DATA COMMUNICATIONS.

Ref: Computer Communications by Robert Cole, MacMillan Computer Science, 1982.

1 Transmission in Wires.

1.1 Information Channel Theory.

3 components of communication : sender, receiver & medium

Waves are used to carry info through most media, esp. in real-time communication.

3 characteristics of wave: $\underbrace{\text{amplitude, frequency}}_{\text{absolute}}$ & $\underbrace{\text{phase}}_{\text{relative}}$.

Any two simple waves having same amp, freq & with zero phase difference are equal & indistinguishable.

Data can be coded & transmitted in waves by changing one or more of the components, eg speech – freq determines tone, amp - volume.

The frequency of a square wave carrying data is the maximum possible number of changes (ie 101010....)

Recall square waves can be built up out of simple sine & cosine waves. As the frequency of the constituent waves increase the amplitude decreases, so the higher frequency waves are less important.

All transmission media, senders & receivers are limited in the frequencies they can manage, specified by their frequency range. The difference between the extremes is the bandwidth

The telephone has a bandwidth of 3100 Hz (300 Hz - 3400 Hz)

Thus the transmission medium / equipment used must have sufficient bandwidth to allow sufficient constituent waves through to enable the receiver to recognise the signals.

3 factors affect the rate of data transfer:

- the encoding method used
- the number of distinct symbols which may be sent
- the channel bandwidth.

If the data consists of P different symbols, then the info content is $I = \log_2 P$ bits.

The capacity of the channel is the maximum rate at which data may pass through the channel, ie, the number of distinct symbols per second.
If T is the minimum time to transmit one symbol, then $C = 1/T$ symbols/sec is the capacity
(ie $C = (\log_2 P) \frac{1}{T}$ bits/sec.)

It can be shown that the bandwidth of a channel is $1/(2T)$ Hz, so $C = 2\omega \log_2 P$ bits/sec (ω - bandwidth)
(Only true for a "perfect" channel).

CLAUDE SHANNONS EQUN: $C = \omega \log_2 (1 + S/N)$ bits/sec
(S/N - signal to noise ratio db)

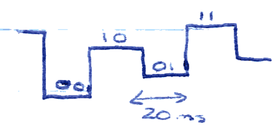
1.2. CHANNEL ORGANISATION

Problems with parallel transmission:
- cost (each bit needs own channel)
- skew (increases with distance)

Within a computer system, these restrictions are less serious as speed takes priority over cost & skew is not a problem. With increasing distance skew becomes very noticeable.

Serial transmission has the disadvantage of being much slower, but eliminates the other two problems.

Information rate: bits per second bps.
Signalling rate: baud

Eg four level code: 

Each symbol takes 20ms ie 50 baud.
2 bits per symbol ie 100 bps.

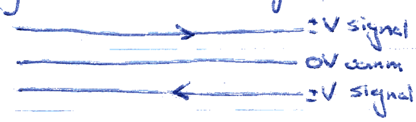
bps = baud iff 1 bit / symbol.

simplex mode – only one sender & 1 receiver
half-duplex mode – both sides can send & receive but not simultaneously
full duplex (2 channels) – both sides can send & receive simultaneously

V24 / RS232C (USA) STANDARD – voltage amplitude system.

Mark 'O' is +V where V ∈ (3..24) Volts, typically 12V

Space '1' is −V   0V is 'break'



3-Wire V24 Full Duplex Connection

1.3 Character Framing.

Most DC involves transferring characters; these are thus the basic unit of transfer. A receiver must be able to recognise a serial stream of bits as characters ie detect the start/end of characters. There are two character framing techniques for recognising and separating characters from the serial bit stream:

- asynchronous framing - no common clock, send/receive at any time - slow (eg unbuffered terminals)
- synchronous framing - common clock, send/receive constantly at a specified rate - fast.

Asynchronous Framing

No characters to transmit ⇒ hold line at −V (Space)

When a character is to be sent, output goes to +V for 1 bit (Mark) (the start bit), followed by the character and a stop bit −V which is held till next character.

The first −V to +V transition (start bit) starts the receiver's local clock. The stop bit is used to return the line to −V as well as to provide a small amount of processing time for slow devices (eg unbuffered printers). The devices must agree on the number of bits in a character as well as the transmission speed. If in doubt about stop bits, send 2 and expect to receive 1.

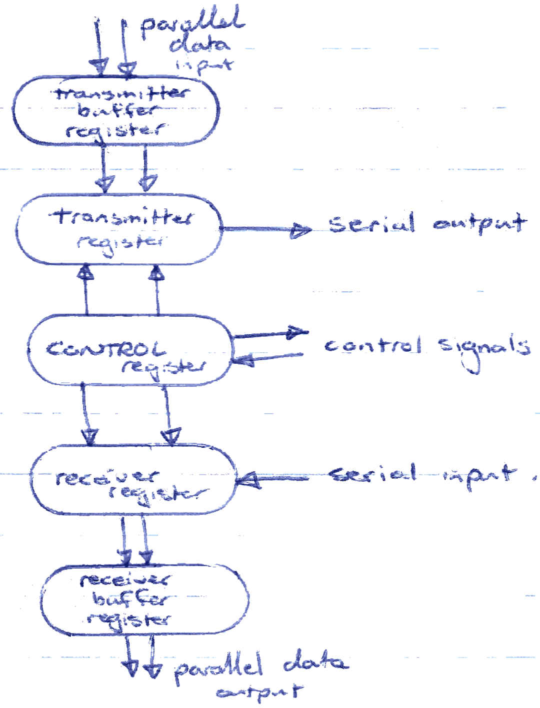
Synchronous Framing.

Characters are transmitted constantly, with sender and receiver having clocks synchronised. A code will be sent to indicate the start of transmission (eg ASCII SYN 00010110) which is bit-shifted by the receiver until it is recognised.

Due to drift the channel clocks must be resynchronised frequently; achieved by splitting data into blocks with a synchronise gap in between.

1.4 SEND & RECEIVE EQUIPMENT.

Asynchronous framing done with a UART (Universal Async. Receiver/Transmitter) which performs serial to parallel conversion, frame generation and checking; can be programmed for 5, 6, 7, 8 bit characters, parity (even, odd or none) and any line speed using an external clock.
Synchronous transfers are done with a a similar device the USRT.



UART BLOCK DIAGRAM.

The UART uses a clock rate of 16 times the bit rate. Immediately a start bit is received (−V to +V transition) a counter is set to 8 and counted down every clock pulse. When it reaches zero if the voltage is no longer +V the logic is reset and an error bit set on the interface. Otherwise the counter continues modulo 16, the value of each bit being found by sampling each time the clock reaches zero. This can be improved by sampling several times and averaging the result and comparing with a threshold voltage (particularly useful in noisy environments). If the stop bit is not found when expected, a frame error occurs. Some machines use frame errors to match line speeds ('automatic bit rate detection').

1.5 COMPARISON OF CHARACTER FRAMING TECHNIQUES.

Async useful for intermittent transfers - results in about 20% redundancy however. Sync. is useful for transferring large amounts of information, with much higher efficiency due to far less resynchronisation than async.

2. The Telephone Network as a Medium.

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2.1 The Telephone System.

The telephone system is important as the PSTN is well-established and extensive (and a protected monopoly by law!). Main problem is that it is designed for speech - narrow frequency range 300 Hz to 3400 Hz which is not continuous as some frequencies are used to operate equipment at exchanges; also no DC current may flow through an exchange. Three solutions

increasing cost & transfer rates

- digital / audio conversion equipment (MODEMs - available 300 bps full dup, 600/1200 half dup; bandwidth divided into two for full duplex)
- private four-wire connections to local exchanges, cutting out some information - destroying equipment (lose dialling ability) private circuit
- direct high-bandwidth links

Telephone exchanges are linked to one another through frequency - division multiplexed wideband channels. Packet switching networks are now becomming more commonly used as well (see later).

2.2 Modulation Techniques.

Modulation, very simply, involves superimposing the information signal onto a simple carrier signal in such a way that one, or more, components (amplitude, frequency, phase) of the carrier are modified to carry the information.

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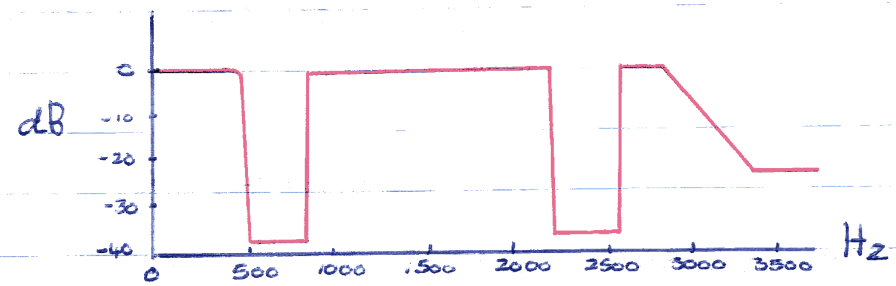
Frequency Shift Keying (FSK)

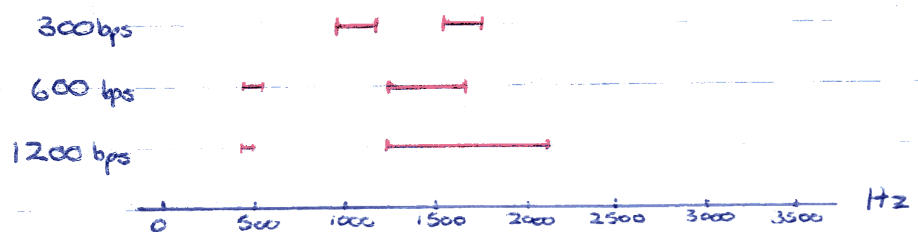
An FSK modem converts voltage signals into frequencies (slow)
Problem is choice of frequencies, within constraints:

- frequencies used should be measurably different
- to detect the frequency at least half a wave must be transmitted ⇒ lowest frequency used must be greater than the baud rate of the data signal.
- certain frequency ranges on the telephone circuit are reserved for use by the switching equipment (internationally CCITT agreed)
- higher frequencies on a telephone circuit are attenuated



PSTN FREQUENCY RESPONSE (above)

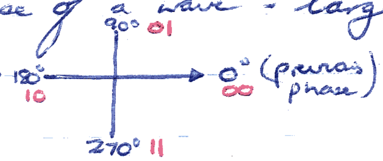
MODEM FREQUENCIES (below)



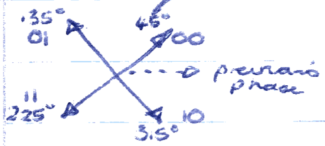
FSK is important, but is limited by noise and the low frequency range of the PSTN.

p31 PHASE MODULATION

The data is coded into the phase of a wave - large phase changes Eg differential phase modulation



. A continuous frequency simple wave is used so that only the phase component changes. To detect the phase changes a complete wave is used, representing 2 bits of information. Eg frequency of 1200 Hz gives transmission rate 2400 bps. Problem: if a stream of zeros is transmitted no phase change occurs and receiver & transmitter can drift. Newer modems thus use an alternative encoding:



Quadrature Amplitude Modulation.

Combination of phase changes with different signal amplitudes to improve the bits/baud ratio. Used on faster modems above 2000 bps to increase bps without increasing frequency. However amplitude is very susceptible to noise.

2.3 The Modem

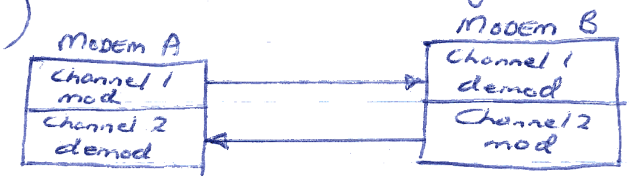
A Timesharing Terminal Modem (Cheap popular slow)

A timesharing terminal has a low data rate and requires a full duplex connection. The PSTN bandwidth is divided into two to provide two channels using frequency shift keying on frequencies (CCITT, not USA).

Channel 1:	binary 0	1180 Hz
	binary 1	980 Hz
Channel 2:	binary 0	1850 Hz
	binary 1	1650 Hz

When a MODEM is reset/switched on, it is set to transmit on channel 1 and receive on channel 2. The MODEM receiving a call changes this arrangement. The caller MODEM can be replaced by an acoustic coupler (providing audio I/O rather than direct electrical)

A calls B:



Higher Speed Asynchronous MODEM

A second type of MODEM is used to provide a higher speed in one direction but much lower speed in the other direction. Up to typically 600 bps can be transmitted via PSTN in one direction while the return channel has only 75 bps - this is the supervisor channel, and is not expected to carry much data. If high speed connections are needed in both directions two modems with a private 4-wire circuit. The MODEMs use FSK with the following frequencies:

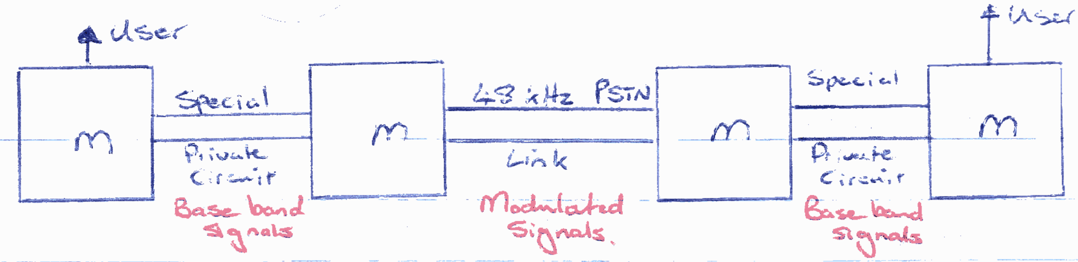
	0	1
Up to 600 bps	1700 Hz	1300 Hz
Up to 1200 bps (private connection)	2100 Hz	1300 Hz
Supervisor channel	450 Hz	390 Hz

Fast Synchronous Modem

Fast MODEMs using differential phase encoding can be used to transfer at 2400 bps (noise permitting) on PSTN, or 4800/9600 bps on private circuits. Connection pattern is as for the async MODEM above, i.e. simplex or half-duplex on PSTN, with full duplex requiring a private four-wire circuit.

Very High Speed Modems

These are for use in inter-computer communications, and allow for data rates of between 40.8K bps and 50K bps over 48 KHz wide-band circuits. The connection involves using four modems and is used for connecting packet exchanges in a wide-area packet switching network.



FOUR-MODEM ARRANGEMENT

The main uses of MODEMS are:

- connecting teletypes or slow VDU's (110/300 bps) to any computer in full duplex via PSTN.
- connecting a faster terminal (600/1200 bps) in half-duplex via PSTN or full-duplex via a private circuit.
- provide a synchronous connection (1200/2400, 4800, 9600 bps) in full-duplex via a private circuit.
- very high speed transmission using base-band signalling on a special circuit.

2.4 DIGITAL DATA NETWORK

We now examine a new development, the conversion of the PSTN to digital transmission using pulse code modulation. This has the advantages:

- noise effect is reduced
- digital multiplexing techniques can be used to make more efficient use of the trunk network.
- the switching can be controlled using information embedded in the encoded signal.
- cheap VLSI digital technology can be used, allowing considerable intelligence to be built into the network.
- digital information can be transmitted directly.

Eg System X in UK means that soon every telephone in the UK will be connected by a 64 kHz circuit, opening up tremendous possibilities for personal information services.

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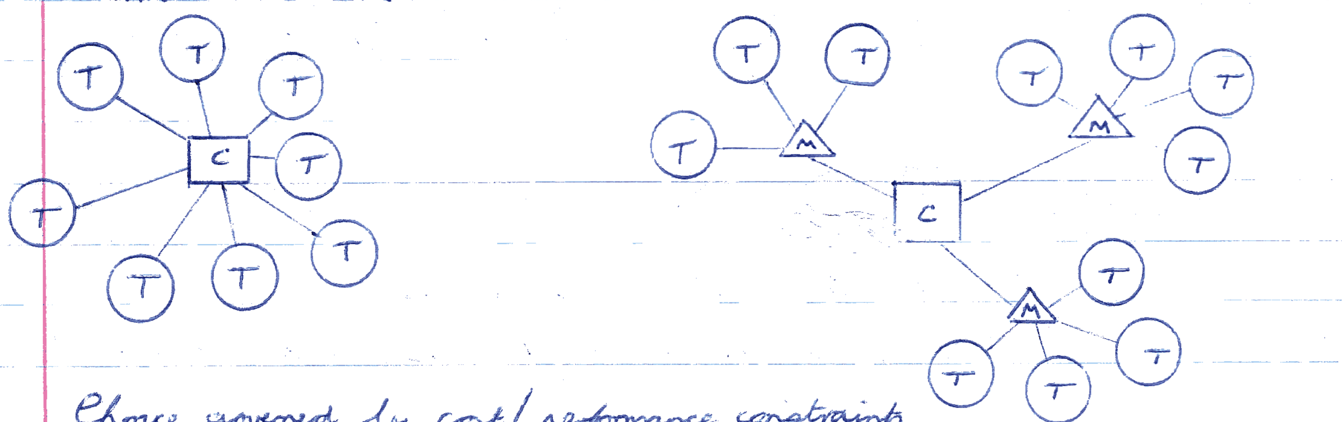
2.5 STANDARDS

The PTT's (Postal, Telegraph & Telephone authority) work through CCITT (English: International Consultative Committee for Telegraphs and Telephones) part of UN. All modems discussed earlier conform to CCITT standards. (also V24)
CCITT standard interface for digital switched network is X21 (X21(bis) is an analogue version). The packet switched network access interface X25 is defined to use X21 as its lowest level.

3 CHARACTER TERMINAL NETWORKS.

We now examine network organisation for interactive asynchronous terminals. Two basic arrangements:

STAR NETWORK REMOTE MULTIPLEXER NETWORK.



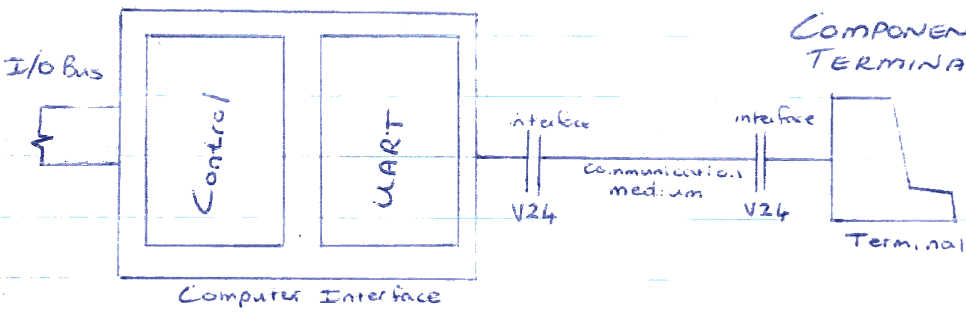
Choice governed by cost/performance constraints
Major criterion is response time ie highest information rate at lowest cost.

3.1 CHARACTER CODES.

ASCII (American Standards Committee for Information Interchange)
EBCDIC (Extended Binary Coded Decimal Interchange Code) (IBM)
IA5 (International Alphabet #5) (CCITT / ISO) same as ASCII
but allows local variations for currency signs, etc.

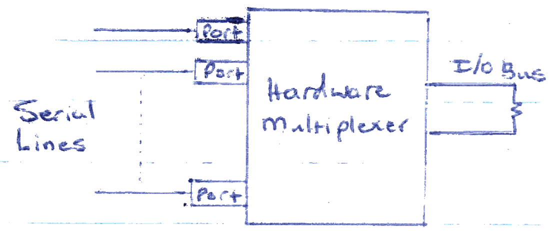
ASCII is 7 bits, no parity recommendation ~35 control characters
and ~93 characters
IA5 is ASCII with even parity and local variation
EBCDIC 8 bit invented & used by IBM

3.2 STAR TOPOLOGY



COMPONENTS OF A TERMINAL INTERFACE

The computer interface uses a local multiplexer, which has one UART for each terminal line



LOCAL TERMINAL MULTIPLEXER

Lines are typically polled in a round robin sequence using a scanner register (modulo n counter, n typically 16 terminals)

33 ## Remote Multiplexers.

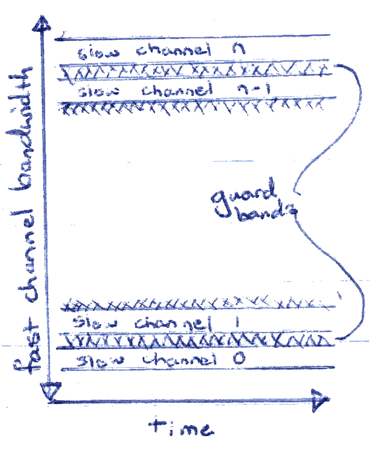
While star networks are ideal for small in-house arrangements, longer distance communication becomes expensive due to the number of transmission connections needed. It is often far cheaper to buy a faster modem and multiplex several terminals through it at the remote site, as the cost of renting a single high speed PSTN line is cheaper than many slow lines

p57 ## Remote Multiplexing Techniques.

Multiplexing is when input and output capacities are the same regardless of channel usage. If the input capacities exceed the output capacity (although not necessarily the average input capacity - typically used for terminals which are idle much of the time) then we call this concentration. Concentrators thus require buffers

Two ways of dividing frequency of the fast channel
- Frequency division multiplexing (FDM) (not used by computer equipment but by PSTN)
- Time division multiplexing (TDM)

Frequency Division Multiplexing



- each slow channel has a permanently allocated (narrow) channel on the fast channel
- if the slow channel is not being used the corresponding narrow channel is wasted
- channels have to be modulated up to the required frequency slots, and demodulated at the other end.

Time Division Multiplexing

Easier & more efficient than FDM. Time axis divided into slots (usually one character, which due to using whole bandwidth is very short). Each terminal has a slot ie if there are n terminals every n^{th} slot contains a character from the same terminal. Sender and receiver synchronise slots by an internal mechanism, and then transmit. Advantage over FDM is elimination of guard band wastage, although idle channels still cause wastage, dealt with using more sophisticated TDM techniques.

Intelligent or Statistical Time Division Multiplexing.

Slots are not allocated on a round robin basis, but as first-come first-serve. The time slot is lengthened to take extra bits identifying the channel. Buffering may also be necessary. These multiplexors are usually microprocessor controlled.

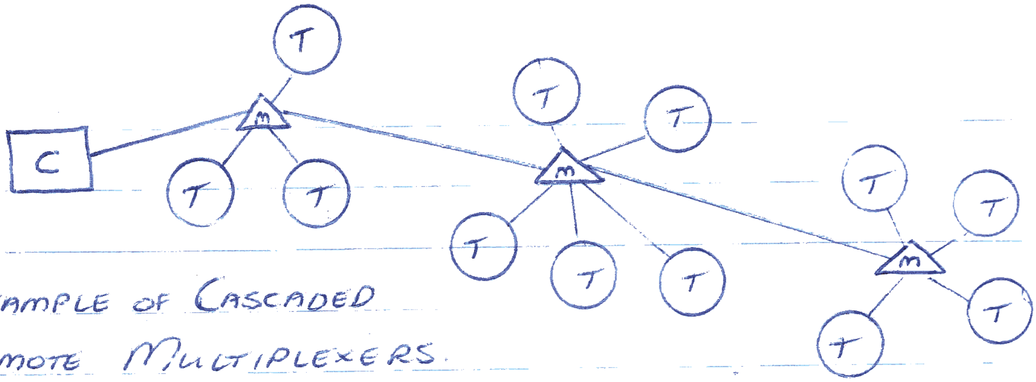
2.4 Multiplexer Networks.

Many computers that have a terminal network use a front-end processor (FEP*) to handle the network and remove much of the character processing overhead from the main computer system. Advantages of FEP:

- it can handle all character echo
- it can edit lines before continuing (eg rubouts, backspaces)
- it can indicate the system status when the main computer is down
- it can perform network monitoring and performance analysis.

With FEP's complex networks can be designed using cascaded multiplexers for remote terminals, and star arrangements for local terminals. Cascaded multiplexers are used to save line charge by connecting (usually intelligent) multiplexers in series.

* typically a minicomputer



Example of Cascaded Remote Multiplexers.

3.5 Designing an Asynchronous Terminal Network.

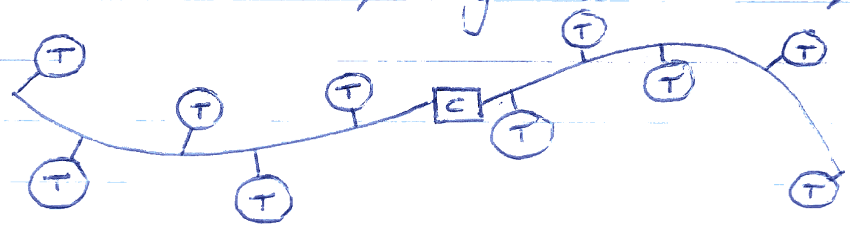
Usual constraints are the number and location of the terminals and the location of the computer. Local terminals are best attached using a star topology; remote terminals depend on many factors such as the cost and availability of PSTN lines, multiplexers, etc.

4 Simple Message-Based Techniques

We now examine more sophisticated communication techniques based on messages, specifically 'page-mode' VDU's and remote job entry stations.

4.1 Multipoint (Multidrop) Lines

The objective here is to reduce the cost of the transmission medium by sharing it between terminals. Each terminal requires a terminator (MODEM if medium is PSTN). The computer has a controller responsible for polling; two main methods are hub polling and roll call polling.



Typical Multipoint Line Network.

Messages to terminals can be sent the moment the channel is free; messages from terminals require polling and slot allocation. Separate lines may be used for each direction. As transmissions must be buffered at times, and also decoded to determine the appropriate terminal, characters are an unsatisfactory transmission quanta. Only messages are exchanged, so the ideal medium is a synchronous character frame channel.

Roll Call Polling

Each terminal is polled individually with a poll command and terminal address message. Each terminal inspects the message to see if it contains its address, and if so replies either with a NAK if it has no data to transmit, or ACK followed by the data. The controller may use RAM to hold a poll list of addresses; terminals requiring higher throughput could then have their addresses appearing more than once in the list.

Hub Polling

The terminals are imagined to be on the hub of a wheel with the polling message being passed round the hub by the terminals themselves (except for first/last terminals connected to controller). Three channels are usually used - one each for input and output (parallel connect), and one for polling (serial connect).

Advantages & Disadvantages

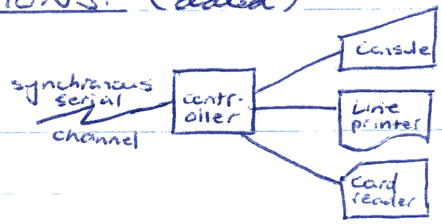
- Response time - roll-call more flexible due to alterable poll list.
- Efficiency - hub polling more efficient as it release the I/O channels from the polling process
- Flexibility - roll-call far better; hub polling is problematic to deal with terminal breakdowns or switch offs.
- Hardware

4.2 Applications

Not very useful in a general interactive environment (where messages will typically be very short). Very useful for transaction processing (eg airline bookings)

4·3 Remote Job Entry Stations (dated)

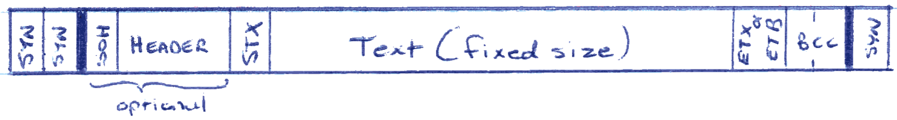
File based transfer typically



4·4 A Simple Message Protocol.

Most common message protocol (IBM) for multipoint and centralised message systems is Binary Synchronous Communication (BSC) Several variations exist as the definition is not rigorous.

BSC MESSAGE FORMAT



CONTROL CHARACTERS:

SOH – Start of Header (ie all text following till next control character is header)

STX – Start of Text

ETX – End of Text (Message)

ETB – End of Transmission Block (ie message incomplete, more to follow).

BCC – Block character check (typically parity or cyclic redundancy)

To establish a connection, sender sends a message having the single character ENQ in the text field. The receiver should reply with a similar ACK0 message. Transmission may then occur. After comparing BCC's, if the message was received correctly another ACK is sent, else it is discarded and a NAK sent. To end the connection an EOT message is sent. Time-outs may occur if a reply is not received for a period.

4.5 DISTRIBUTED COMPUTING.

Distributed computing requires more complex protocols; minicomputers are thus typically used as controllers

5. ERRORS, ERROR AND FLOW CONTROL.

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5.1 THE NATURE OF ERRORS.

Major cause of error is noise (big advantage of fibre optics). Shannon's equation including signal-to-noise ratio (p2); however this is based on averages, and may vary considerably locally. Normally a sequential group of bits become changed due to a burst of noise (called an error burst). There is a trade-off between the probability that the received data is correct, and the efficiency with which it is transferred.

5.2 ERROR CONTROL. (requires coding extra information into bit patterns)

3 main methods:

- echo checking
- forward error correction (FEC)
- Automatic Repeat Request (ARQ)

HAMMING DISTANCE. - the number of bits in respective positions that are different eg ascii A = 01000001 } HD = 2.
B = 01000010

A and C have a Hamming distance of 1, so an A is more likely to be changed into a C than a B.

5.3 ECHO CHECKING

The data is echoed back to the sender for checking. A full duplex connection with ample capacity is needed. Used for asynchronous interactive terminal-to-computer character transmissions, but otherwise very inefficient.

5.4 Forward Error Correction (FEC)

Extra bits contain information used by the receiver to detect and correct errors. Different types exist to cater for eg long bursts of noise, or many short bursts. As an example, we consider:

Hamming Code.

Check bits placed in "power of 2" positions, eg for 4-bit data

Bit Position	7	6	5	4	3	2	1
Bit	D_4	D_3	D_2	C_3	D_1	C_2	C_1

Data bits are covered by the check bits whose positions add up to the position of the data bit. Thus D_1 is covered by C_1 and C_2, while D_4 is covered by C_3, C_2 & C_1.

Thus:

$$\left.\begin{array}{l} D_1 \text{ covered by } C_1, C_2 \\ D_2 \text{ covered by } C_1, C_3 \\ D_3 \text{ covered by } C_2, C_3 \\ D_4 \text{ covered by } C_1, C_2, C_3 \end{array}\right\} \Rightarrow \left\{\begin{array}{l} C_1 \text{ covers } D_1, D_2, D_4 \\ C_2 \text{ covers } D_1, D_3, D_4 \\ C_3 \text{ covers } D_2, D_3, D_4 \end{array}\right.$$

So we set

exclusive-or

$$C_1 = D_1 \oplus D_2 \oplus D_4$$
$$C_2 = D_1 \oplus D_3 \oplus D_4$$
$$C_3 = D_2 \oplus D_3 \oplus D_4$$

When a code is received, it may be checked in two ways:

- recalculating check bits
- looking up the code in a table, if there are not too many bits.

Note that this works only if at most one bit is corrupted. Furthermore, out of 128 possible bit patterns only 16 can be used.

Despite these inefficiencies, FEC is very useful particularly if only simplex or half-duplex transmission is possible.

Also useful for long distance transmission (eg satellite communication). If the noise characteristics of the channel are known then more complex FEC techniques can be used to improve efficiency.

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5.5 Automatic Repeat Request (ARQ)

Used for block- and message-based transmission. A check sum is transmitted with the packet which is then recalculated by the receiver and compared. If these do not agree the packet is discarded, and a repeat-request sent to the sender.

~~log~~ physical unit — logical unit

Polynomial Codes (most efficient algorithm available 1982)

The message or block is treated as a general polynomial with the bits the coefficient of the polynomial terms, eg 1011 is x^3+x+1. Call the message to be transmitted $M(x)$, a polynomial in x. A generating polynomial $G(x)$ is used to calculate the check bits by dividing $M(x)$ by $G(x)$, so we get $\frac{M(x)}{G(x)} = \underbrace{R(x)}_{\text{remainder}} + \underbrace{D(x)}_{\text{result of division}}$ (and $R(x) < G(x)$.)

This calculation leaves $M(x)$ in a mess – it must be reconstituted by multiplication. To overcome this it is multiplied by x^n where n is the degree of $G(x)$.

This gives $\frac{M(x).x^n}{G(x)} = M(x) + R(x)$, far more useful.

Polynomial codes can detect the following types of errors

- all odd numbers of bits in error
- all error bursts with length less than the size of the remainder (usually 16 bits)
- all error burst that are longer than the remainder and are not exact multiples of the generator polynomial

The international standard is: $G(x) = x^{16} + x^{12} + x^5 + 1$.

IDLE RQ. (simplest ARQ strategy).

This is used in the BSC protocol with the addition of a sequence number, to detect duplicates as well as determining what part of the file the block is. Main problem is inefficient use of the channel, as the sender transmits a block and waits for an acknowledgement or NAK or timeout before transmitting again.

CONTINUOUS RQ

Blocks are transmitted continuously. Acknowledgements from the receiver contain the block sequence number. There are two schemes for handling retransmission when an error occurs:

- GO-BACK-N - the receiver sends a negative acknowledgement for block N. Transmission then returns to block N and continues sequentially from there (leading to possible redundant transmission of a few blocks).
- SELECTIVE RETRANSMISSION - only the NAK'ed block is retransmitted. Requires more storage than GO-BACK-N, but uses channel more efficiently.

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5.6 SEQUENCE NUMBERING AND ACKNOWLEDGEMENT

The use of blocks and ARQ presents three problems:

- the receiver must be able to maintain the correct sequence of data in the information unit
- any blocks already transmitted but not acknowledged must be stored by the sender in case they need

to be retransmitted.

- the receiver may have a limited amount of memory so it is possible for the sender to transmit faster than the receiver can handle the data.

These three problems introduce what is known as flow control. As flow control and error handling in ARQ-based protocols are combined, it is useful to look at such a technique at this point. The example of the combined flow and error control mechanism which will be shown here is based on GO-BACK-N ARQ, and is very similar to the mechanism used in X25.

Normally data is exchanged so acknowledgements can be piggy-backed onto data blocks being transmitted in the return direction. The header placed in front of each data block then contains two numbers: the sequence number of the block, and the sequence number of an acknowledged data block that was transmitted in the opposite direction. A special message is still needed for negative acknowledgement. Such a protocol is

P101 called a positive acknowledgement protocol.

The receiver must know the sequence number of the first data block. Each end keeps two variables, the sequence numbers of the next block it will/should transmit/receive ($T(s)$ / $R(s)$). (Both are usually 0 at start). Each transmitted data block contains two numbers, S ($= T(s)$), the sequence number of the block, and K ($= R(s)$), the positive acknowledgment of all data blocks $0 \ldots K-1$.

So sender: copies $S = T(s)$
increments $T(s)$
copies $K = R(s)$.

when receiver receives a block, it: checks for errors (if so discards, see later)
if no errors compare S with $R(s)$
if equal: increments $R(s)$
examines K to see which blocks can be removed from the retransmission buffer.

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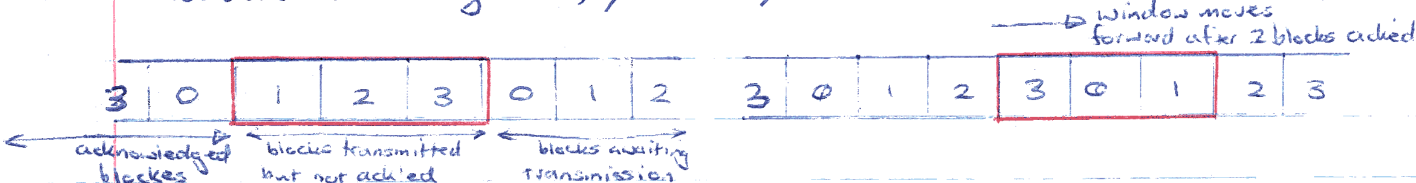
Example

COMPUTER A T(s)	COMPUTER A R(s)		COMPUTER B T(s)	COMPUTER B R(s)
0	0	initial synchronisation	0	0
1	0	data block S=0 K=0 →	0	0
1	0		S=R(s) ⇒ accepted	
2	0	data block S=1 K=0 →	0	1
2	0		S=R(s) ⇒ accepted	
2	0	← data block S=0 K=2	0	2
S=R(s) ⇒ accepted			1	2
K acks blocks 0 & 1			1	2
2	1	data block S=2 K=1 →	1	2
3	1	data block S=3 K=1 →	error ⇒ discarded	
3	1		S≠R(s) ⇒ error discovered.	
3	1	← negative ack K=2	K acks block 0	
retransmit from 2			1	2
2	1	data block S=2 K=1 →	1	2
3	1		S=T(s) ⇒ accepted	
			1	3

To save space, sequence numbers are usually modulo numbers (eg X25 is mod 8 ie 3 bits). To achieve this a window is used, the size of which is the maximum number of data blocks that can be outstanding (unacknowledged) at any time.
In selective retransmission, the window size w is related in the worst case to the modulus n by $n = 2w + 1$.
In GO-BACK-N retransmission it is $n = w + 1$. Thus X25 windows have size 7, for example.



Window size is normally set by the receiver to indicate the buffer space available for the connection.

If both ends know the window size, the situation is as follows
S must be in the range $R(s) - 1 \leq S < R(s) + W$

If $R(s) = S$ the data is accepted.

If $R(s) = S + 1$ the data block is a duplicate (special case, can only occur if the transmitted sequence is maintained)

If $R(s) < S < R(s) + W$ an error has occurred.

If S is outside the range a sequence number error has occurred and the two ends are unsynchronised. The protocol must allow for recovery from such an error.

The acknowledgement number K must be in the range $T(s) > K \geq T(s) - W$
If this is not so, recovery is necessary (see above).

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6 Computer Network Technology.

We now consider the technology involved in wide-area networks, broadcast networks, and private networks.

6.1 Wide-Area Networks.

Mainly 50kHz analogue or 64kHz digital dedicated circuits eg PSS, ARPAnet, etc. A fully interconnected network requires many connections, many of which will be idle much of the time. Switching is thus used to reduce the number of connections.

Circuit Switching

This is used on the PSTN to connect telephone subscribers. Each computer would have a private circuit to a local exchange. Not used on established networks, due to:

- equipment performance/cost ratio is poor.
- the connections are dedicated connections between only 2 computers at a time.
- the time taken to establish a connection is long in computer terms.

However a few networks do use it, as:

- charges are only made for the duration of the connection, making circuit switching economically advantageous for low-volume networking.

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Message Switching

Partially connected networks with permanent switching circuits. The points on the network are called nodes. Messages are sent together with an address identifying the intended receiver, and are routed

through the network. Each message is received into the node storage so that it can be processed as a unit. An error-check is performed, and the routing information examined to determine what to do with the message. All this processing is carried out by the node; a number of hosts (user computers) may be connected to a single node. The nodes are usually minis suited to real-time communication processing. The multiplexing of messages leads to efficient channel usage (main advantage).
Note that a message is of arbitrary size, dependent only on the user or the application, and that a whole message is passed as a single entity. Furthermore:

- any node can communicate with any other node without having a direct physical link.
- any node can communicate with several other nodes by multiplexing the message
- delays are caused by routing (not by establishing connections as in circuit switching). However, in most networks, the internet protocol (virtual circuit) introduces some delays.

Advantages:

- sender may dispatch message at any time, even if the receiver is not ready, as the network will buffer the message
- broadcasting of a message may be possible
- equipment is used more efficiently
- messages may be handled by priority

Disadvantages

- long messages may monopolise the circuit or node, or be lost

Thus the disadvantages may be treated by restricting the length of messages...

PACKET SWITCHING.

Message broken up into packets for transmission and reassembled by the receiver. The message header is included as data in the first packet, as packets have their own headers.
Physical organisation of network same as for message switching.
Most networks use special node computers to perform the switching (in USA ARPAnet these are called Interface Message Processors IMPs)

Advantages:

- small maximum packet size makes storage allocation and management easier in the switching nodes.
- packet multiplexing enables several messages to be interleaved on one circuit
- total transmission time (delay) is reduced.

Eg

(k = packet size / t = packet time through link, d = intermediate node time)

Send a message of length $3k$ through takes $3t$ to get to the intermediate node where a delay d occurs before it is retransmitted ie $3t + d + 3t = 6t + d$ delay.

2 packets, send first packet (t), wait for d then complete (t) ie first packet takes $2t + d$.
However the other packets follow immediately giving delay $2t + d + t + t = 4t + d$.

Packet switching requires more powerful processing, but as the cost of processing falls becomes more and more widespread.
Efficiency is partially reduced by the extra header information

6.2 RADIO AND SATELLITE BROADCAST NETWORKS

Broadcast networks use a channel to which all the users are connected, so all the users receive any transmission on the channel. Channels may be radio or satellite (or twisted-pairs, co-ax or fibre-optics for local (considered later) networks). First operational broadcast computer network built using local radio by University of Hawaii. Every node receives all packets that are broadcasted, checks for errors, and then discards or processes depending on the intended receiver. Allocating the channel for transmission is a problem; we consider some schemes here in order of increasing complexity.

Simplest technique is pure aloha; when a node has a packet to transmit it does so immediately, using a positive ack protocol. If an ack is not received by timeout, the packet is assumed lost. In this case the node waits for a random length of time before retransmitting. Nevertheless, collisions can snowball, and analysis has shown a pure aloha scheme can only use a maximum of about 18% of channel capacity.

An improvement is slotted aloha - the channel time is divided into slots of a fixed length and all nodes are synchronised by a time signal on the channel. Broadcast (to / from) can only begin at the start of a time slot (36% of channel capacity used).

A collision avoidance scheme would obviously improve efficiency. In packet radio networks, a scheme called carrier sense multiple access (CSMA) is used. If a node wants to transmit, it first checks for a carrier signal indicating a broadcast. If it finds the signal it waits a random length of time before rechecking; otherwise it transmits. By using a slot length of the propagation delay (rather than packet transmission time; or pure aloha), collisions can be further reduced.

CSMA has been further refined into persistent CSMA, or p-persistent

transmission. When a node finds the channel free, it transmits its packet with a probability p, or waits for a further delay with probability (1-p). p represents a trade-off between channel efficiency and delay. Efficiencies of about 80% may be achieved using these techniques.
The long propagation delay of satellite broadcast networks means a different channel allocation technique must be used. This involves making reservations for the future use of slots in the channel. The channel is divided into frames consisting of a number of reservation slots and a number of packet slots. Each reservation slot represents a corresponding packet slot. All ground stations are synchronised to the frame timing by the satellite transmission. During the first part of the frame, each station selects a slot and transmits a marker during the appropriate reservation slot. When the slot is received back by the ground station, if it contains the ground station's marker, the packet is sent during the appropriate packet slot.
There are a number of algorithms used for ground stations to select reservation slots eg an allocated slot is allocated to the same ground station repeatedly until the ground station fails to transmit, whereupon the slot is freed.

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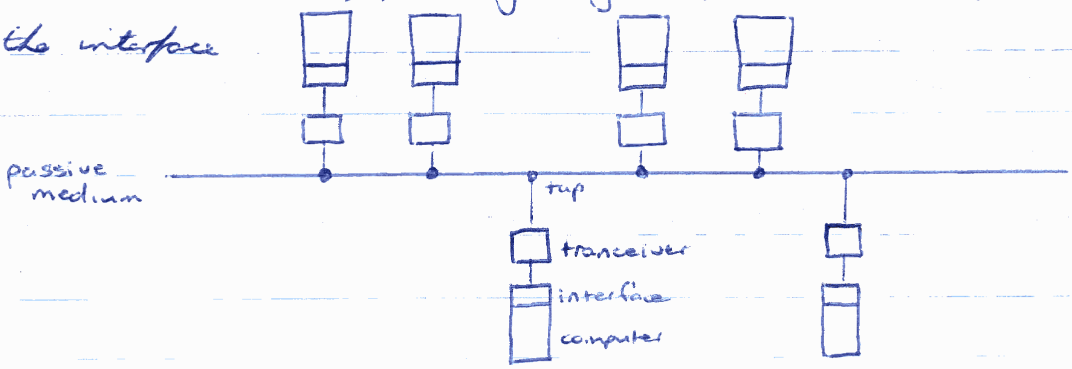
6.3 LOCAL NETWORKS

These are small private networks, generally using cable or bus technology, or ring technology. In both types all the computers are directly attached to the transmission medium by a special interface, and all the interfaces receive all the packets. The difference between them is in the way the channel is allocated. Usually a local network uses a high-bandwidth low-noise channel, so simple protocols usually suffice.

Cable or Bus Networks

The channel is used in a broadcast fashion, with allocation by CSMA or centralised slot allocation techniques. Most popular is Xerox's Ethernet, whose principles can be used on any broadcast medium: radio, telephone lines, co-axial cable, optical fibre, etc. Ethernet uses a base-band signal rather than a modulated signal for simplicity, and uses CSMA-CD (CSMA-Collision Detection, or "listen-while-talk") channel access. The sender monitors its own transmission, aborting if it detects interference. No slots are used; the ability to detect and abort on collisions causes up to 95% channel usage. A transceiver handles the CSMA-CD access, passing only complete, correct packets up to the interface.



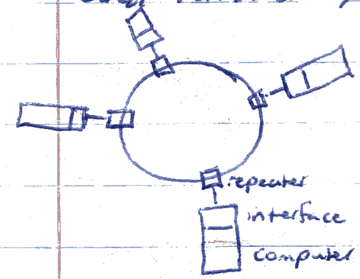
ETHERNET TOPOLOGY

ETHERNET PACKET FORMAT

The shaded parts of the packet are used by the Ethernet hardware and not used by the station. Checksum generation and checking is hardware-performed; bad packets are discarded. A timeout is used to retransmit lost packets.

Ring Networks

We consider the British Cambridge ring network. The interface is divided into two parts: the repeater and the interface proper (sometimes called the station). Twisted-pairs of wires are used as the medium. To improve reliability, the repeater is powered from the ring and electrically isolated from the interface, so four wires are used to connect the repeaters (for signalling and power). This allows a computer to be switched off and removed from the network without switching off the repeater.



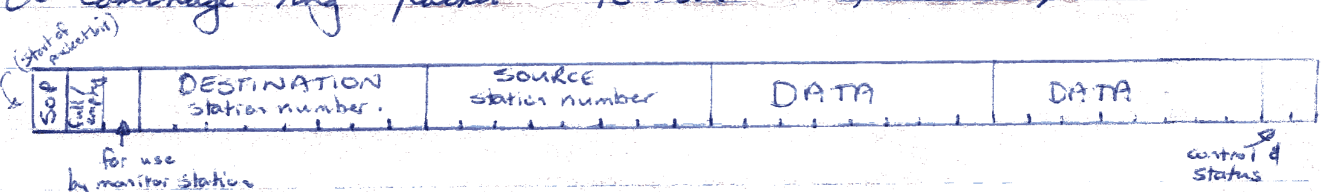
The repeater is a simple device containing a shift register and some logic to detect certain conditions in the information being passed round the ring.

The repeater normally passes any information received on to the next repeater, so transmission only occurs in one direction. When a station transmits data, the repeater transmits the information from the station rather than the information received from the previous repeater.

In the Cambridge ring a fixed size mini-packet is used which constantly circulates in the ring, and a special station called a monitor is used to generate and maintain the packet and provide the power for the repeaters.

In American rings a variable-sized packet is used which may be lost and regenerated as required by each repeater.

A Cambridge ring packet carries 16 bits of data, with the format.



The monitor provides a buffer if the ring is not physically long enough to hold a complete packet. The monitor checks the mini-packet each time it passes to ensure that it is intact, and to replace it if it is not. The monitor is thus the weakness of the system, but in practice monitor failure is rare, and the use of a monitor makes the repeaters simple and cheap.

To use the ring, a repeater would normally send a packet on unchanged unless

- the packet was intended for the repeater, in which case the data field is copied and the status set to accept.
- the repeater wishes to transmit, in which case if the full/empty bit is set to empty, the various fields are copied into the packet before sending on.

When the packet returns to the sending station the full/empty bit is reset to empty (0). and the status is checked. The channel is allocated on a round-robin basis to prevent collisions. Special source numbers are 377_8 meaning accept any source, and 000_8 meaning accept no packets.

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7. THE X25 NETWORK ACCESS PROTOCOL

7.1 LEVEL 1 - THE PHYSICAL CIRCUIT

Specification of electrical signals, voltages, plug shape and pin assignment (CCITT X21 recommendation). This level provides a bit stream transport between two sites, as well as control signals indicating the status of the physical circuit.

7.2. LEVEL 2 - LINK LEVEL

Level 2 uses a link control protocol based on the ISO HDLC (Higher Data Link Control) protocol. IBM uses a similar protocol called SDLC (Synchronous Data Link Control). Information is passed between level 2 and level 1 in blocks
p136 called frames, enabling easier storage management and error control. Each frame consists of some level 2 protocol information, and possibly an information packet from level 3.
Level 2 functions are

- establishing the link connection between DCE's.
- secure transmission in sequence of level 3 packets
- link disconnection without loss of level 3 information.

To perform these three frame formats are used:

Flag	Address	Control	FCS	Flag
8 bits	8 bits	8 bits	16 bits	8 bits

TYPE 1 U and S Format

Flag	Address	Control	Data	FCS	Flag
8 bits	8 bits	8 bits	n bits	16 bits	8 bits

TYPE 2 I FORMAT.

The fields are

FLAG FIELD - a unique 8 bit pattern (01111110) marking beginning/end of frames, used as the level 1 synchronisation character. As the flag must not appear anywhere else in the frame (as the flag is checked for by hardware, the frame must be transparent to the hardware), bit stuffing is used on the frame contents: the sender checks for sequences of five 1's and then inserts a 0 (so 111111 becomes 1111101, and 111110 becomes 1111100). The receiver also checks for sequences of five 1's. If such a sequence is followed by a 0, the 0 is discarded; otherwise the flag may have been found - the next (7^{th}) bit is inspected - if 0, the flag has been found; if 1, a frame error has occurred and the frame is aborted.

ADDRESS FIELD (redundant in a point-to-point link, and X25 does not currently support multipoint lines). The addresses used in this field allow commands and responses (receiver replies) to be separated.

CONTROL (COMMAND) FIELD - used for frame typing, sequencing and acknowledgment. Three frame types, with different control formats:

INFORMATION FRAME

0	n(s)	P/F	n(r)

SUPERVISORY FRAME

1	0	S	P/F	n(r)

UNNUMBERED FRAME

1	1	m	P/F	M

Information frames carry level 3 packets. n(s) is the frame sequence number, n(r) the acknowledgment number. P/F is the poll/final bit, set by the sender as a poll bit, and matched by the P/F bit in the response, to enable the sender of a command to recognise the corresponding response frame.

S is the supervisory frame type; m, M the unnumbered frame type.

A summary of the frame types is given overleaf.

INFORMATION FIELD - occurs only in I frames, contains the level 3 packet.

FRAME CHECK SEQUENCE - 16 bit CRC using $x^{16} + x^{12} + x^5 + 1$

Command & Response Frame Summary

Format	Commands	Responses	Encoding 1 2 3 4 5 6 7 8
I	Information		0 ← n(s) → P ← n(r) →
S		RR - Receiver Ready	1 0 0 0 F ← n(r) →
		RNR " Not "	1 0 1 0 F ← n(r) →
		REJ - Reject	1 0 0 1 F ← n(r) →
U	SARM - Set A synchronous Response		1 1 1 1 P 0 0 0
	DISC - Disconnect		1 1 0 0 P 0 1 0
		UA - Unnumbered Ack	1 1 0 0 F 1 1 0
		CMDR - Command Reject	1 1 1 0 F 0 0 1

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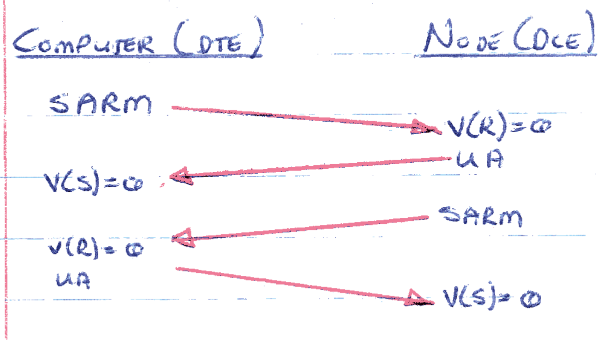
7.3 Operation of Level 2

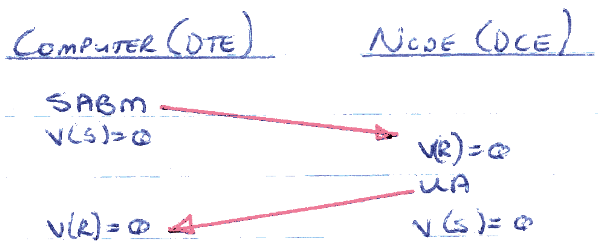
Establishing the Link Connection

First draft of X25 used LAP (Link Access Protocol); later drafts use LAPB. The DTE sends an SABM (LAP - SARM) unnumbered command to the DCE (and sets V(S) = 0). When the DCE receives this, it sets its internal variable V(R) to zero (next expected sequence number), returns an unnumbered acknowledgement, and sets its V(S) (next seq # to be sent) to zero. When the DTE receives the UA it sets its V(R) = 0. Data transfer may then begin.

LAB Link Setup



LAPB Link Setup



Data Transfer.

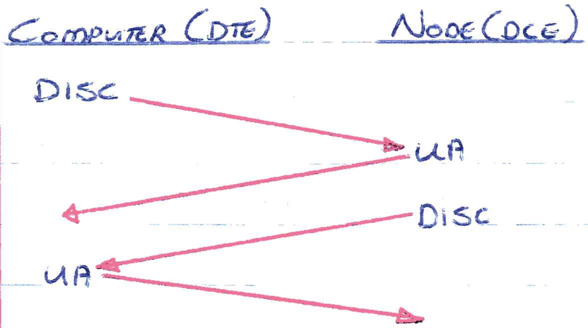
Go-Back-N ARQ with a window size of 7 is used. Two basic errors can occur:

- the FCS indicates an error in the packet
- the packet appears to contain no errors, but the address and control fields are incorrect.

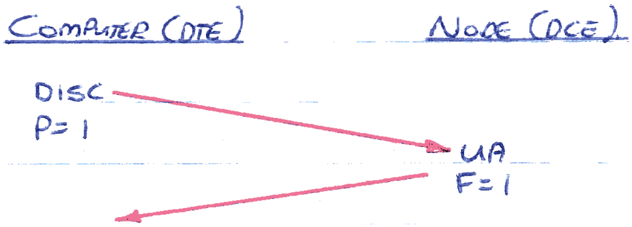
In the first case the frame is discarded as usual. The second case can be due to a number of possibilities; the most common is a sequence error due to a previous packet being lost or discarded. If the sequence number of the frame is in the window, the receiver responds REJ with the n(r) field indicating from which frame the sender should retransmit. If the number is outside the window, a CMDR response is sent, initiating a reset procedure. If the receiver is short of buffer space and wants to avoid timing out, a RNR can be sent, to be followed by a RR once space has been made.

Link Disconnection.

LAP Link Disconnection



LAPB Link Disconnection



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7.4 Level 3 - The Network Level

X25 allows multiple connections via a single level 2 link using level 3 logical connection (virtual host-to-host) Level 3 connections are thus logical, and are identified by logical channel numbers. A logical channel is a virtual circuit.

Packet Format

Level 3 headers contain a variety of information which varies for different types of packets so the format is complex. Each starts with a 4-bit General Format Identifier (GFI) indicating the packet format, followed by a 12-bit channel numbering field, the rest of the header, and the packet information. Call requests and incoming call packets contain the address of the remote host (DTE). The addresses may contain fields indicating a particular process on a host as well as identifying a host. The address fields are of variable length, preceded by a byte count.

Connection Establishment.

To begin a call, a host uses the full address of the remote host in a call request packet sent from the DTE to the DCE. The DCE will respond with either a call accepted packet if the request was successful, or a clear indication packet if the call was refused.

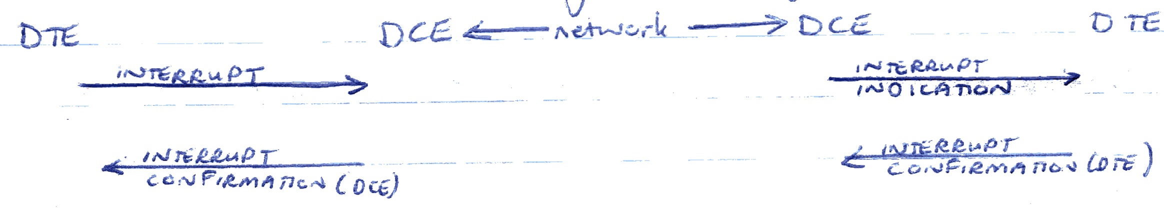
p149. At the remote host the DCE will send an incoming call packet to the host to indicate that a remote call has been requested. The DTE receiving an incoming call packet will inspect the header to decide if the call is to be established or not; if the call is accepted a call accept packet is sent to the DCE; if rejected a clear request is sent to the DCE. The channel number chosen for a new outgoing call by the DTE is the lowest available number; the channel number for an incoming call is chosen by the DCE as the highest available number. If the DTE and DCE agree on the channel number, and the DCE accepts the call, all subsequent packets on that call will only use that channel number, not the full address. A call has been established to a remote DTE when a host receives a call accepted packet having the same channel number as the call request packet. The call is established at the remote DTE when the call accepted packet is sent to the DCE. X25 level 3 also has a datagram facility that allows a DTE to send a single packet to a remote DTE and receive a single packet in reply; the information is placed in the call request and call reject packets, thus effectively opening and closing a call in a single packet exchange.

p149 INFORMATION TRANSFER.

Sequenced stream using GO-BACK-N ARQ. At level 3 the level 2 link is assumed to be error-free, so windows and sequence numbers are now used primarily for end-to-end flow control. Complete suspension of the data flow can be achieved by a level 3 RNR, but the flow can be throttled back, or quenched, by the rate of acknowledgement, as the window limits the number of data packets in flight at any one time.

p180 INTERRUPTS

Normal user data on an X25 virtual call is kept in sequence, called "in-band" transmission. Interrupt packets are not subject to normal flow control, but are "out-of-band". Interrupt packets contain a field indicating the cause of the interrupt, and it is up to the application to decide what to do with both the interrupt and subsequent data. When the DTE sends an interrupt packet the DCE responds with an interrupt confirmation. X25 permits only one interrupt to be outstanding on a logical call.



Resets and Restarts.

If information frames or sequence numbers become out of synchronisation on a single call (or an error is detected by a higher protocol), only one logical channel is affected so a reset is used, clearing the sequence numbers at DTE and DCE to 0. The reset can be initiated by either the DTE or DCE (usual request/confirm) and is propogated to the remote DTE-DCE pair.

If the DTE and DCE become unsynchronised over global level 3 operation (eg choice of channel number) a restart is used, affecting all the logical channels (same as a level 2 reset)

Call Disconnection

DTE issues a clear request packet to DCE. DCE forwards any remaining information packets. At the remote end the DCE sends a clear indication packet to the DTE, which, after acknowledging all outstanding packets, responds with a clear confirmation packet. The first DCE also sends a clear confirmation packet to its DTE, and the channel is freed.

Data Flow

X25 recommendation specifies packet sizes of 16, 32, 128, 256, 512 and 1024 bytes, with 128 the preferred maximum. Level three uses an M bit to indicate that the current packet is not the last (ie More) and a Q bit set when the packet contains control information.

7.5 X25 Summary

X25 suffers in that it only defines the interaction between the host and its local network node (ie DTE - DCE communication). The host-to-host transfer seen through X25 can be very different depending on how the protocol is implemented. There appear to be redundant features in flow control, connection and disconnection, as both level 2 and level 3 are only defined across the DTE-DCE interface. However, a network using X25 throughout can make more use of the level 3 flow control on logical calls.

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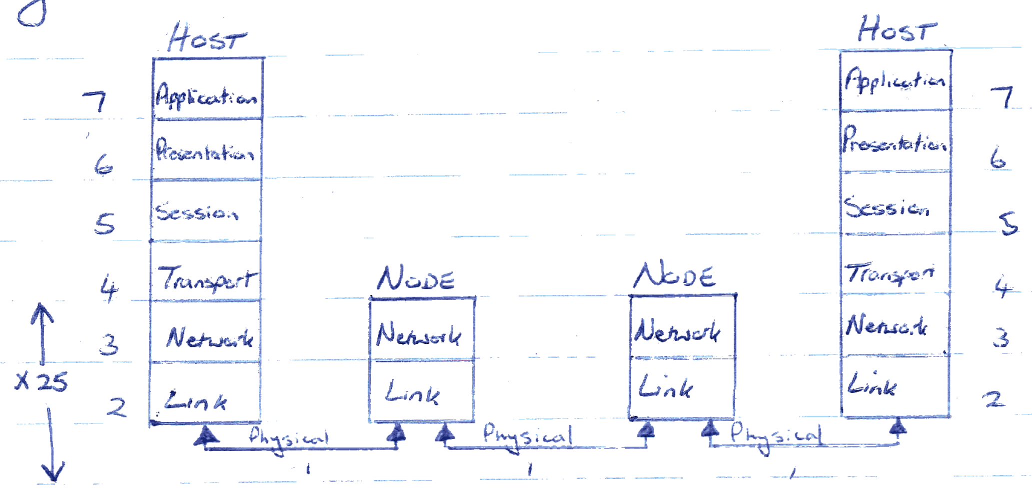
8. End-to-End Network Protocols

8.1 Protocol Hierarchy

The ISO Open Systems Interconnection (OSI) specification based on X25 (and thus not compatible with, say, Arpanet) has 7 layers:



Transport Level

Responsible for the safe transfer of messages from one application process to another, providing addressing to a user-application (rather than a host as in level 3), sequencing, multiplexing of connections and some error and flow control.

Session Level

Responsible for the management of resources used by the network applications, and address mapping between network addresses and actual processes.

Presentation Level

Maps user requirements into network actions, ie responsible for things such as device independence, address to service mapping, and virtual terminals). Many special protocols are implemented at this level for specific functions such as file transfer, message service, interactive use, etc.

APPLICATION LEVEL.

These are the actual user application processes.

8.2 DATAGRAMS AND VIRTUAL CALLS. (level 3½).

Each packet may be treated as a seperate unit or datagram, and forwarded to its destination by the best available route. The receiver will then have to reassemble the packets into order and detect any missing packets. Each datagram carries the full source and destination address. Advantages of datagrams are:

- routing can be very flexible and dynamic, leading to better reliability
- network implementation is simpler and cheaper.
- transport level will have some flow & error control anyway, so datagrams do not duplicate this.

p159 A virtual call