Digital Communications Engineering 1

(COMM2108)

Multiplexing

Multiplexing

- *Multiplexing* and *Multiple Access* refer to the sharing of a fixed communications resource (CR) between a number of signal sources.

- The overall objective is to allow the various signal sources share a common CR without creating unmanageable interference to each other in the detection process.

Multiplexing

- With *multiplexing* - the sharing of the CR is fixed and the resource allocation is assigned *a priori* (i.e. beforehand) and is usually performed locally.

- With *multiple access* - the sharing of the CR is performed remotely (usually in a distributed or decentralised manner). The signalling or control requirements of such a scheme constitute an overhead which sets an upper limit on the efficiency and utilisation of the CR.

Multiplexing

- The basic ways to distribute or share out the communications resource (CR) are:
  - Frequency Division
  - Time Division
  - Code Division
  - Space Division
  - Polarization Division
- Here we will only consider frequency division and time division in any detail.

Multiplexing

- In *Frequency Division Multiplexing (FDM)/ Frequency Division Multiple Access (FDMA),* the bandwidth of the CR is divided up into sub-bands which are shared out amongst the various signal sources.
- In other words, each signal is allocated its own frequency channel for its transmission.
- Simple technique where no timing or synchronisation is required between the channels.
- Example: Cable TV where 10-15 TV channels share the same coaxial cable medium without interfering with each other.

Multiplexing

- In *time division multiplexing (TDM)/time division multiple access (TDMA),* access to the CR is shared out on a time basis using periodically recurring time slots.
- Each signal source takes it in turn to use the full bandwidth of the CR for a short time.
- The simplest TDM/TDMA scheme is the fixed assignment scheme where the *M* time slots that make up a *frame* are pre-assigned to the various signal sources.

Multiplexing

- Multiple access schemes are termed fixed assignment when a source has periodic access to the CR independent of its actual need.
- A fixed assignment TDM/TDMA is extremely efficient when the source requirements are predictable and the traffic is heavy (i.e. all the time slots are filled).
- Timing or synchronisation information needs to be included in the scheme.

Multiplexing

- Dynamic assignment (or demand assignment multiple access) schemes give the source access to the CR only when it requests access.
- Dynamic assignment schemes operate on the basis that the actual demand for access rarely equals the peak demand for access.
- When the source requirements are unpredictable there are more efficient schemes involving the dynamic assignment of time slots. Such schemes are known as packet-switched systems, statistical multiplexers or concentrators.

Multiplexing

- *Code division multiplexing (CDM)/code division multiple access (CDMA)* may be viewed as a hybrid of FDM and TDM techniques where both frequency and time are shared out among the signal sources.
- This technique is part of an important class of techniques known as spread-spectrum which have a number of significant advantages over conventional FDM and TDM.
- CDM/CDMA is the basis for 3G mobile phone systems.

Multiplexing

- In *space division (or multiple beam frequency reuse)* highly directional antennas are used to separate radio signals by pointing in different directions. This allows the reuse of the same frequency band.
- In *polarization division (or dual polarization frequency reuse)* orthogonal polarizations are used to separate the signals. This allows for reuse of the same frequency band.

Multiplexing

- Performance Comparison of FDMA and TDMA
- We assume that:
  - The communications resource is capable of supporting a total of R bits/sec.
  - There are M sources that generate information at an average rate of R/M bits/sec. Furthermore, the information is transmitted in groups or packets of b bits/pkt.
  - In the FDMA system, the system bandwidth W (Hz) is divided equally into M disjoint frequency sub-bands of width W/M (Hz).
  - In the TDMA system, the frame is divided into M time slots.

Multiplexing

- In the FDMA system, the *b* bit packets are transmitted in *T* seconds over the *M* disjoint channels. Therefore, the total bit rate R_{FDMA} required is:

$$R_{FDMA} = M\,(b/T)\ bits/sec$$

- In the TDMA system, the *b* bits packets are transmitted in *T/M* seconds from each source. Therefore the total bit rate R_{TDMA} required is:

$$R_{TDMA} = b/(T/M) = M\,(b/T)$$
$$= R_{FDMA}$$

- Both systems have the equivalent bit rate performance.

Multiplexing

- Considering another performance metric, for example the average packet delay.
- For simplicity we assume deterministic data sources are used and that the CR is fully utilised, i.e. all the frequency sub-bands in FDMA are used and all the time slots in TDMA are used.
- Furthermore, we assume there are no overhead costs as a result of guard bands or guard times.
- The message delay *D* can be defined as

$$D = w + \tau$$

where w is the average packet waiting time and τ is the packet transmission time.

Multiplexing

- In the case of FDMA, each packet is sent over a *T* second interval, so the packet transmission time for FDMA is

$$\tau_{FD} = T$$

- Since the FDMA channel is continuously available and packets are sent as soon as they are generated, the waiting time

$$w_{FD} = 0$$

- The average delay time for FDMA is therefore

$$D_{FD} = T$$

Multiplexing

- In the case of TDMA, each packet is sent in slots of *T/M* seconds, so the packet transmission time for TDMA is

$$\tau_{TD} = T/M = b/R$$

- From the handout, it can be seen that each slot begins at a different point in the *T* second frame, i.e. each packet S_{mk} will start at $(m-1)T/M$ seconds. Therefore, the average waiting time that a TMDA packets undergoes before transmission is

$$w_{TD} = \frac{1}{M}\sum_{m=1}^{M}(m-1)\frac{T}{M}$$

Multiplexing

- Substituting *n* for *(m-1)* we get

$$w_{TD} = \frac{T}{M^2}\sum_{n=0}^{M-1} n$$

$$= \frac{T}{M^2}\frac{(M-1)M}{2}$$

$$= \frac{T}{2}\left(1 - \frac{1}{M}\right)$$

Multiplexing

- The average delay time for TDMA is

$$D_{TD} = \frac{T}{2}\left(1 - \frac{1}{M}\right) + \frac{T}{M} = \frac{T}{2} - \frac{T}{2M} + \frac{T}{M}$$

$$= \frac{T}{2} + \frac{T}{2M} = T - \frac{T}{2}\left(1 - \frac{1}{M}\right)$$

$$= D_{FD} - \frac{T}{2}\left(1 - \frac{1}{M}\right)$$

- This result indicates that TDMA is inherently superior from a message delay point of view.

Multiplexing

- TDM Case Study:
  - TDM is best illustrated by a study of the European standard used for multiplexing digitized voice channels in the modern digital telephone network (PSTN).
  - Known as the 30 Channel PCM/TDM System.
  - Recall that for telephony purposes, voice channels are bandlimited to 3.4 kHz and oversampled at 8 kHz. The samples are quantised using A-law companding and encoded using an 8-bit code word. This whole process yields a digitized toll-quality voice signal at 64 kbits/sec.

Multiplexing

- 30 Channel PCM/TDM System:
  - In this scheme 30 PCM encoded voice channels (including 2 additional control channels) are multiplexed together using TDM to an overall digital signal at 2.048 Mbits/sec (known as the *E1 rate*).
  - In this 30+2 channel scheme there is a *frame* and *multiframe* structure used for synchronisation and signalling purposes.
  - It is worth noting that North America and Japan have their own telephony TDM scheme based multiplexing 24 voice channels and 1 control channels to give a 1.544 Mbits/sec digital signal (known as the *T1 rate*).

Multiplexing

- 1 multiframe (2 msec) = 16 frame slots
- 1 frame slot (125 μsec) = 32 channel slots
- 1 channel slot (3.9 μsec) = 8 pulse slots
- 1 pulse slot (488 nsec) = Logic 0/1

- Bit Rate = $1/488 \times 10^{-9}$ = 2.048 Mbits/sec
  (or 2 Mbits/sec for short)

Multiplexing

- Channel slots 0 and 16 in a frame are reserved for synchronisation and signalling purposes.
- Channel slot 0 is always used for frame synchronisation.
- Channel slot 16 is used for multiframe synchronisation (in frame slot 0 only) and for signalling.
- Signalling is used to support features such as dialling tones, engaged tones, alarms etc.

Multiplexing

- Frame 0 (in a multiframe)
  - Channel slot 0 = Frame alignment word (frame synchronisation)
  - Channel slot 1 = 1st sample from voice channel 1
  - Channel slot 2 = 1st sample from voice channel 2

    : :
  - Channel slot 15 = 1st sample from voice channel 15
  - Channel slot 16 = Multiframe alignment word (multiframe synchronisation, <u>frame 0 only</u>!!)
  - Channel slot 17 = 1st sample from voice channel 16

    : :
  - Channel slot 31 = 1st sample from voice channel 30

Multiplexing

- Frame 1 (in a multiframe)
  - Channel slot 0 = Frame alignment word (frame synchronisation)
  - Channel slot 1 = 2^{nd} sample from voice channel 1
  - Channel slot 2 = 2^{nd} sample from voice channel 2

    : :
  - Channel slot 15 = 2^{nd} sample from voice channel 15
  - Channel slot 16 = Bits 1-4 Signalling for voice channel 1
Bits 5-8 Signalling for voice channel 16
  - Channel slot 17 = 2^{nd} sample from voice channel 16

    : :
  - Channel slot 31 = 2^{nd} sample from voice channel 30

Multiplexing

- Frame 2 (in a multiframe)
  - Channel slot 0 = Frame alignment word (frame synchronisation)
  - Channel slot 1 = 3rd sample from voice channel 1
  - Channel slot 2 = 3rd sample from voice channel 2

    : :
  - Channel slot 15 = 3rd sample from voice channel 15
  - Channel slot 16 = Bits 1-4 Signalling for voice channel 2

    Bits 5-8 Signalling for voice channel 17
  - Channel slot 17 = 3rd sample from voice channel 16

    : :
  - Channel slot 31 = 3rd sample from voice channel 30

Multiplexing

- Frame 15 (in a multiframe)
  - Channel slot 0 = Frame alignment word (frame synchronisation)
  - Channel slot 1 = 16^{th} sample from voice channel 1
  - Channel slot 2 = 16^{th} sample from voice channel 2

    : :
  - Channel slot 15 = 16^{th} sample from voice channel 15
  - Channel slot 16 = Bits 1-4 Signalling for voice channel 15

    Bits 5-8 Signalling for voice channel 30
  - Channel slot 17 = 16^{th} sample from voice channel 16

    : :
  - Channel slot 31 = 16^{th} sample from voice channel 30

Multiplexing

- The 2.048 Mbits/sec multiplex level is known as the primary multiplex level group as it represents the first level in a hierarchy of multiplexing.
- By combining 4 such 2.048 Mbits/sec signals (or *tributaries*) together, one gets a 8.488 Mbits/sec signal that can support 120 voice channels.
- It is possible to continue multiplexing signals together in this fashion. This leads to the development of a *multiplexing hierarchy*.

Multiplexing

Multiplexing Order	Bit Rate (Mbits/sec)	No. of Voice Channels
1	2.048	30
2	8.488	120
3	34.368	480
4	140	1,920
5	565	7,680
6	1.2 Gbits/s	30,720
7	2.4 Gbits/s	122,880

Multiplexing

- The digital telephone network in North America and Japan use a completely different multiplexing hierarchy.
- Based upon 24 voice channels and 1 control channel multiplexed together to give a primary multiplex rate of 1.544 Mbits/sec.
- A similar multiplexing hierarchy exists for this standard.
- Unfortunately, the two standards are incompatible which means that complex and costly interworking equipment is required.

Multiplexing

- There is a major drawback associated with this PCM/TDM multiplexing hierarchy.
- In assembling a PCM/TDM multiplexing hierarchy, the various tributaries will have been generated at different locations and in many cases using different equipment.
- As there is no master clock available for the system, there will always be slight variations in the exact bit rates of the tributaries.
- Consequently, the system must be able to accommodate slight variations in the bit rates if it is to be able to operate in the real world.

Multiplexing

- The fact that the system can tolerate variations in the tributary bit rates, means that the system is *plesiochronous* or almost synchronous.
- Hence, this multiplexing hierarchy is often referred to as the *Plesiochronous Digital Hierarchy* or *PDH* for short.
- In order to bring all the bit streams up to the same rate, extra “dummy” or *justification bits* are added to the bit streams.

Multiplexing

- Elastic stores (i.e. buffers) are used in a typical multiplexer to ensure sufficient bits are always available for transmission or reception.
- Stores are required because PDH interleaves bytes together for the primary multiplex level but interleaves bits at higher levels.
- A *frame alignment signal* is used for synchronisation and the correct mapping of tributaries to the appropriate outputs.
- The *ITU-T G.702* recommendation defines the complete PDH.

Multiplexing

- PDH was developed at a time when transmission costs were low relative to switching costs. However, this is no longer valid.
- PDH was designed for point-to-point transmission applications in which the entire multiplex is decoded at the far end. This is a complicated process requiring complete demultiplexing at each level and removal of the justification bits.
- A single 2 Mbits/sec channel cannot be added to or extracted from a higher multiplex level without multiplexing down and remultiplexing up.
- PDH does not support definite or clear identification of the channels being carried.

Multiplexing

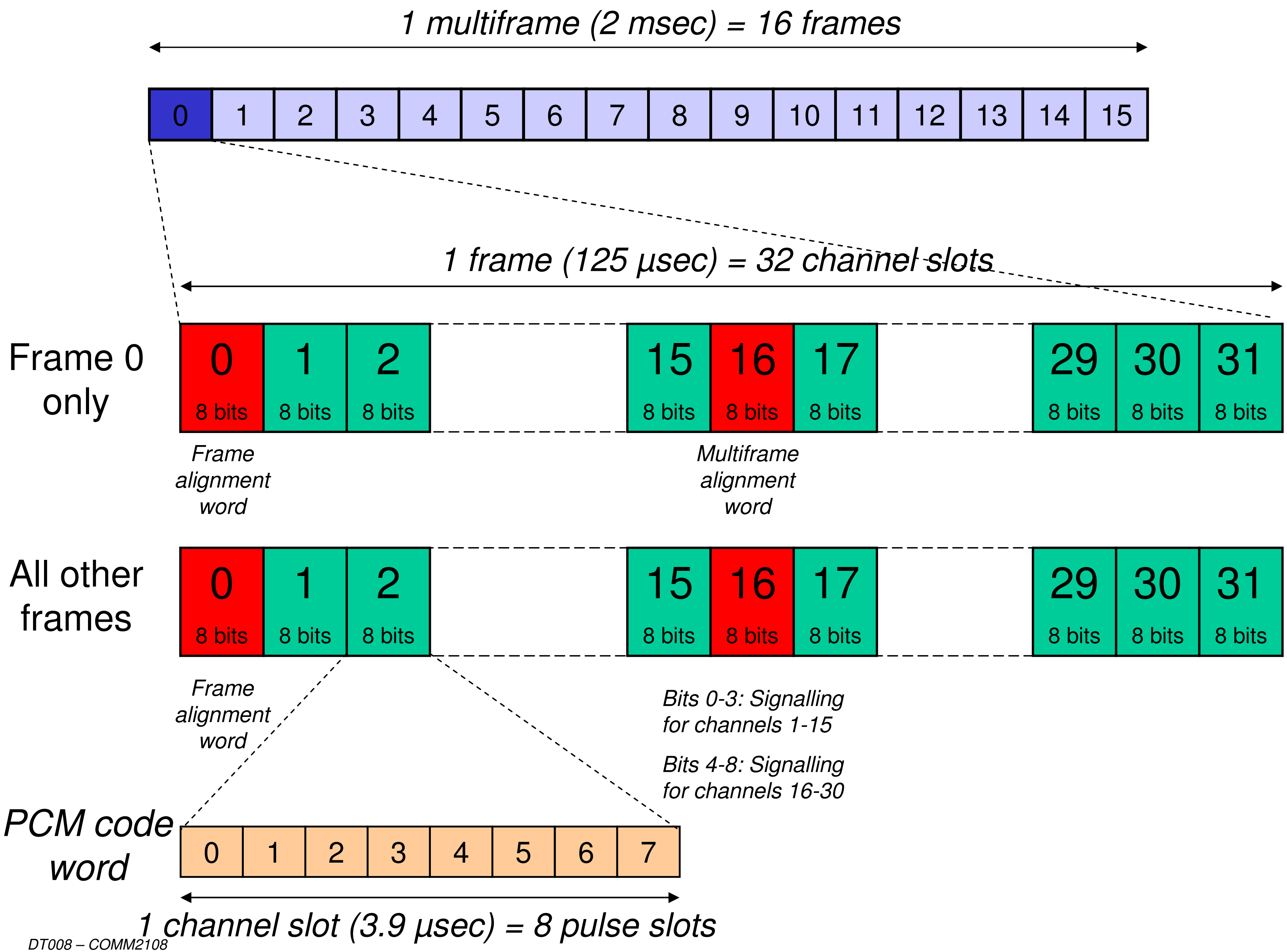
- PDH is severely limited in its ability to meet the requirements of future networks.
- PDH does not provide a cost effective, flexible architecture for today's networks.
- There is no extra capacity in the PDH frame structure to support network OAMP&P (Operations, Administration, Maintenance, and Provisioning).
- Finally, PDH is <u>not</u> a global standard, i.e. the standard for North America and Japan is based upon a primary multiplex level of 1.544 Mbits/sec.

Multiplexing

- The *SONET (Synchronous Optical Network)* concept was introduced in the USA in 1986 to establish wideband transmission standards so that international operators could interface using standard frame formats and signalling protocols.
- SONET also included network flexibility and intelligence, and additional channels to carry control and performance information between network elements and control centres.
- In 1988, adopted by the ITU-T (and ETSI) and renamed *SDH (Synchronous Digital Hierarchy),* establishing a world wide standard.

Multiplexing

- Advantages of SONET/SDH:
  - Easier to add and drop signals
  - More bandwidth available for network management.
  - Easier to introduce new services.
  - World wide standard.
- Some ITU-T Recommendations for SDH are:
  - G.707 at 155 Mbits/sec
  - G.708 at 622 Mbits/sec
  - G.709 at 2.5 Gbits/sec



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