DATA COMMUNICATIONS

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DATA COMMUNICATIONS

INTRODUCTION

1. The telecommunications industry is currently undergoing a great change, with the replacement of traditional analogue communications systems by those using digital techniques. To cope with this, the System Engineer must achieve certain skills in data communications.

2. These include:

(1) Knowledge of basic data communication techniques, including line transmission techniques, modem functions and the V24 interface between modem and computer equipment. An ability to interpret V24 indications on patch panels, modems, terminals and test equipment is advantageous.

(2) Ability to distinguish between a problem caused by network hardware and a problem in the software.

(3) An overall understanding of the communication system (network) in use, sufficient to allow a reasonable consideration of all possible causes of a failure and to confidently direct its localisation and isolation. This is particularly important when dealing with organisations outwith the Royal Navy.

GENERAL

3. Data communications may be understood as meaning the transfer of non-voice information between locations, the information being structured and codified to aid 'processing'. The data is that normally associated with computer systems.

4. The field of computer terminals and modems with potential for new data application has evolved rapidly as has the change of telephone network to integrated digital, combined with optical fibre transmission. This has introduced the possibility of a truly Integrated Services Digital Network (ISDN).

5. While this is still some way in the future for the Services, BT and other circuit providers around the world are already offering these facilities to subscribers. Since the Services appear to be structuring their future telecommunications systems requirements around current commercial practices, personnel involved in military telecommunications system operations, maintenance and management must be aware of the techniques and nomenclature involved.

Baseband Processing

6. Although the information basebands developed in computer systems is digital and will be suitable for use on digital transmission systems with little or no processing, for some considerable time we will have to cope with:

(1) Analogue basebands on Digital systems or

(2) Digital basebands on Analogue systems.

7. For this reason it is necessary to consider baseband Analogue to Digital (A to D) conversion and vice versa.

8. The method specified by CCITT and adopted by the world's telephone authorities is Pulse Code Modulation (PCM).

9. At the sending end, the audio baseband input is sampled at regular intervals to give Pulse Amplitude Modulation (PAM) (Fig 9.1).

10. While this produces a digital signal, transmission of information in PAM form would not be practical, due not only to the amount of distortion the pulses would be subject to, but also the amount of interference such a signal would cause to other system users.

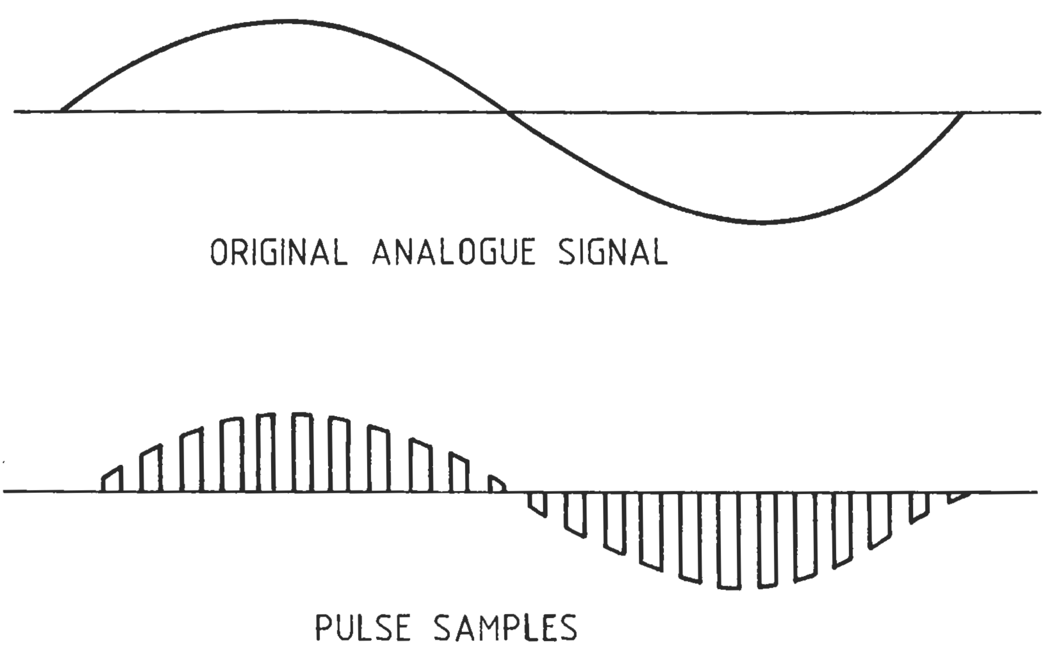

FIG 9.1 PULSE AMPLITUDE MODULATION (PAM)

11. To overcome this, Pulse Code Modulation is used whereby each sample is sent to line as a digital code formed by a pattern of bits, each code uniquely representing the amplitude of the sample. Using 'm' bits, 2^m different sample values can be identified. This process converts the analogue input to a digital output (A to D).

12. At the receiving end, the decoding process uses the reverse procedure (D to A) to reconstruct each sample from the incoming groups of bits. When the recovered bits are fed through a low pass filter, the original analogue base band is recovered.

Quantisation

13. The process of adopting a number of sample levels, each of which is assigned a discrete value is known as 'quantisation'.

14. To represent the value of an analogue signal over a wide range of amplitude values would require a large number of sampling levels and hence many binary digits to represent the level (Fig 9.2).

15. The greater the number of sampling levels, the greater the number of bits in the transmitted code and the less the quantising distortion. The smaller the number of levels, the less the number of bits, but the greater the quantising distortion.

16. The encoder finds a quantum level nearest below that of the input sample level and describes this in binary code. An 8-digit code is normally used giving 256 different sample values. This is realised by 128 levels in each +ve and -ve half cycle (7 bits) with the remaining bit indicating the sign.

Quantising Distortion

17. PCM transmission introduces a transmission impairment known as 'quantising distortion' due to the small difference between the output signal and the original input signal and the signal reconstructed from the quantised signal levels differs from the original signal.

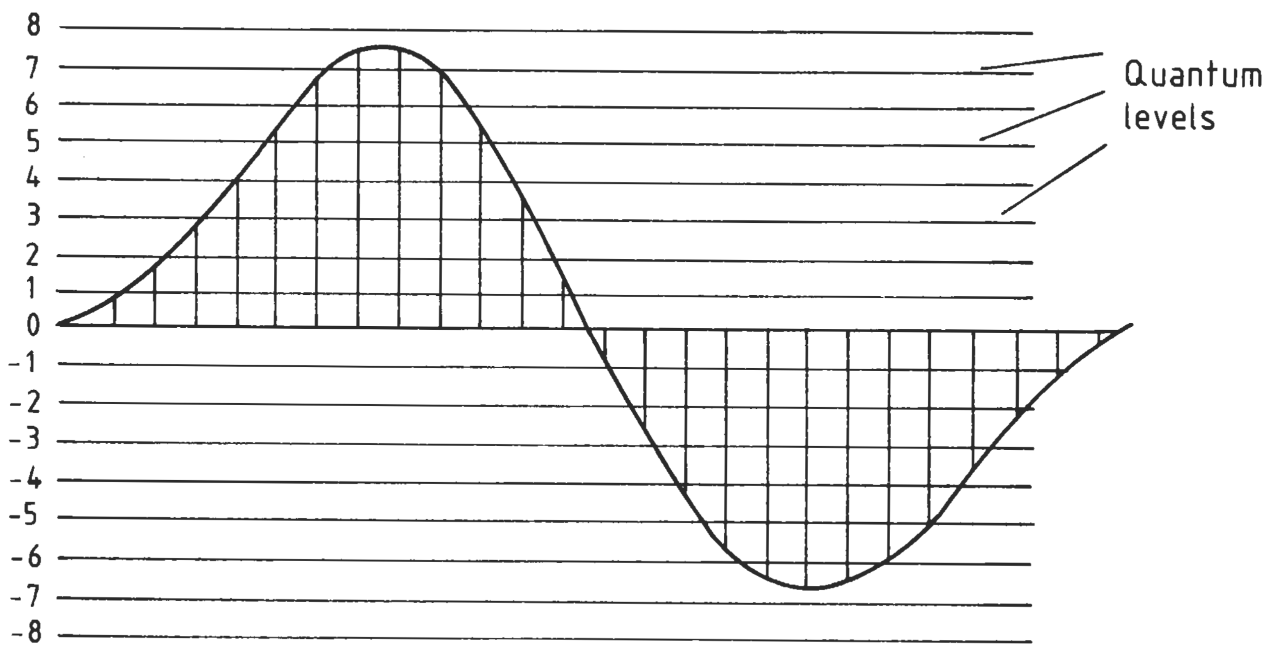

FIG 9.2 SAMPLE VALUES OF SIGNAL (LARGE AMPLITUDE)

18. If the encoder uses quantising steps of uniform size, then a large amplitude signal is represented by a large number of levels and is reproduced with little distortion (Fig 9.2). A small amplitude signal however will require only a small number of steps and a large percentage distortion will be present in the output signal (Fig 9.3).

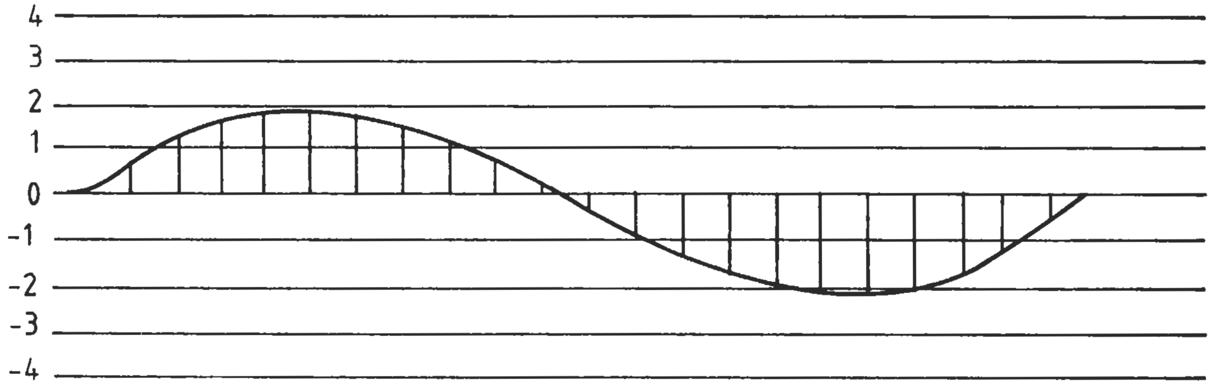

FIG 9.3 SAMPLE VALUES OF SIGNAL (SMALL AMPLITUDE)

19. Most speech is at low level and for speech signals in general, most of the information is contained in the waveform near the axis. Use of non-uniform quantisation gives the effects of small quantising steps for small input voltages and large steps for large input voltages (Fig 9.4).

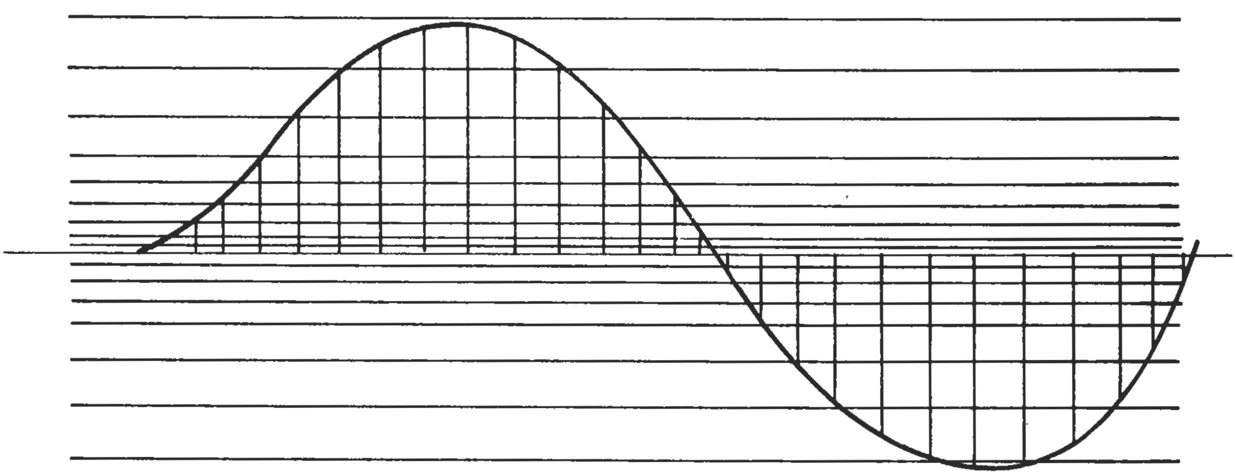

FIG 9.4 NON-UNIFORM QUANTISATION

20. This process, known as companding, improves the signal to noise ratio of the low amplitude samples at the expense of the high amplitude samples, but results in the signal to noise ratio remaining nearly independent of signal level over a wide range of input levels.

21. The non-linear quantising scale accepted by the CEPT is arranged so that in the positive (negative) half cycle, 128 quantum levels are arranged in 7 groups, each group having a number of equally spaced levels, the number and spacing being unique to a group.

22. From 0 V, group 1 contains 32 very narrow levels. Group 2 contains 16 levels, each of which is twice the width of Group 1 levels. For Groups 3-7, each contains 16 levels; in each group, the width of a level is twice that of the preceding level.

23. This scale, recommended by CCITT, is known as the A-law companding characteristic (Fig 9.5).

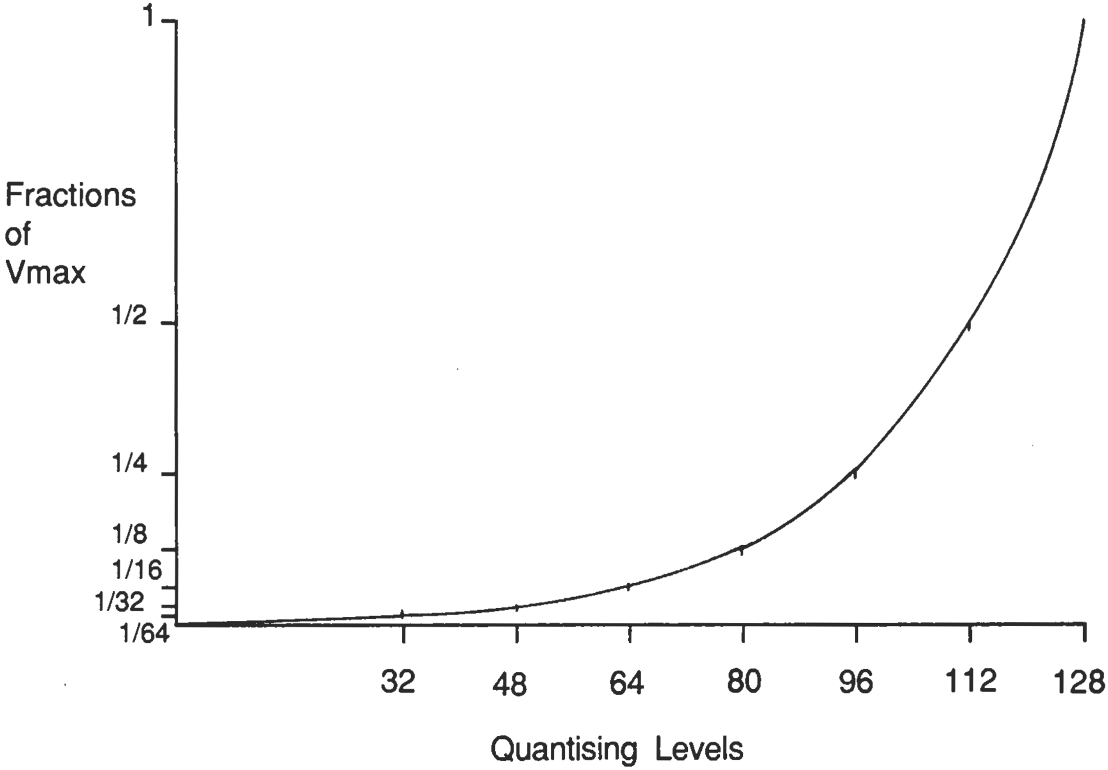

FIG 9.5 CCITT A-LAW COMPANDING CHARACTERISTICS

24. The USA has adopted the other recommendation of CCITT (the μ-law companding characteristic). This arrangement comprises 8 groups of 16 levels in each of the positive and negative directions, each group being twice the width of preceding groups.

<u>Sampling Rates</u>

25. Nyquists's Sampling Theorem indicates that it is necessary to use a sampling frequency of at least twice the highest frequency of the input. In practice, the sampling frequency is made slightly higher than this to simplify filter design. The highest frequency of commercial telephone speech is 3.4 kHz, the sampling frequency adopted for PCM is 8 kHz. With an 8 bit code transmitted for every sample this results in a line data rate of 8 × 8k bits per sec ie 64 Kbps.

Use of PCM in Practical Data Transfer Systems

26. CCITT has specified two distinct PCM systems for practical use:

(1) The 24-channel PCM system adopted by USA and Japan and

(2) The 30-channel PCM system standardised by the European Conference of Postal and Telecommunications Administrations (CEPT).

17. The 30-channel system has been developed over a period of time to:

(1) Be suitable for both long haul (national trunk, international) and short haul application and

(2) To cover the PCM requirements of future integrated digital networks (UK System X, ISDN), but with point-to-point admitted, and

(3) To give an expanded signalling capacity with emphasis on requirements for common channel signalling in integrated digital networks and stored programme control of switching, and

(4) To give redundancy in the system to facilitate transfer of information.

30-Channel PCM System

28. The system uses classical time division multiplexing for the 30 information (speech) channels. Each channel is pulse code modulated using 8-bit speech encoding with an 8 kHz sampling rate.

29. Channel samples are sent to line in frames, each frame containing one sample from each channel. Each frame contains 32 time slots designated TS0-TS31. Since channel rates and thus time-slot rates are at 64 Kbps,the line signalling rate is 32 × 64 Kbps ie 2.048 Mbps.

30. In a frame, TS0 is used for frame alignment, TS1-TS15 contain information from channels 1-15 and TS17-TS31 contain information from channels 16-30.

31. TS16 is used for signalling information for all 30 channels. Since TS16 contains only 8 bits, this signalling is sub multiplexed over a period of 16 frames (1 multiframe) (Fig 9.6). In Frame 0, TS16 is used for multiframe alignment, in Frames 1-15, 4 bits of each TS16 provide signalling for each of 30 channels. As each multiframe is repeated every 2 mSec, this corresponds to a signalling speed of 2 Kbps, more than sufficient for normal purposes.

BEARER SYSTEMS

32. To take advantage of the extensive telephone (speech) network throughout UK, data communications were originally made to look like speech. With the advent of a digital network, speech is processed to appear as data signals.

33. It is therefore necessary to have a good understanding of existing bearer systems and techniques viz analogue, digital and optical fibre.

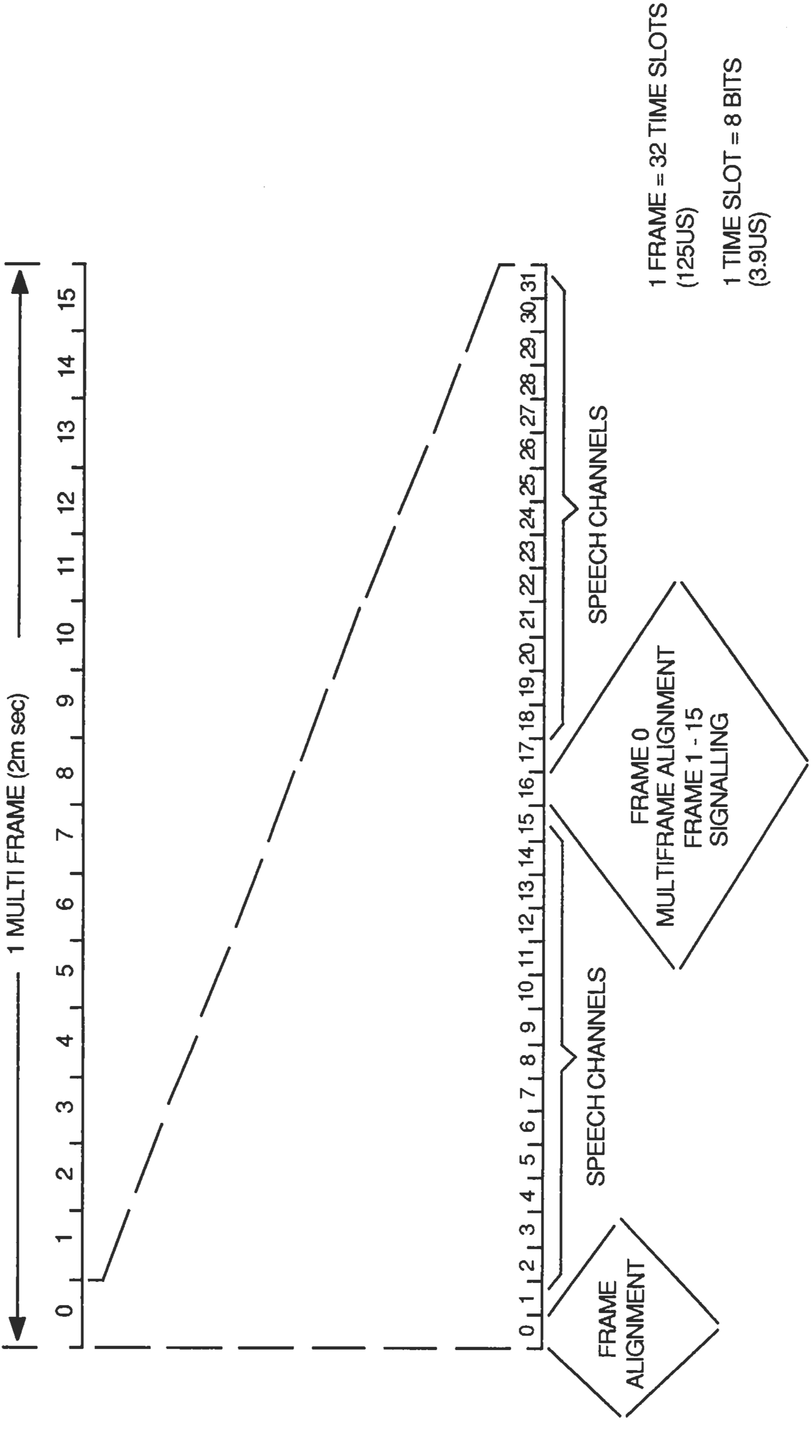

FIG 9.6 PCM FRAME AND MULTIFRAME FORMAT

Analogue

34. The existing analogue network standards for permissible national network transmission losses are specified in the CCITT Sending and Receiving Equivalents. The current figures are:

Sending Equivalent	21 dB
Receiving Equivalent	12 dB

measured at the international exchange with a maximum permissible transmission loss between speaker and listener of 36 dB.

35. Lines may be 2-wire, 4-wire or hybrid depending on requirements, and are made up of Local Distribution lines, the Junction Network and the Trunk Network.

36. Local Distribution lines are usually 2-wire, of average length 1 mile. The Junction Network, used to interconnect local exchanges, has lines, usually 2-wire, varying in length from 1-25 miles. The Trunk Network, used to interconnect trunk exchanges are long (over 25 miles) lines and, since permitted line loss is apportioned to various levels of the network on the basis that least used parts (ie small concentration of traffic) have greater loss and most used (ie large concentration of traffic) have small loss, trunk circuits are required to be low loss.

2-Wire Circuits

37. The simplest circuit is a pair of wires having attenuation of approximately 2 dB/mile at 1600 Hz, a severe constraint on permissible lengths of circuit. Attenuation various with frequency as in Fig 9.7.

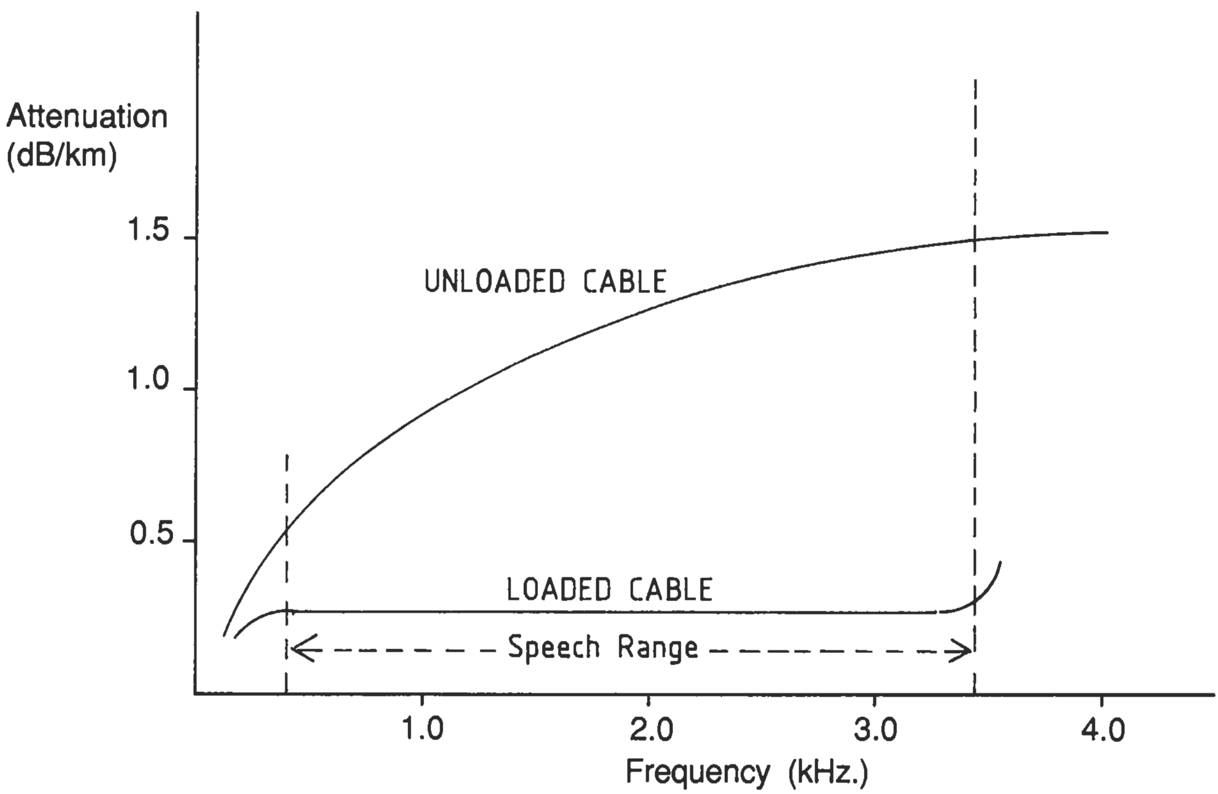

FIG 9.7 2-WIRE AUDIO CIRCUIT - EFFECT OF LOADING

38. Conditions may be improved over the significant frequency range (300 Hz - 3.4 kHz) by "loading" the cable. Loading coils, typically 88 mH, are inserted in the cable at regular intervals of approximately 2000 yds. Attenuation to cut off is significantly reduced allowing longer lengths of 2-wire circuit.

39. Due to sharp frequency cut off, phase distortion is introduced. Though speech transmission is not affected, it can be harmful to data. Additionally, propagation time is increased by a factor of 10 making it undesirable to use loaded cables for very long circuits.

4-Wire Circuits

40. This circuit is one which uses two separate 2-wire channels, one in each direction. Provided we use amplifiers in each leg, losses can be kept to a minimum. All trunk circuits (long haul) use 4-wire amplified.

4-Wire/2-Wire Termination (Hybrid)

41. To convert a 2-wire channel to a 4-wire channel, a hybrid is used, normally using two transformers as in Fig 9.8.

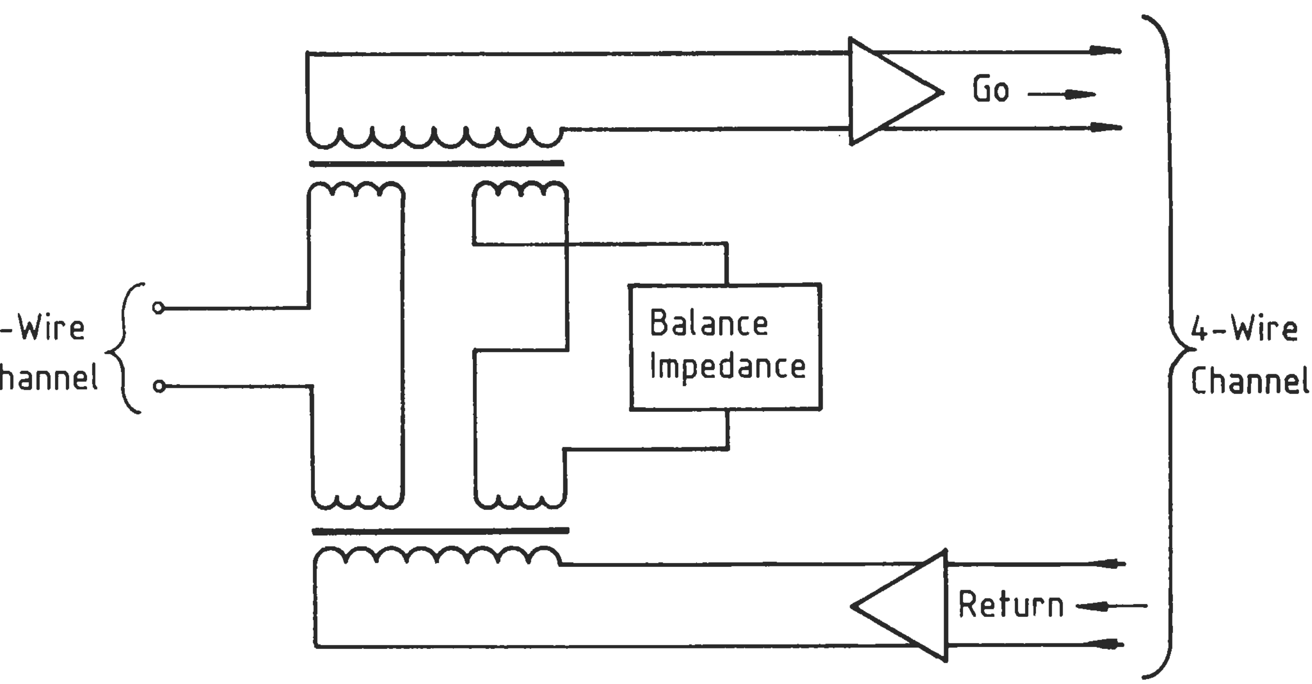

FIG 9.8 4-WIRE/2-WIRE HYBRID

42. Signal power arriving at the hybrid from the 2-wire channel is passed to each amplifier, but will only pass on the GO channel. Signal power arriving from the RETURN channel will be shared between balance impedance and the 2-wire channel. Provided impedances of 2-wire channel and balance impedance are equal, no signal passes to the GO channel. This method gives a 3 dB loss in each direction, thus the total hybrid loss between 2-wire points is 6 dB.

43. The performance is critically dependant on the equality of balance and line impedances. Thus to achieve correct operation it would be necessary to measure the frequency/impedance characteristic of each line and design an exact balance network. In practice, a 'compromise balance' is used - this being equal to the nominal impedance of the line (usually 600 ohm).

44. This leads to a small portion of the power received from one side of the 4-wire circuit being passed through the hybrid and retransmitted in the other direction. This 'balance return loss', unless prevented can be repeatedly amplified to cause 'singing' and instability.

Digital

45. For public telecommunication networks, the CCITT has standardised the Pulse Code Modulation (PCM) technique, utilising TDM with the signals from a number of baseband channel inputs being transmitted over a wideband transmission path. Each input channel occupies the whole of the bandwidth for a small proportion of the time, the TDM being a sampling technique.

46. All PCM systems are 4-wire. Each direction must be in synchronism, 'frame alignment' being achieved by a unique pattern of bits.

47. At the receiving end, the incoming bit stream is searched for this code pattern, which, once found, gives the framing information.

48. If the code pattern disappears (loss of synchronism) then communication must be interrupted to allow frame alignment to be re-established.

49. At the receiving end, the time slots in a frame are derived on a time assigned basis from the received frame, thus time slot synchronism is assured as a consequence of frame alignment.

50. An example of a practical system is the 30-channel PCM system described in para 28-31.

Digital Line Circuits

51. Since digital signals cannot be transmitted satisfactorily on the cabling provided for the existing analogue network, cables must be specially designed for digital use.

52. Once such development is the Transverse Screen Cable, containing two groups of cable pairs separated by a metallic screen (Fig 9.9).

53. Each group of cable pairs carries the digital signals in one direction and the metallic screen provides an electro-magnetic barrier between them to reduce interference and cross talk.

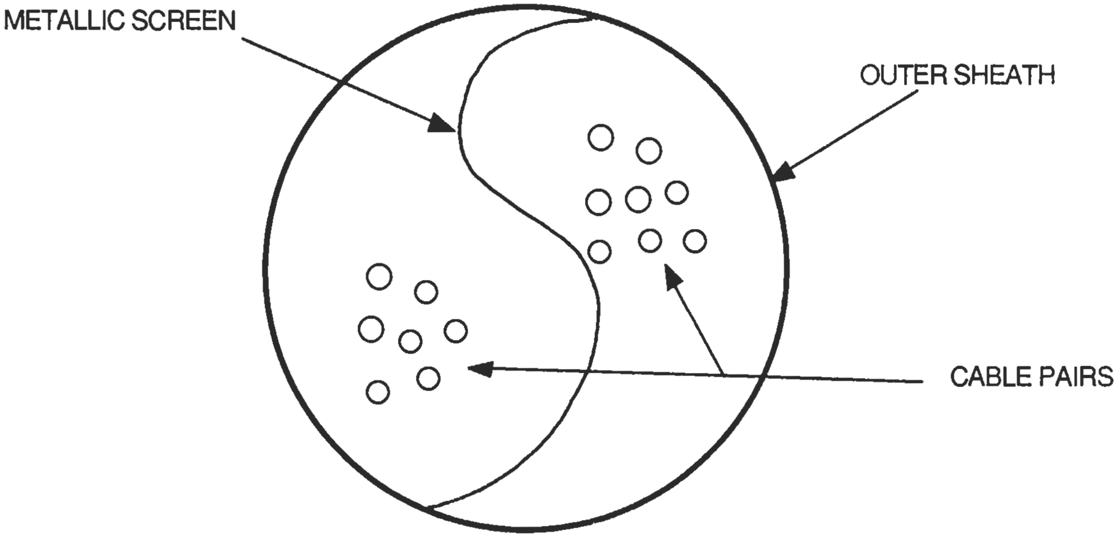

FIG 9.9 TRANSVERSE SCREEN CABLE

54. The digital signals passing along the cable pairs will become distorted and accordingly regenerators are inserted in the line at approximately 2 km intervals to restore the signal. Regeneration uses amplification only to make the signal identifiable, it actually involves creating a new signal under the control of the distorted input signal (Fig 9.10).

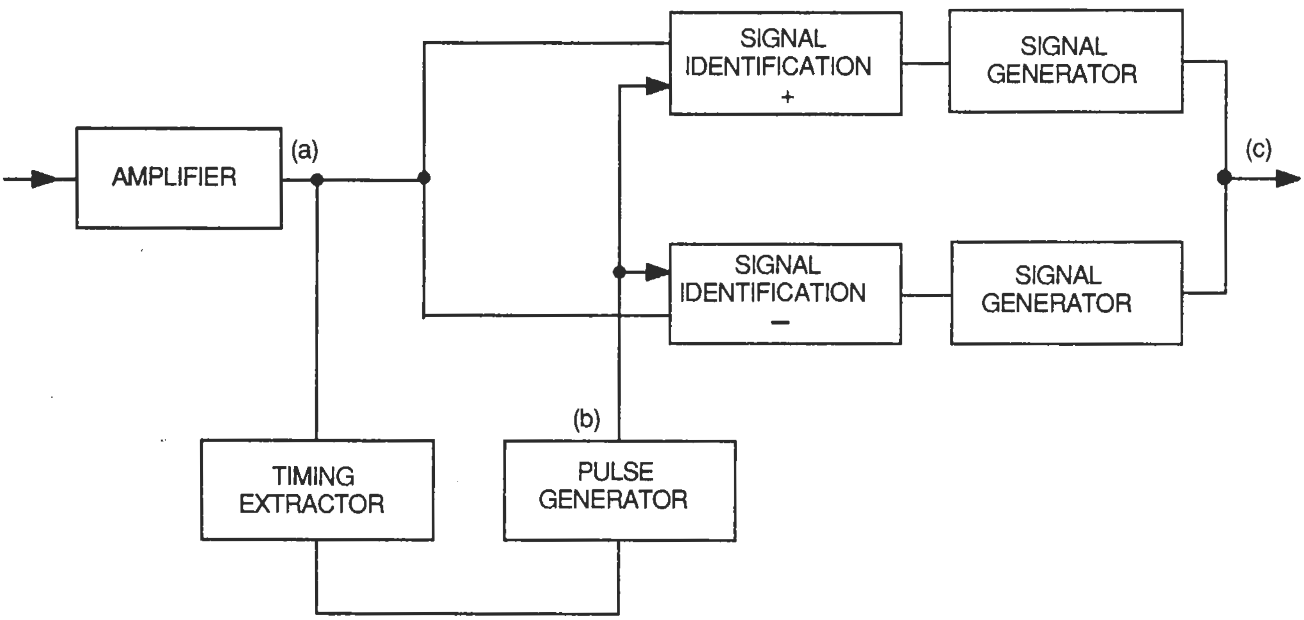

FIG 9.10 REGENERATOR BLOCK DIAGRAM

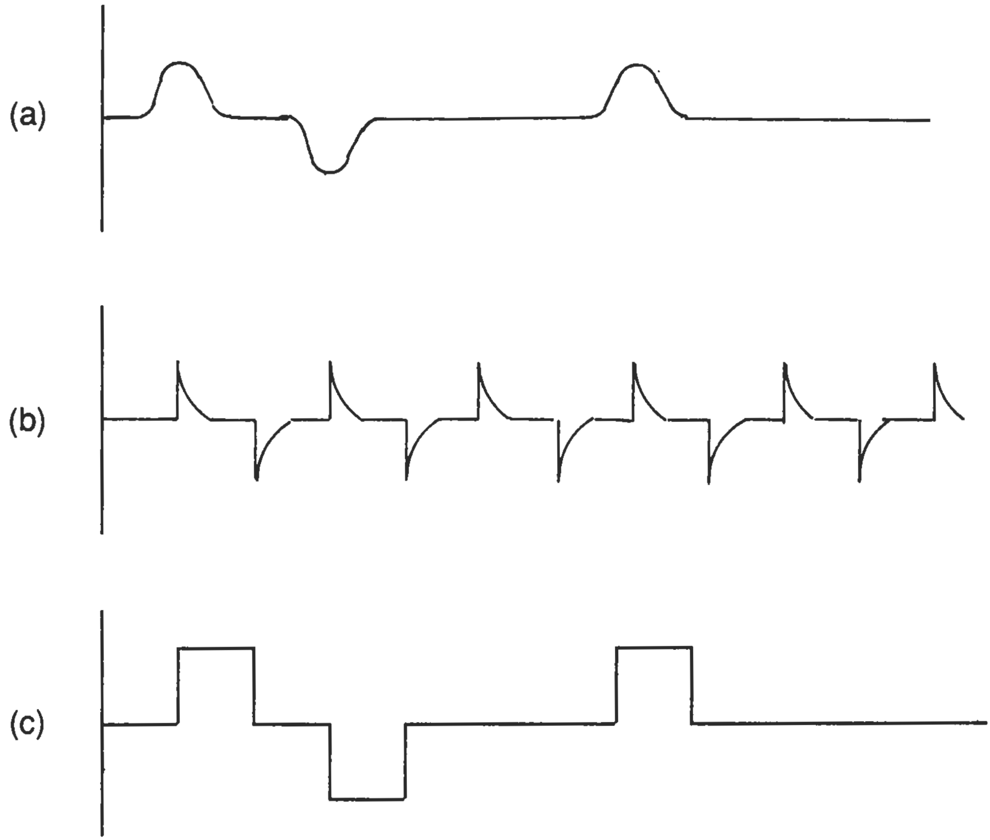

FIG 9.11 REGENERATOR SIGNALS

55. The distorted input signal is amplified (Fig 9.11a), timing information is extracted from which the pulse generator produces a series of timing signals (Fig 9.11b) to be used for signal identification.

56. Digital signals from the amplifier are also passed to the signal identification stage, where, provided the amplitude exceeds a pre-determined threshold value and coincides with a timing signal, a new signal with the correct parameters will be created in the signal generator (Fig 9.11c).

57. The output of the regenerator is thus a "clean" noise-free signal identical to that transmitted (provided that any noise/interference signal has an amplitude below that set in the Signal Identification stage.

58. Other suitable digital lines may be provided by use of microwave and optical fibre techniques.

PUBLIC SWITCHED TELEPHONE NETWORK (PSTN)

59. The existing analogue PSTN, though programmed to be replaced by an integrated digital network (System X) will continue in service for some time and much low speed data communication will take place on the PSTN or on leased analogue telephone type plant.

60. The analogue PSTN was designed for commercial quality speech and thus a 3 kHz bandwidth through the network was sufficient, together with 2-wire circuits to connect subscribers to the network and to switch local calls. 4-wire circuits were only used where amplification was required. Higher level multiplexing systems using FDM provided multiple channels on cable or radio links for the long haul trunk network.

61. The nationwide telephone network provided a convenient vehicle for data transmission and modems were developed to enable digital signals from the computer to be converted to analogue form for transmission over the PSTN.

62. BT-provided modems allow low speed data only, 300 bps duplex or 1200 bps half-duplex on PSTN dial-up connections with 2-wire lines. Up to 9.6 Kbps full duplex is possible over 4-wire equalised leased lines.

63. As a data network, the PSTN has limitations:

(1) Call set up on the analogue based PSTN takes too long for most real-time data applications.

(2) The requirements to access remote data bases on-line cannot be satisfied by the analogue PSTN due to call set-up time.

(3) The PSTN tariff is duration oriented, whereas a volume based tariff is fairer for data traffic.

64. To overcome the limitations imposed by the PSTN, many dedicated data networks have been introduced. Unlike the PSTN which is a circuit switched network, most public data networks introduced throughout the world operate in the packet switched mode.

SYSTEM X INTEGRATED DIGITAL NETWORK (IDN)

65. The BT replacement for the analogue PSTN, already under way, is programmed for completion about 2000. Eventually this will function as part of an Integrated Services Digital Network (ISDN) capable of taking all data communication at customers option.

66. The complete UK trunk network digital transmission medium will be by optical fibre, showing a cost saving over PCM transmission over coaxial cable and radio. Eventually optical fibre will be used on junction networks and even on some local lines.

LONG DISTANCE DATA TRANSMISSION

67. Should a data transmission system consist only of short lengths of correctly matched coaxial cable, digital transmission would pose no problems since the output waveform would be virtually identical to the input waveform.

68. When data is sent over networks which have not been designed for data transmission, such as analogue Public Switched Telephone Network (PSTN) the transmission path is not ideal and suffers from:

(1) Restricted Bandwith. Telephony bandwidth is usually 300-3400 Hz, acceptable for commercial speech, tolerable for low speed data, but not acceptable for high speed data, since frequency components of a data signal beyond some upper limit are severely attenuated.

(2) No DC Response. Frequency components below some limit are greatly attenuated with no transmission at all at zero frequency.

(3) Gain-Frequency and Phase-Frequency Distortion. The gain of a channel may vary between upper and lower cut-off frequencies. This causes the output waveform to differ from the input because frequency components have amplitudes changed by varying amounts. Similarly if phase shift versus frequency characteristic of the channel is not linear (ie transmission delay varies with frequency) the frequency components of the signal are delayed by differing amounts.

(4) Echoes. Electrical discontinuities in the transmission path causes reflection of the transmitted signal which appear as signal echoes at the receiver.

(5) Noise. Any system with amplifiers will introduce random noise which adds to the signal waveform. This noise has zero dc value and an approximately Gaussian amplitude distribution. Since for a given signal-to-noise ratio, there is a probability that the signal and noise will appear closer to the value of an incorrect symbol than the value of the correct one, a proportion of the symbols received will be in error.
This effect diminishes rapidly with increase in signal-to-noise ratio.

(6) Burst Noise. Any transmission system will suffer from occasional bursts of high amplitude noise. The majority of symbols occurring during such bursts are likely to be corrupted (short burst error rate).

69. Other types of degradation exist and can give trouble in special circumstances. These include non linear transmission characteristics caused by eg amplifier overload, frequency offsets in FDM systems and time varying channel characteristics as in HF radio and underwater data transmission.

70. Because of redundancy, the above impairments can be tolerated for the economic transmission of commercial speech.