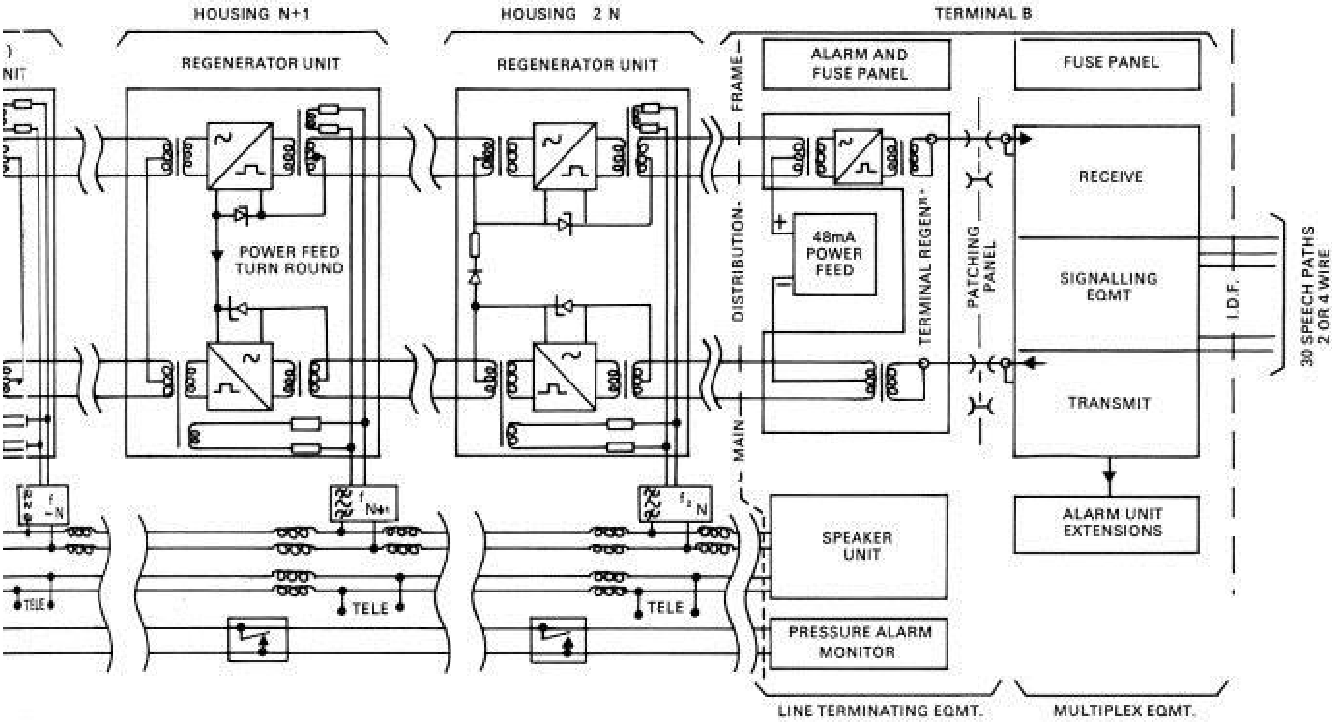


Figure 3. Transmit terminal



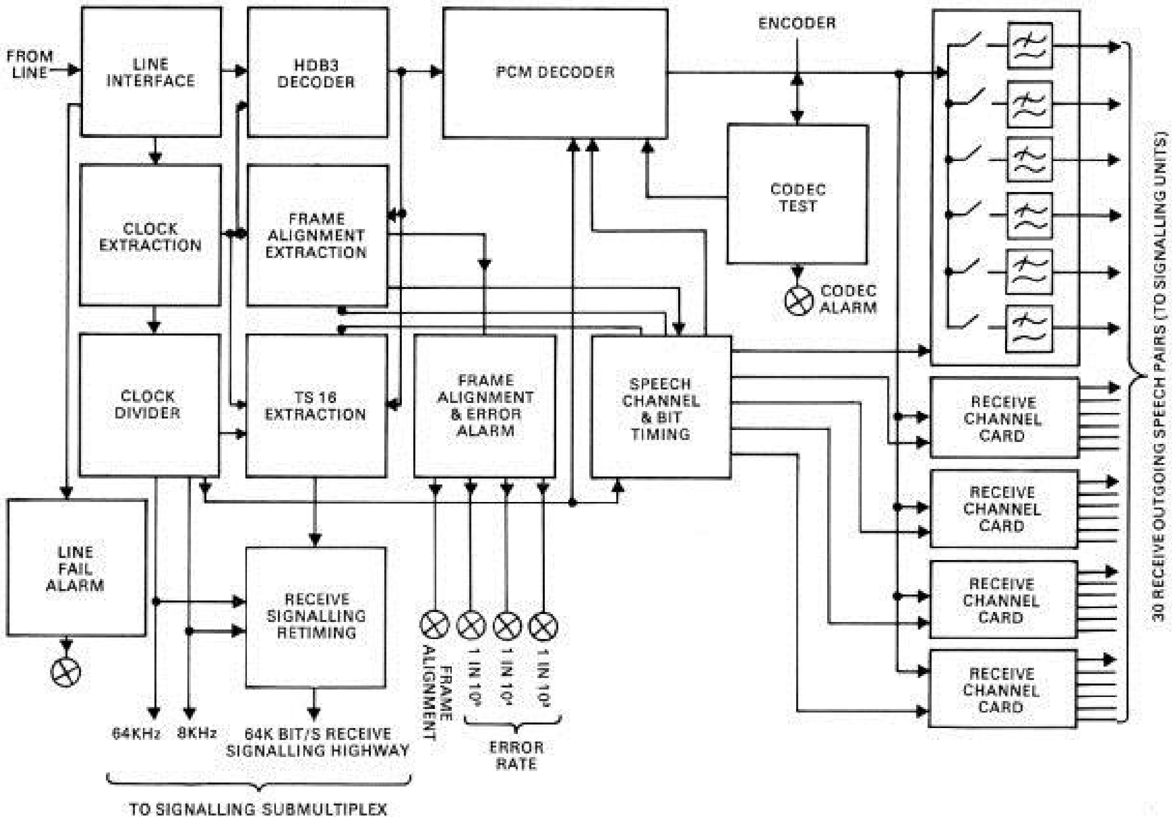


Figure 4. Receive terminal

Each p.a.m sample is temporarily held in a capacitive store and its amplitude compared with reference voltages generated by the encoder, a successive approximation method being used which results in the production of a serial 8-bit binary code word, in which the first digit indicates the polarity and the remaining seven digits denote the magnitude of the sample. Note that sample coding takes place in 4μs, and is equivalent in accuracy to a linear code using 12 bits. The encoder output words are fed to the data combining logic in the transmit clock unit, where the even digits are inverted before insertion in the correct time slots in the outgoing bit stream. The words are also fed to the Codec Test Unit. To perform its sequence of operations correctly the encoder depends on timing signals generated in the transmit clock unit.

QUANTIZING AND COMPANDING

The number of quanta into which the peak-to-peak input signal amplitude range can be divided for coding purposes is limited, by restrictions on transmitted bit-rate, to a finite number, in this case 256 (2^8). This represents the number of code words defining decision levels. A sample whose amplitude lies between adjacent levels is coded as the lower level so introducing an error referred to as quantizing noise. Strictly speaking, noise is random and uncorrelated with the signal whereas the distortion introduced by coding error is signal dependent. The mean square error is interpreted as noise power and can be shown to have a constant value of $\Delta^2/_{12}$ for a uniform Codec for all signal levels, where Δ = one quantum step.

With telephone speech signals the probability distribution of amplitudes is such that small amplitudes predominate, hence total quantizing noise power can be reduced by reducing the quantum step size at low levels and increasing it at high levels. Since the dynamic range of telephone speech traffic due to differences between talkers and connecting circuits can exceed 40dB it is clear that a Codec using linear quantizing could produce a S/N ratio for weak signals 40dB less than that for strong signals. Hence non-linear companding is essential to maintain an adequate standard of transmission quality.

The companding law [2 and 3] used is the CCITT recommended A law which is a segmented approximation to the curve given by:

$$y = \frac{1+\log Ax}{1+\log A} \quad \text{for } 0 \leqslant x \leqslant 1/A$$

and

$$y = \frac{Ax}{1+\log A} \quad \text{for } 1/A \leqslant x \leqslant 1$$

The value of A determines the S/N ratio over the dynamic range. $A = 87{\cdot}6$ is used for speech companding on telephone circuits in the UK and Europe. Figure 6 shows the realization of the law by a 13-segment piece-wise linear curve. This law is now emerging as a world standard except in North America and Japan. It is

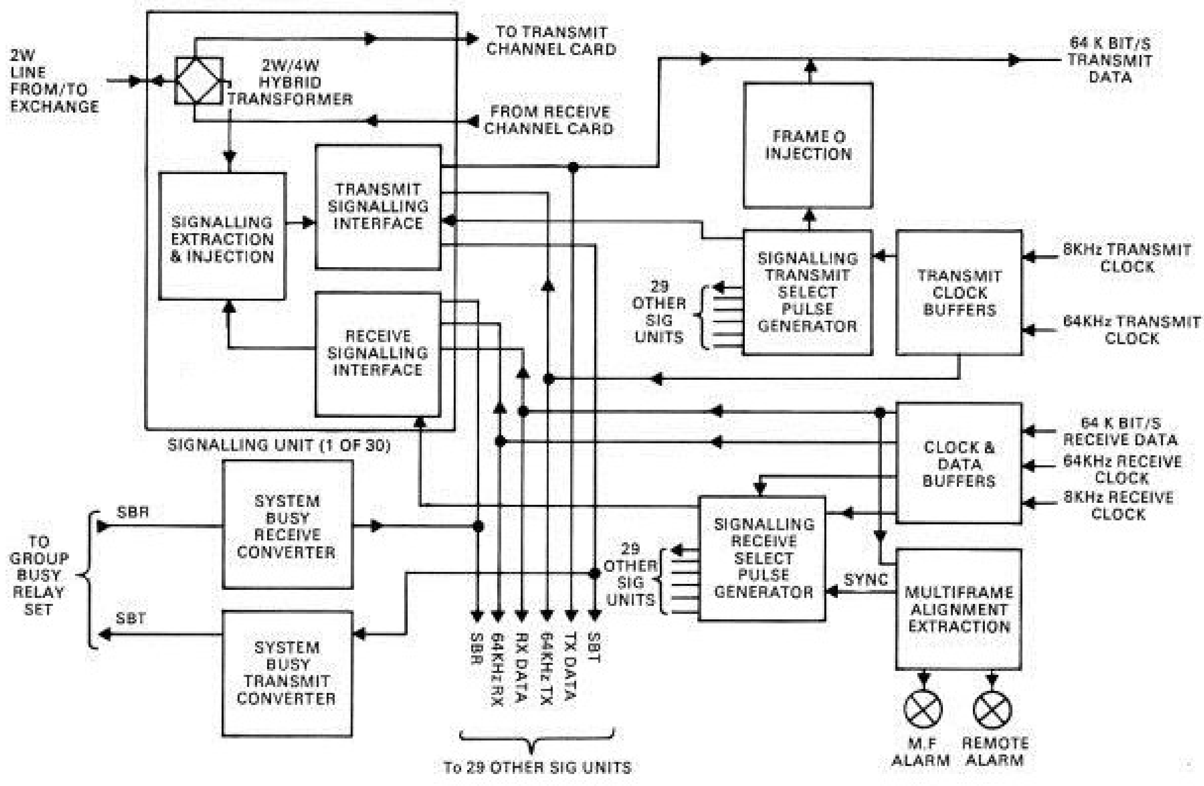


Figure 5. Signalling sub-multiplex

produced digitally, the slopes of the segments all being related by powers of 2.

The centre segment covers plus/minus 32 quantum steps giving effective linear coding for weak signals. The other 12 segments each cover 16 quantum steps and progressively compress higher level signals to a logarithmic characteristic.

TIMING AND PULSE DISTRIBUTION

The drive for all timing functions in the transmit section of the multiplex equipment is generated in the transmit clock unit which contains a voltage-controlled crystal oscillator incorporated into a phase-locked loop. It can be locked either to the receiver clock in the same multiplex equipment or to an external 2048kHz sine-wave source. In the absence of a locking signal the oscillator will function in the free-running mode, maintaining the bit-rate of 2048kbit/s within ± 50 parts per million. The oscillator output provides the drive for various divider and decoder circuits which produce all other timing and control signals necessary for the operation of the transmit section. These include eight sequential 488ns bit timing pulses, which are fed to the encoder unit, 32 sequential channel timing pulses, each

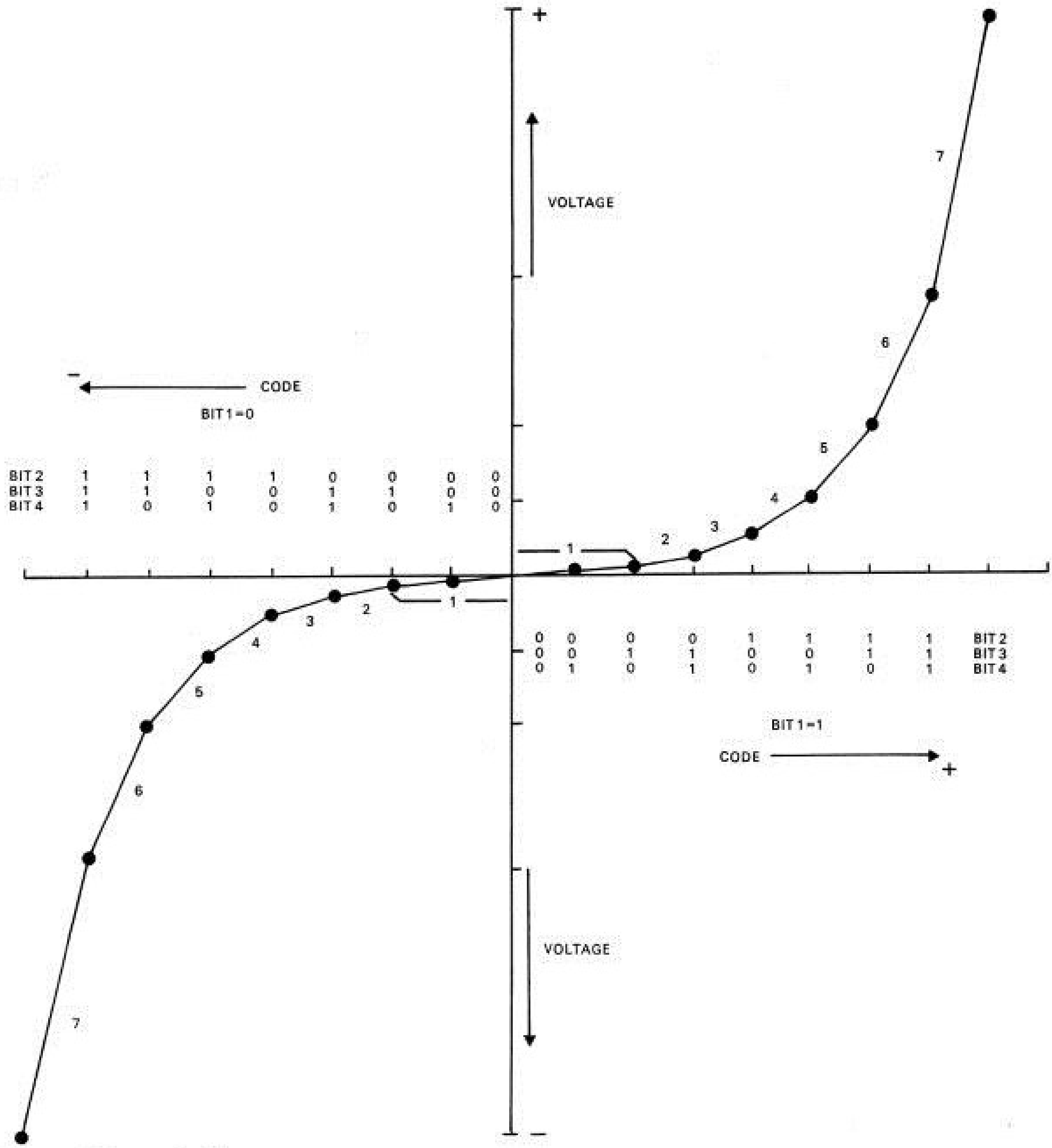


Figure 6. Companding law

2928ns (6-bit periods) in duration and having a recurrence frequency of 8kHz, and clocks to the transmit signalling sub-multiplex. This unit also generates the frame-alignment word and time-slot O data word which are subsequently combined with encoded speech data and signalling information.

The output of the combining circuit is the outgoing bit stream; this is then converted into an HDB3 coded signal to increase further the mark density and, therefore, the timing information content of the line signal. The unipolar signals are then converted to the bipolar line code format in which binary 1 digits are represented by nominally rectangular 50 per cent duty cycle pulses of amplitude $+2.37$V or $-2.37\text{V}\pm10$ per cent (into a 75Ω unbalanced load) and binary O digits are represented by spaces (nominally 0V). The signals are then fed via a matching transformer mounted on the terminal regenerator card to line (at ±3V into 120Ω bal. line).

RECEIVE TERMINAL

This consists of receive clock and timing units, decoder and five receive channel cards each handling six speech channels, shared power supply, alarm unit and receive signalling sub-multiplex.

CLOCK RECOVERY AND PULSE DISTRIBUTION

These functions are handled by the receive clock unit which is fed with the 2048kbit/s, HDB3 coded signal from the terminal regenerator mounted in the line terminating equipment. The signals, which are in bipolar, 50 per cent duty cycle form at line signal levels, are applied to the line interface circuit which converts them to unipolar, 50 per cent duty cycle signals at nominal TTL levels and feeds them to the clock extraction circuit, to the line alarm circuit and to the HDB3 decoder.

The timing information inherent in the received signal is used to drive the clock extraction circuit which produces the 2048kHz receive clock signal. The line alarm circuit monitors the output of the line interface circuit. In the event of a signal failure it generates a line alarm signal and also temporarily replaces the decaying 2048kHz receive clock signal (which is dependent on the incoming signals) by the 2048kHz transmit clock signal. This allows the equipment to continue operation until the alarm has been completed and all channels have been 'busied out'.

The output of the line interface circuit is the unipolar equivalent of the HDB3 coded line signal. Before the encoded speech and other information contained in the signal can be extracted and decoded the signal must be restored to its original binary digital form (as it appeared at the output of the signal combining circuit in the transmit clock unit at the other end of the link). This function is performed by the HDB3 decoder circuit which, if it recognizes a sequence comprising two consecutive 0s followed by a bipolar violation, substitutes four consecutive 0s in place of the bipolar violation and the three preceding digits. The output of the HDB3 decoder is a unipolar, 100 per cent duty cycle, bit stream which is fed to the receive timing unit.

The unit also contains the signalling data transmission-rate converter. The function of this circuit is to extract the information conveyed in time-slot 16 (TS 16) from the 2048kbit/s received data stream (after it has been retimed in the receive timing unit) and re-route it on the primary-receive highway to the TS16 receive unit, thence to the signalling units as a 64kbit/s t.d.m signal.

As well as generating the 30 sequential secondary receive signalling select pulses referred to later, the TS16 receive unit generates two other pulses. One is applied to the multiframe alignment circuits to extract the multiframe alignment word from the time-slot 16 information appearing on the 64kbit/s highway from the receive clock unit.

If the multiframe alignment word in each frame O is recognized, the multiframe alignment is correct and no further action is necessary. If two consecutive multiframe alignment words are incorrectly received, loss of alignment is assumed. Local and distant terminal alarms are raised, incoming signalling units are released and outgoing signalling units are inhibited against further seizure by applying 'camp on busy'. A multiframe realignment search is commenced and when alignment is regained, indicated by the receipt of the multiframe alignment word at the correct time, normal operation is restored.

The second of these pulses, performs a similar function, activating the input shift register at the correct time for the extraction of the multiframe data word.

The unit also controls the 'busying out' of signalling units under fault or maintenance conditions.

At the receive terminal the incoming digital stream is monitored by a recognition circuit. If the frame alignment word is recognized, i.e correctly received, the system is in frame alignment and no further action is necessary. If three consecutive frame alignment words are incorrectly received, i.e contain one or more errors, loss of frame alignment is assumed. Local and distant terminal alarms are given, the receive side speech circuits are suppressed and the receive signalling highway is inhibited. A frame realignment search is then initiated whereby the appropriate timing circuits are adjusted until frame alignment is regained, as indicated by the presence of a correct frame alignment word in one frame, absence of it in the next frame and the presence of a correct alignment word in the following frame whereupon normal operation is restored.

DECODING AND SPEECH RECOVERY

The receive timing unit also controls the routeing of speech data to the decoder unit. This decodes the 8-bit words representing the speech samples, producing reconstructed amplitude modulated pulses which, allowing for the inherent quantizing distortion, equals those which were encoded at the transmit terminal.

The decoder is a non-linear device which decodes and simultaneously expands each encoded sample, its transfer characteristic being the inverse of that of the encoder at the transmit terminal. It comprises control logic (consisting of an 8-bit serial in-parallel out shift register and decoding logic), current sources (with associated precision voltage supplies), a voltage divider network and an output circuit consisting of an electronic switch and a voltage follower amplifier.

The serial bit stream from the receive timing unit is clocked into the 8-bit shift register at the bit-rate. The parallel outputs of the shift register are connected to the 8-bit store in the decoding logic. When the shift register is full the data present on its output terminals is clocked into the store by the decoder strobe pulse from the receive timing unit. The stored data is then used by the decoding logic to operate the appropriate current sources which, in conjunction with the voltage divider network and the electronic switch, determine the polarity and magnitude of the resulting pulse applied to the voltage follower amplifier. The output of the amplifier is thus a train of amplitude modulated pulses, each a reconstructed speech sample, (or a 'Codec test' signal), and this is applied to the receive p.a.m highway which serves the receive channel units and the Codec test unit. The reconstructed samples are offset from the encoding decision levels by half the appropriate quantum step size, so reducing quantizing error.

Each receive circuit, which consists essentially of a capacitive store followed by a lowpass filter, is preceded by an electronic switch. The switch, controlled by the appropriate channel timing pulse from the receive timing unit, is closed at the correct time to admit only the reconstructed sample pulses pertaining to the required channel of the receive circuit.

The series of pulses held in the capacitive store are fed via emitter follower stages to the 3.4kHz lowpass filter. The output of the filter is an analogue audio signal which, after passing through a frequency-compensating network to correct the aperture effect introduced during the conversion from p.a.m signals to analogue audio signals, is a replica of the original speech analogue signals. These signals are amplified and applied via the balanced, 600Ω, 4W output of the receive circuit to the 4W receive input of the associated signalling unit, and thence to the called subscriber.

The signalling information pertaining to the channel is extracted from the 64kbit/s multiplexed signalling information on the highway from the TS16 receive unit, referred to earlier. This is effected by the appropriate select pulse which activates the signalling word decoder in the signalling unit at the correct time to enable it to enter only the relevant signalling word. This is then decoded and converted to the appropriate d.c signalling conditions on the lines to the exchange.

Signalling units

Thirty-channel p.c.m equipment (BPO Type 2200A) caters for two basic forms of signalling,[4] 'loop disconnect' and 'earth disconnect'. The facilities offered by these two forms are as follows:

LOOP DISCONNECT SIGNALLING UNITS

These are basically of two types, outgoing and incoming. The first of these connects the calling subscriber to the p.c.m multiplex and the second connects the p.c.m multiplex to the called subscriber via the appropriate switching equipment.

The connection to the switching equipment is made via three- or four-wire circuits. Two wires, designated positive (+ve) and negative (−ve), are used for carrying speech current, a hybrid transformer connected at this point providing the two- to four-wire speech interface with the multiplex transmit and receive channel units. A third wire designated the 'private' or P wire is used for guarding and holding any connection set-up. The P wire can also be used for extending signals back to the exchange for metering purposes. Alternatively, metering may be carried out over a fourth wire designated the 'modulator' or M wire.

FACILITIES

Each signalling card provides a number of facilities realized by logical conversion, some of which are selectable by means of straps, to suit particular exchange interface requirements. These include earth or battery testing switches, manual hold, forward or backward holding, dialled pulse correction, single-fee metering, metering-over-junction (MOJ) coin and fee checking (C & F C), trunk offer (TKO), forced release, called number interception (CNI), silent reversal of d.c loop potentials during metering, etc. The codes employed are listed in table 1.

No other form of conversion set is required in the exchange when this type of signalling unit is employed.

Table 1
Signalling codes employed

Digits 1–4 (or 5–8) appearing serially in TS16 in all frames except O	Forward signalling condition	Backward signalling condition
a b c d		
1 1 1 1	Circuit idle	Circuit busy
0 0 1 1	Circuit seized	Called subscriber answer
1 0 1 1	Dial break	Not used
0 1 1 1	Not used	Circuit free
0 0 0 1	TKO	Manual hold
1 0 0 1	Not used	CFC
1 1 0 1	Disconnection	Disconnection
0 1 0 1	Earth	Earth
0 0 0 0	Not used – signifies multiframe alignment in frame O	

EARTH DISCONNECT SIGNALLING UNITS

These are four-wire units, interconnecting outgoing, incoming or bothway exchange units via transmit and receive pairs to the speech channels of a p.c.m multiplex.

For this type of signalling the unit must provide some form of conversion for the transmission of the signalling information. This is usually achieved by a two-wire connection, one lead designated 'M' (modulator) the other designated 'E' (Exchange) and referred to as E and M signalling. This method of conversion has been employed for many years with f.d.m transmission equipment.

The E and M leads are connected to the phantom of the four-wires speech circuit. Since the main signalling logic functions are carried out by the exchange equipment only simple conversion from the earth-disconnect conditions to the appropriate p.c.m code and vice versa need be performed by these units.

Alarm system

A comprehensive alarm system forms an integral part of the multiplex equipment and continually monitors certain aspects of both the equipment and the p.c.m system performance. The alarm system provides visual indication of a number of fault conditions as below:

(a) POWER indicates failure of the input or derived supplies
(b) CODEC indicates failure of the encoder, decoder, receive timing unit or p.a.m highway
(c) LINE indicates loss of the digital input to the multiplex equipment
(d) FRAME ALIGNMENT indicates loss of frame alignment
(e) MULTIFRAME ALIGNMENT indicates loss of multiframe alignment
(f) ERRORS indicates that the error-rate in the frame alignment words has exceeded 1 in 10^3, 1 in 10^4 or 1 in 10^5, depending on the lamp or lamps lit
(g) REMOTE ALARM indicates that a failure has ocurred at the remote multiplex
(h) REMOTE MF ALARM indicates that loss of multi-frame alignment has occurred at the remote multiplex equipment.

The extinguishing of the alarm lamps (with the exception of the 1 in 10^5 errors lamp) is controlled by the lamp lock/auto reset circuit. Depending on the position of the LAMP LOCK/AUTO RESET switch on the alarm unit front panel, the lamps either extinguish automatically when the appropriate fault condition is cleared or remain lit until extinguished manually. Four alarm outputs are derived by the alarm combining logic from the inputs applied to the alarm unit (with the exception of the 'errors' input) and the outputs of the 1 in 10^3 and 1 in 10^4 counters. These four outputs are available to:

(a) 'Busy-out' the multiplex equipment and its associated signalling units
(b) Operate the alarm circuit on the associated Alarm Unit
(c) 'Busy-out' relay equipment associated with certain four-wire signalling units
(d) Initiate the transmission of an indication to the remote terminal that the local multiplex has detected a fault condition.

In general the monitoring and fault-detection circuits are located in the unit most convenient for monitoring a particular aspect of equipment or system performance. The circuits which monitor the performance of the encoder, decoder and p.a.m highways (in the same multiplex equipment) are sufficiently complicated to warrant their inclusion in a separate unit, the Codec test unit. Alarm outputs from the various units, including Codec test, are mainly applied direct to the alarm unit although some outputs of the power unit are applied via the time-slot 16 receive unit.

The alarm unit, (BPO Type No.19A) which is normally mounted on the same rackside as the multiplex equipments with which it is associated, provides alarm extension and cut-off facilities for three multiplex equipments. The unit mounts 'Alarm' and 'Receiving Attention' lamps and an alarm cut-off key for each multiplex equipment and incorporates a relay which extends an alarm condition to the station alarm system.

Any of the fault conditions listed with the exception of the 1 in 10^5 errors alarm, causes the appropriate system alarm lamp to light and operates the station alarm to alert the exchange maintenance staff. The alarm lamp on the alarm unit indicates in which system the fault has occurred and the alarm lamp on the multiplex alarm unit of the faulty system indicates the type of fault. By operating the alarm cut-off key associated with the faulty system the alarm lamp on the alarm unit is extinguished, the 'Receiving Attention' lamp lights and the alarm condition is removed from the station alarm. This is then free to give another alarm should a fault occur on either of the two systems which are still operational and will also give an alarm when the original fault is cleared, until the alarm cut-off key is returned to its original non-operative position.

Power supply

This is derived from a high-efficiency switching mode d.c to d.c converter operating at 25kHz. Regulation is controlled by varying the duty cycle of the switching waveform. With a nominal -50V input from the station battery, the p.s.u provides regulated supplies at $+5$V and ± 12V, plus other partially-regulated supplies for the units forming the multiplex package. Regulated supplies at $+5$V and ± 12V and three $+50$V outputs for the signalling units are also provided. Inputs and outputs are filtered to minimize the effect of incoming transients from both the station battery supply and from signalling units. The unit is protected against externally-applied short circuits.

Careful attention has been paid in the design of the unit to minimize electromagnetic and ultrasonic radiation from the switching components.

Design features

In the signalling units conversion of the analogue information to the required digital format and vice-versa is performed by low power transistor-transistor logic (LPTTL). A typical two-wire loop-disconnect unit would house 21 integrated circuits while a four-wire E & M unit would employ seven. The logic circuits perform the functions of re-timing digital signals, persistence checks on the signalling codes, sequencing exchange signals and encoding and decoding time-slot 16 signalling information.

In the multiplex circuits standard TTL has been extensively used to provide the required logic functions, together with MSI where appropriate.

Equipment engineering

The multiplex and line terminal equipments are accommodated in racksides manufactured to construction practice Type 62. This uses standard sheet steel racks in which edge-mounted circuit cards slide into shelves inclined at 15 deg. to the horizontal.

Units are all 330×152mm but of varying widths. Access to units while the equipment is in operation is made possible by the use of test extension frames.

A 9ft rackside can accommodate three 30-channel multiplex systems and the associated alarm extension apparatus, whilst a similar rackside can contain line terminal equipment for up to 48 separate p.c.m systems.

Circuit cards use double-sided-plated-through-hole printed-circuit wiring on fibre-glass base laminate.

Line-terminating equipment and ancillaries

In figure 1 the part between and including the line-terminating equipment comprises a 'Digital Link Section'. In the terminal building the line-terminating equipment is also fitted into Type 62 practice racksides. These are interconnected with the multiplex racks with 75Ω coaxial cable. A typical rack would contain;

(a) A fuse panel providing protected distribution of the station battery to each panel in the rack.

(b) One to four equipment terminating shelves (BPO Type No.2000B). Each of these can contain up to four terminal regenerators Type 13A (or Type 15A surge protected), up to four power-feed units Types 1A or 2A (for 50 or 24V supplies) or their surge protected equivalents Types 8A or 9A, and a centrally mounted patching panel. A fully-equipped shelf caters for four system ends. The patching panel allows the signals in each direction to be selectively connected between the multiplex equipments and the lines to the distant terminals, these being taken out via a main distribution frame. The terminal regenerators, handling the incoming signals, are card-mounted single circuits, identical with those employed in the dependent regenerators, and powered by series connection across the station battery. The card also mounts the 75–120Ω matching transformer connecting the outgoing signal from multiplex to line. The power feed units apply a stabilized 48 ± 2 mA at up to 75–0–75V to the phantom circuits of two digital line pairs to supply the dependent line regenerators. The number of regenerator units, each containing two regenerators, which can be power fed with this voltage from a single terminal end, is typically 8, 10 or 11 when using 0,63, 0,9 or 1,27mm cable.

(c) Speaker equipment (BPO Type No.2000A). This shelf accommodates up to three speaker units which provide access to the supervisory test pairs and engineering order wire circuits in the cable.

BEARER CIRCUITS

The actual bearer circuit is usually a multipair audio junction type cable made up of 0,63mm (10lb) 0,9mm (20lb) or 1,27mm (40lb) diameter wires. The transmission system per p.c.m multiplex terminal is four wire (one pair each go and return, de-loaded). An additional four pairs are required at each regenerator housing to provide for supervisory, engineering order wire or exchange line, housing pressure contactor alarm circuit, and spare. Two modes of operation are possible, single cable working and two-cable working.

The former is generally limited by near-end crosstalk and the need for separation within the cable for the two directions of transmission. The latter is limited by far-end crosstalk, but more pairs can be used.

Regenerator spacing follows 24-channel practice, nominal spacing between housings located in footway or carriageway manholes being 1830m (2000 yards) with a maximum spacing of 1886m (2062 yards). At this distance and on 10lb cable, the attenuation at 1MHz, where the HDB3 line code energy spectrum peaks, can exceed 37dB. The distance from the exchange terminal to the first regenerator housing position is usually held to about 915m (1000 yards) to reduce the effect of impulsive interference from exchange switching equipment. This spacing is also used at intermediate points in the cable where spur cables lead to exchange equipment. The cast iron housings are available in three cylindrical sizes, as well as the rectangular BPO Type CRE No.1. Ten-metre stub cables of appropriate capacity are spliced into the main cable, passing through a waterproof gland on entry to the housing and connect with the regenerators and auxiliary services, including frequency filter unit via flying leads and mating sockets.

REGENERATORS

Each dependent regenerator unit contains two regenerator circuits. Hybrid thick film techniques are used to optimize component packaging in the volume available. In the BPO network the two regenerators within the unit operate in the same direction of transmission, a further unit then being required for the return paths. This method restricts system growth to multiples of two. Circuits can however be connected to provide go and return paths within the same unit. Power is obtained from the voltage drop across a zener diode connected across the centre taps of the line sides of the input and output transformers of each circuit, the power return being made via the second circuit. At the end of a power loop the zener diodes are connected in series, observing polarity. In the case of the surge protected units an additional heavy-duty diode, connected in parallel with the voltage stabilizer, shunts longitudinally induced surge currents around the unit.

Protection against transverse surge potentials is given by careful design of input transformers and by diodes in shunt with the primary of the output transformer.

The regenerator arrangement, as shown in figure 7 although following an almost standard sequence of building blocks, provides a serious challenge to the designer in terms of meeting conflicting requirements. For example, minimization of pattern-sensitive delay (wide bandwidth) and signal noise ratio (narrow bandwidth), the latter being critical in maintaining acceptable error rates.

The circuits provide:

(a) Matching of line to regenerator input impedance followed by an optional line build-out network. This is linked into circuit when preceding line loss is below 20dB at 1MHz

(b) Automatic equalization to optimize pulse shape at the sampling point, thus correcting the inevitable effects of attenuation and phased distortion related to frequency in the preceding cable section. This is achieved by using a d.c current derived from an automatic gain control feedback path, to control a variable network in the regenerator. This current is a function of the loss in the preceding section. Equalization accuracy is optimized for the higher loss cable sections since these would produce the worst signal-to-noise ratios, the effective range being 4–37dB at 1 MHz

Further shaping and amplification follows to bring the signal to a constant peak level before phase splitting to feed level decision and timing extraction circuits

(c) A temperature compensated threshold level equal to half the peak value of signal

(d) Generation of timing pulses. This is achieved by full wave rectification of the equalized and threshold detected signal, and using the resultant pulses to resonate an L-C circuit accurately tuned to 2·048Mbit/s. Its output is squared and differentiated to produce positive and negative timing pulses.

Phase compensation is provided to ensure that the sampling pulse occurs at the centre of each bit period to gain maximum immunity to interference and noise. To maintain system performance clock-tuning drift is held within 6kHz over the life of the equipment (25 years).

The Q of the tuned circuit is a compromise. It must be low enough to satisfy the drift requirement yet high enough to adequately suppress timing jitter which, if allowed to build up along the digital line, causes pulse position modulation at the receive terminal which leads to crosstalk and distortion in the recovered analogue signal. Typical values used in the UK lie between 30 and 50.

(e) Finally, regeneration to the original pulse shape is achieved by applying the amplified and equalized signal and the timing pulses to two pairs of diodes connected as AND gates. These are each followed by a Schmitt trigger circuit; one generates an accurately defined positive going output pulse in the line-matching transformer whenever the positive timing pulse coincides with a positive signal pulse exceeding the threshold value; the other trigger circuit similarly handles negative going output pulses. The negative going timing pulse clears the triggers.

Supervision of regenerator performance continues as in 24-channel practice with the transmission from the terminals of a special trios pattern consisting of groups of pulses in which twice as many pulses of one polarity occur compared with the other polarity. This unbalance is varied at an audio rate at frequencies unique to each housing and provides an audio component which is fed to the associated filter frequency unit for transmission back to the terminal on a common loaded audio pair.

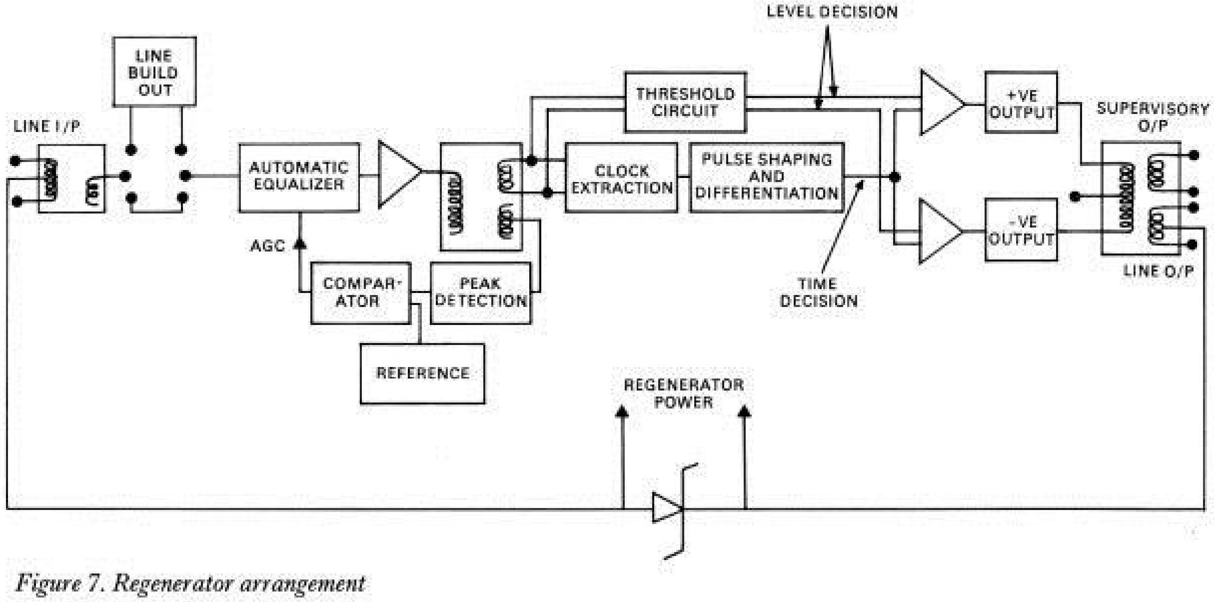


Figure 7. Regenerator arrangement

It is interesting to note that, based on service data derived from installed 24-channel systems, the BPO have calculated that the MTBF for a typical regenerator is of the order of 250 years.

Equipment development programme for TDM in the UK

The BPO development programme[5] shown in figure 8 is geared in such a way that (1) the t.d.m network can gradually interwork with the f.d.m network (2) the t.d.m systems, as far as possible, should be capable of using existing resources and in the early phases with little or no change to these resources and (3) it will be possible for t.d.m system capacities to build up rapidly

BPO Equipment development programme

PHASE 1 In service from 1978 onwards	(a) 2Mbit/s line system (b) 120Mbit/s line system (c) Multiplex equipment upto 120Mbit/s (2, 8 and 120Mbit/s) (d) Supergroup Codec
PHASE 2 In service from 1980 onwards	(a) 11GHz digital radio relay (140Mbit/s) (b) 140Mbit/s tie system (c) 8/140Mbit/s multiplex equipment (d) 8Mbit/s line system
PHASE 3 In service from 1982 onwards	(a) Hypergroup Codec (b) *Circa* 500Mbit/s line system (c) Multiplex up to 500Mbit/s (d) Waveguides (under consideration)

Figure 8. BPO equipment development programme

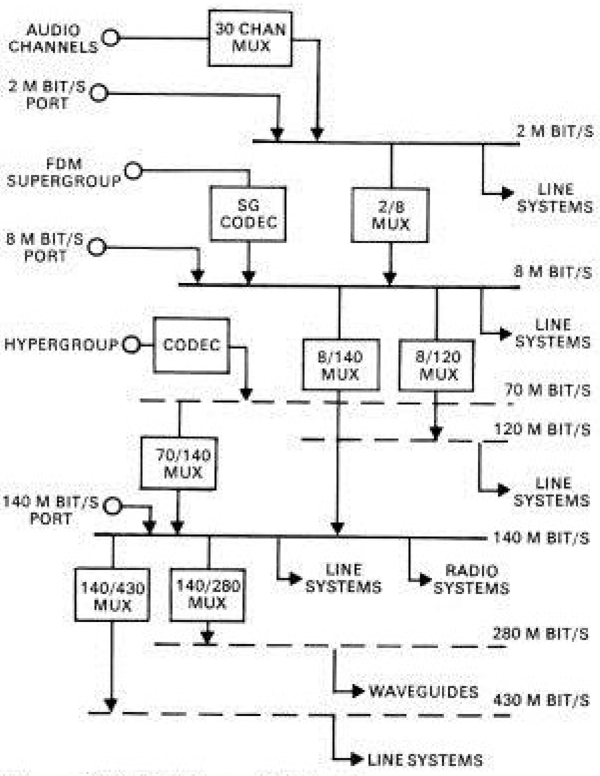


Figure 9. Digital hierarchical levels

so that they are comparable with existing f.d.m systems. The BPO envisage that a period of 10 ± 2 years will be required to achieve the situation whereby 99 per cent of the annual growth of the network will be met by t.d.m plant. At the end of this period the proportion of the total network capacity in t.d.m form will be in the range of 40 to 50 per cent.

Figure 9 illustrates the various digital levels that will be used in the network and how they relate to one another. It will be observed that the chosen hierarchical structure consists of three levels these being, 2, 8 and 140Mbit/s as standardized by CEPT[6], together with a choice of levels which are suitable for line systems operating in the existing infra-structure of the UK trunk network, consistent with the present state of high-speed digital technology. At the same time the structure is flexible enough to respond without much difficulty to the following services:

Music	2Mbit/s (6 channels)
Viewphone	2 or 8Mbit/s
Confravision	35 or 69Mbit/s
Television	69Mbit/s

Conclusion

The available information on the t.d.m expansion programme planned for the UK clearly shows the importance of the 2Mbit/s level of equipment (30-channel) in providing the essential and increasing rate of channel capacity necessary to drive the higher order systems; initially complementary to the f.d.m inputs.

Looking at the present state of the art it is apparent that 30-channel p.c.m is now an engineering fact and must therefore, in time, become the established system. In the meantime, 24-channel systems will obviously continue in use until such time as they become due for replacement, having reached the end of their useful operational life or because they are uneconomic in terms of traffic requirements.

Acknowledgement

The design and production of complex electronic equipment is invariably a group activity, involving a large number of people. Here it is only possible to acknowledge, in particular, the hard work and enthusiasm which has been maintained over a long period of time, by each and every member of the design group.

References

[1] A. CROISIER: 'Introduction to pseudo ternary transmission codes', *IBM Journal of Research and Development* (July 1970) pp.354–367.

[2] Bell Telephone Laboratories book: *Transmission Systems for Communications*, 4th edition (1970).

[3] K. W. CATTERMOLE: *Principles of Pulse Code Modulation* (Iliffe Books 1969).

[4] N. D. GILBERT 'Controlling call connections for 30-channel p.c.m multiplex equipment', I.E.E. 1975 Conference Publication No.131, *Telecommunication Transmission*, pp.61–64.

[5] W. G. SIMPSON: 'A review of transmission network planning', *ibid.*, pp.1–5.

[6] Conférence Européenne des Administrations, Postes et Télécommunications.