CMR INSTITUTE OF TECHNOLOGY

USN										

Internal Assesment Test – II

Sub:	Multimedia Communication		Sec	A, B ,C& D			Code:	10EC841
Date:	19 / 04 / 2018	Duration:	90 mins	Max Marks:	50	Sem: VIII	Branch:	ECE

Note: Answer any five full questions.

		Marks	OBE CO	OBE RBT
1.	Explain the Principle of operation of a PCM Speech CODEC with a neat diagram.	10	CO1	L2
2.	Messages comprising seven different characters, A through G, are to be transmitted over a data link. Analysis has shown that the relative frequency of occurrence of each character is A 0.10, B 0.25, C 0.05, D 0.32, E 0.01, F 0.07, G 0.2 (i) Derive the entropy of the messages (ii) Use static Huffman coding to derive a suitable set of codeword and construct a tree (iii) Derive the average number of bits per codeword for your codeword set.	10	CO3	L3
3.	With a neat block diagram explain JPEG Encoder.	10	CO3	L2
4.	Explain with a neat diagram IP Adjunct Protocol.	10	CO5	L2
5.	Define Fragmentation and Reassembly. Explain how the various fields in each packet header are used to perform fragmentation and reassembly by considering transport protocol in a host that is attached to a token ring LAN transferring a block of 7000 bytes. Assume the MTU associated with Token ring LAN is 4000 byte and that of Ethernet LAN is 1500 bytes. The header of each IP Datagram requires 20 bytes.	10	CO5	L3
6.	(a) Explain the concept of Interlaced Scanning.	5	CO1	L2
	(b) Assuming the bandwidth of a speech signal is from 50 Hz through to 10 kHz and that of a music signal is from 15 Hz through to 20 kHz, derive the bit rate that is generated by the digitization procedure in each case assuming the Nyquist sampling rate is used with 12 bits per sample for the speech signal and 16 bits per sample for the music signal. Derive the memory required to store a 10 minutes passage of stereophonic music.	5	CO1	L3
7.	Explain the concept of Arithmetic coding. Compute the arithmetic code word for the message “went.” Comprising a string of characters with probabilities e=0.3, n=0.3, t=0.2, w=0.1, . =0.1	10	CO3	L3

CMR Institute of Technology

Department of ECE

Scheme and Solution –IAT 2

Multimedia Communication-10EC841

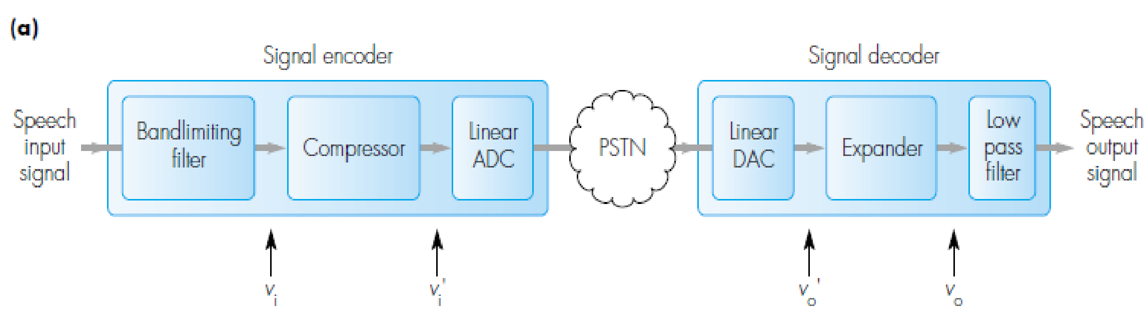
1. PCM Speech CODEC (Diagram-5 M,Explanation-5 M)

   It is a digitization process. Defined in ITU-T Recommendations G.711. PCM consists of encoder and decoder. It consists of expander and compressor. As compared to earlier where linear quantization is used – noise level same for both loud and low signals.

   AS ear is more sensitive to noise on quite signals than loud signals, PCM system consists of non-linear quantization with narrow intervals through compressor. At the destination expander is used. The overall operation is companding. Before sampling and using ADC, signal passed through compressor first and passed to ADC and quantized.

   At the receiver, codeword is first passed to DAC and expander.

   Two compressor characteristics – A law and μ law.



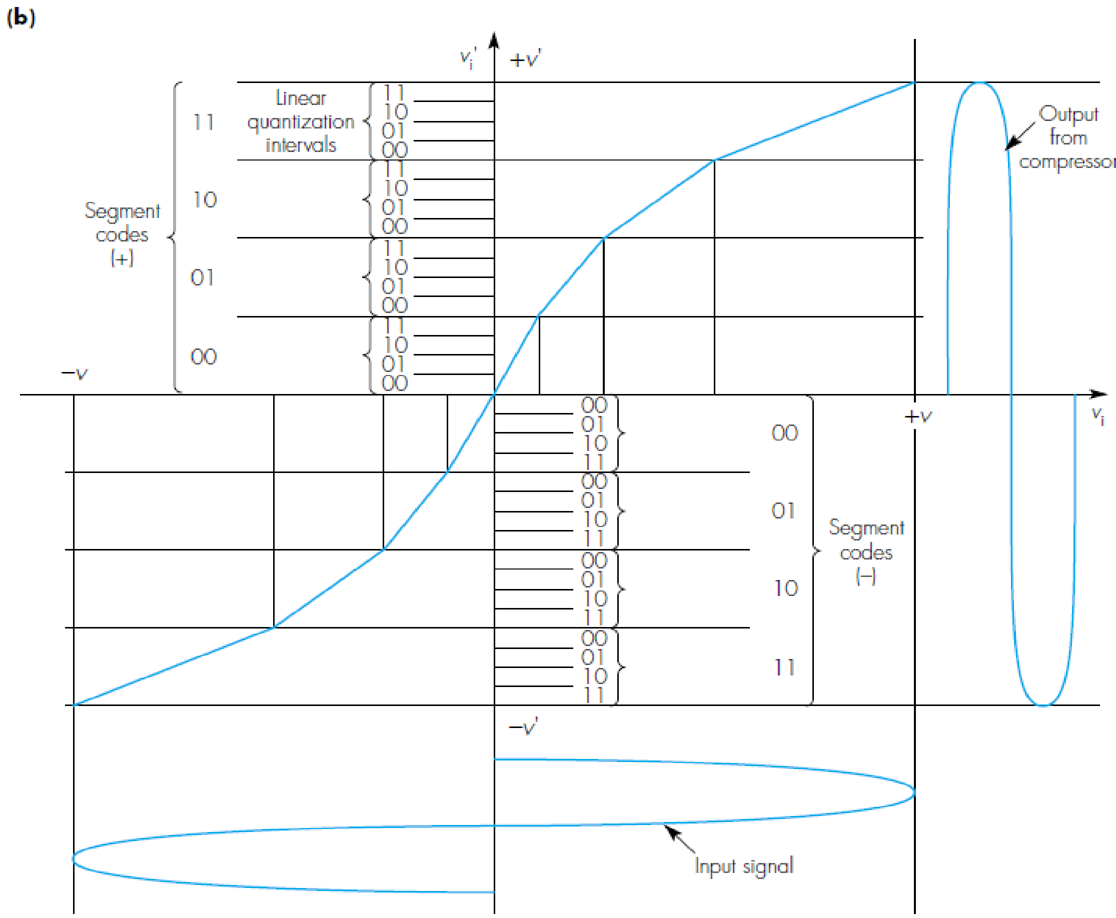
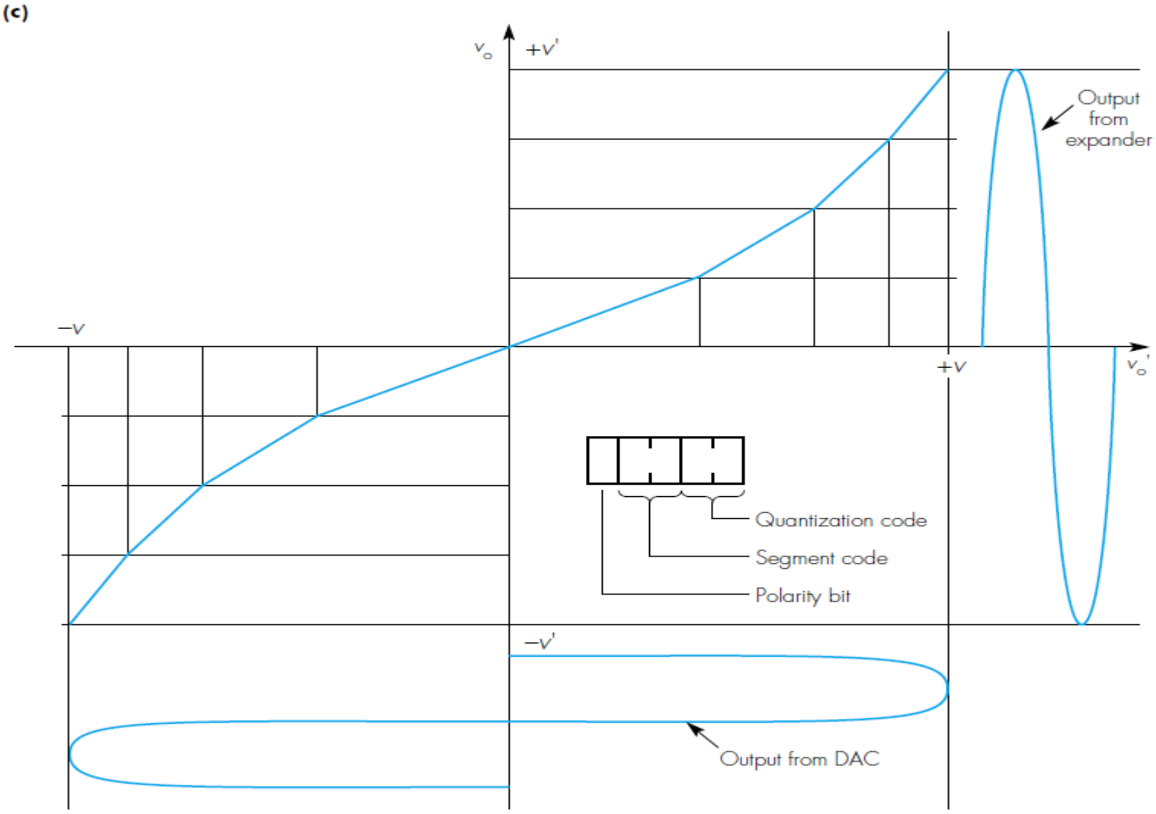


Fig: Compressor Characteristics

(c)



Note that in the G.711 standard a 3-bit segment code and 4-bit quantization code are used.

Fig: Expander Characteristics

2. **Static Huffman Encoding Problem**

$-\left[-(0.3010) \phi\ 0.4515\right)$

Entropy $H = -\sum_{i=1}^{n} P_i \log_2 P_i$

$= -\left[0.10 \log_2 0.10 + 0.25 \log_2 0.25 + 0.05 \log_2 0.05 + 0.32 \log_2 0.32 + 0.01 \log_2 0.01 + 0.07 \log_2 0.07 + 0.2 \log_2 0.2\right]$

$= -3.32\left[-0.1 \ \cancel{\phi}\ 0.15 - 0.065 - 0.158 - 0.002 - 0.080 - 0.139\right]$

$= -3.32\left[-0.712\right] = 2.36$

D 0.32 →	D 0.32 →	D 0.32 →	D 0.32 →	D 0.32 →	0.5 (1)
B 0.25 →	B 0.25 →	B 0.25 →	B 0.25 →	B 0.25 (1)	D 0.32 (0)
A 0.10 →	A 0.10 →	A 0.10 →	0.15 (1) →	0.25 (0)	
F 0.07 →	F 0.07 →	0.08 (1) →	A 0.10 (0)		
C 0.05 →	C 0.05 (1) →	F 0.07 (0)			
G 0.02 (1) →	0.03 (0)				
E 0.01 (0)					

0.82

A → 100

B → 11

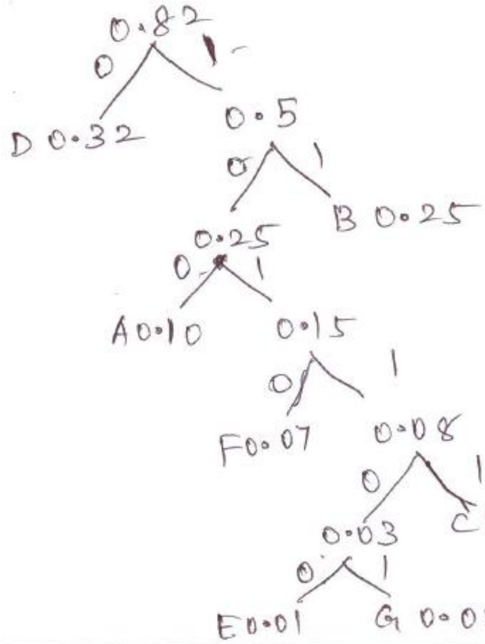
C → 10111

D → 0

E → 101100

F → 1010

G → 101101



Average no. of bits per codeword

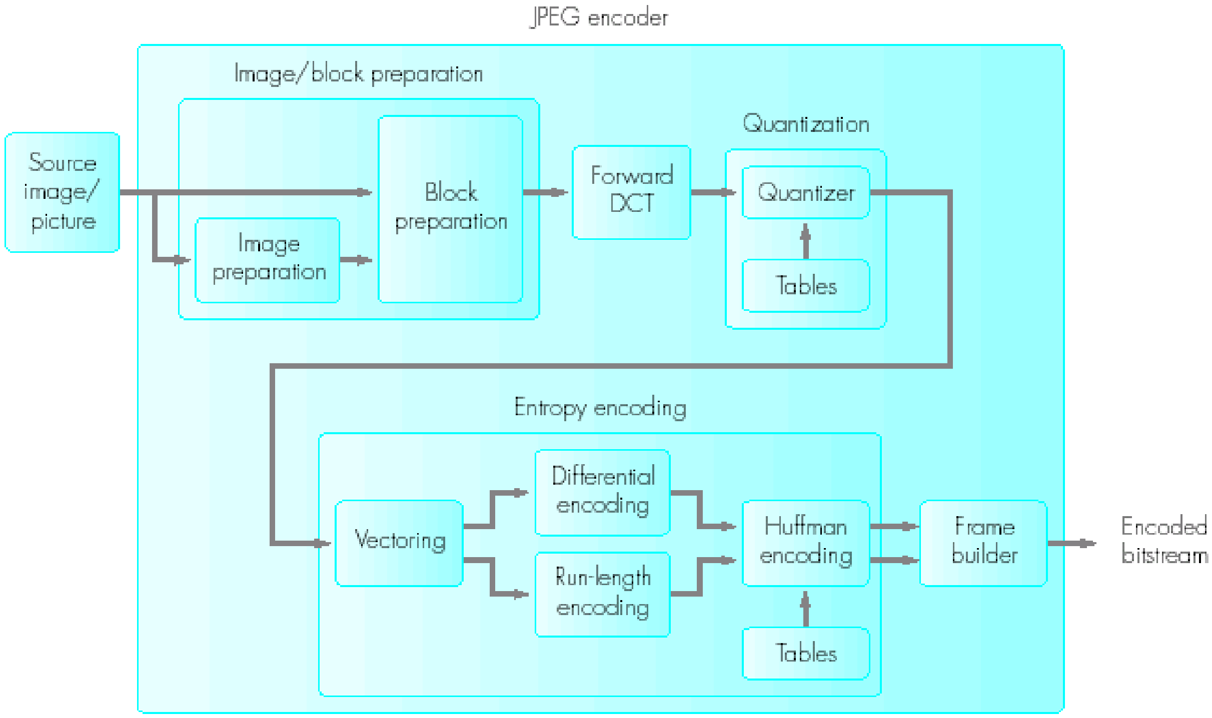
$= \sum_{i=1}^{n} N_i P_i$

$= 3(0.10) + 2(0.25) + 5(0.05) + 1(0.32) + 6(0.01) + 4(0.07) + 6(0.2)$

$= 0.3 + 0.5 + 0.25 + 0.32 + 0.06 + 0.28 + 1.2$

$= 2.91$

3. JPEG Encoder (Diagram-5 M, Explanation-5 M)



Source image is made up of one or more 2-D matrices of values. 2-D matrix is required to store the required set of 8-bit grey-level values that represent the image. For the colour image if a CLUT is used then a single matrix of values is required. If the image is represented in R, G, B format then three matrices are required. If the Y, C_r, C_b format is used then the matrix size for the chrominance components is smaller than the Y matrix (Reduced representation)

Once the image format is selected then the values in each matrix are compressed separately using the DCT. In order to make the transformation more efficient a second step known as **block preparation** is carried out before DCT. In block preparation each global matrix is divided into a set of smaller 8X8 sub matrices (block) which are fed sequentially to the DCT.

- Each pixel value is quantized using 8 bits which produces a value in the range 0 to 255 for the R, G, B or Y and a value in the range –128 to 127 for the two chrominance values C_b and C_r.If the *input matrix* is *P[x,y]* and the *transformed matrix* is *F[i,j]* then the DCT for the 8X8 block is computed using the expression:

$$F[i,j] = \frac{1}{4} C(i) C(j) \sum_{x=0}^{7} \sum_{y=0}^{7} P[x,y] \cos \frac{(2x+1)i\pi}{16} \cos \frac{(2y+1)j\pi}{16}$$

- All 64 values in the input matrix *P[x,y]* contribute to each entry in the transformed matrix *F[i,j]*. For *i = j = 0* the two cosine terms are 0 and hence the value in the location *F[0,0]* of the transformed matrix is simply a function of the summation of all the values in the input matrix.

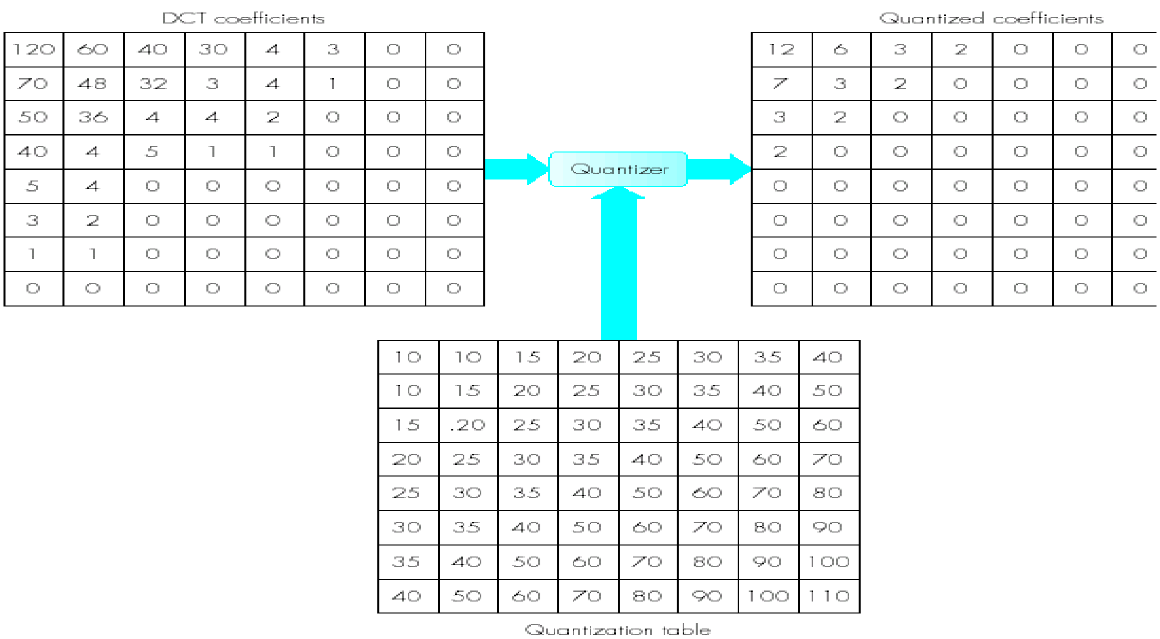
This is the mean of all 64 values in the matrix and is known as the **DC coefficient.** Since the values in all the other locations of the transformed matrix have a frequency coefficient associated with them they are known as **AC coefficients.**

- ***for j = 0 only the horizontal frequency coefficients are present***
- ***for i = 0 only the vertical frequency components are present***
- ***For all the other locations both the horizontal and vertical frequency coefficients are present***

Quantization:

- Using DCT there is very little loss of information during the DCT phase. The losses are due to the use of fixed point arithmetic. The main source of information loss occurs during the quantization and entropy encoding stages where the compression takes place. The human eye responds primarily to the DC coefficient and the lower frequency coefficients (The higher frequency coefficients below a certain threshold will not be detected by the human eye). This property is exploited by dropping the spatial frequency coefficients in the transformed matrix (dropped coefficients cannot be retrieved during decoding).
- In addition to classifying the spatial frequency components the quantization process aims to reduce the size of the DC and AC coefficients so that less bandwidth is required for their transmission (by using a divisor). The sensitivity of the eye varies with spatial frequency and hence the amplitude threshold below which the eye will detect a particular frequency also varies. The threshold values vary for each of the 64 DCT coefficients and these are held in a 2-D matrix known as the **quantization table** with the threshold value to be used with a particular DCT coefficient in the corresponding position in the matrix. The choice of threshold value is a compromise between the level of compression that is required and the resulting amount of information loss that is acceptable. JPEG standard has two quantization tables for the luminance and the chrominance coefficients. However, customized tables are allowed and can be sent with the compressed image.
- *Entropy encoding consists of four stages*
- **Vectoring – The entropy encoding operates on a one-dimensional string of values (vector). However the output of the quantization is a 2-D matrix and hence this has to be represented in a 1-D form. This is known as vectoring**
- ***Differential encoding*** **– In this section only the difference in magnitude of the DC coefficient in a quantized block relative to the value in the preceding block is encoded. This will reduce the number of bits required to encode the relatively large magnitude**
- **The difference values are then encoded in the form *(SSS, value) SSS indicates the number of bits needed and actual bits that represent the value***
- **e.g: if the sequence of DC coefficients in consecutive quantized blocks was: 12, 13, 11, 11, 10, --- the difference values will be 12, 1, -2, 0, -1**

•



DCT coefficients

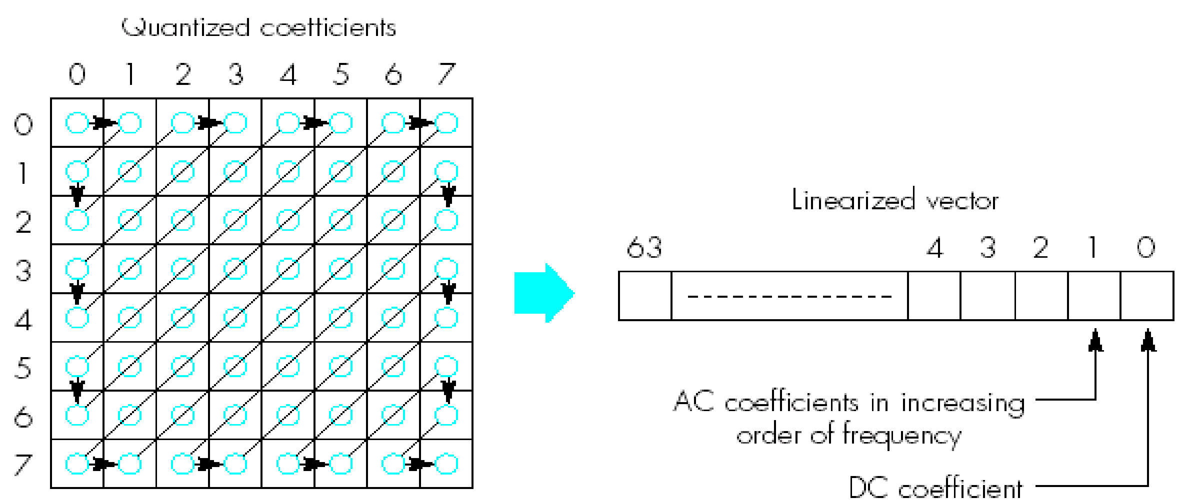
120	60	40	30	4	3	0	0
70	48	32	3	4	1	0	0
50	36	4	4	2	0	0	0
40	4	5	1	1	0	0	0
5	4	0	0	0	0	0	0
3	2	0	0	0	0	0	0
1	1	0	0	0	0	0	0
0	0	0	0	0	0	0	0

Quantized coefficients

12	6	3	2	0	0	0
7	3	2	0	0	0	0
3	2	0	0	0	0	0
2	0	0	0	0	0	0
0	0	0	0	0	0	0
0	0	0	0	0	0	0
0	0	0	0	0	0	0
0	0	0	0	0	0	0

10	10	15	20	25	30	35	40
10	15	20	25	30	35	40	50
15	.20	25	30	35	40	50	60
20	25	30	35	40	50	60	70
25	30	35	40	50	60	70	80
30	35	40	50	60	70	80	90
35	40	50	60	70	80	90	100
40	50	60	70	80	90	100	110

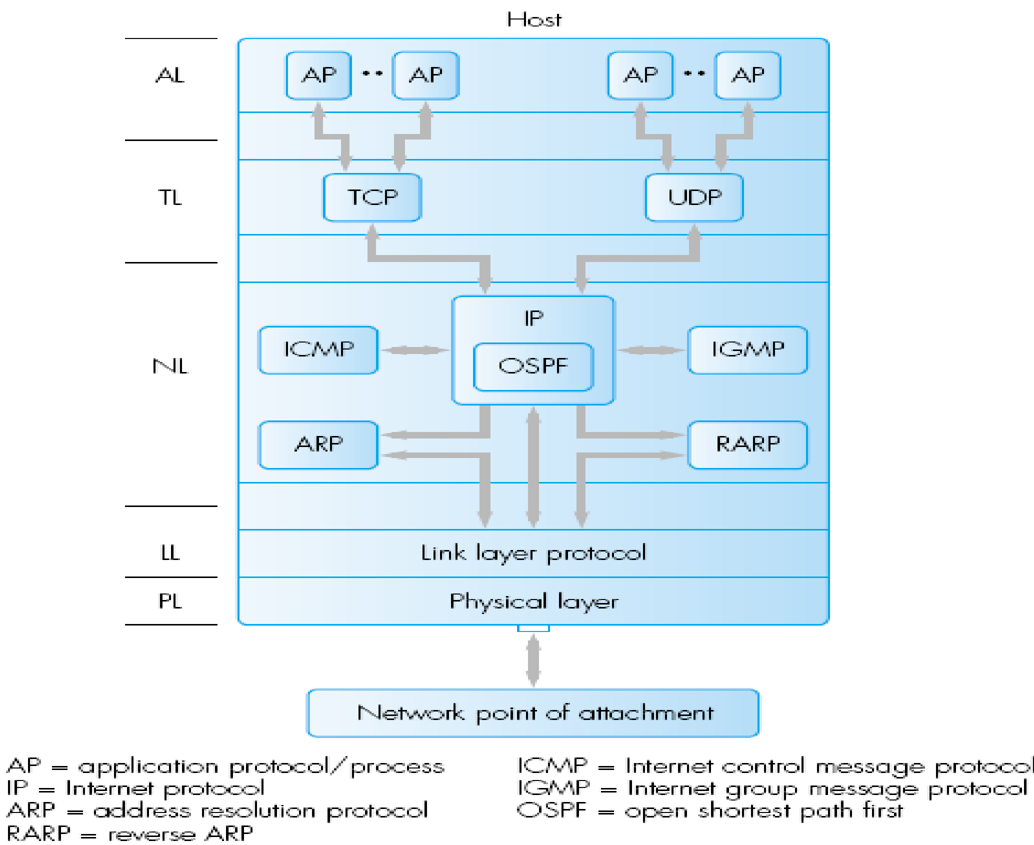
Quantization table



- In order to exploit the presence of the large number of zeros in the quantized matrix, a zig-zag of the matrix is used.   The remaining 63 values in the vector are the AC coefficients
- Because of the large number of 0's in the AC coefficients they are encoded as string of pairs of values
- Each pair is made up of (*skip, value*) where *skip* is the number of zeros in the run and *value* is the next non-zero coefficient. The above will be encoded as
- *(0,6) (0,7) (0,3)(0,3)(0,3) (0,2)(0,2)(0,2)(0,2)(0,0)*
- Final pair indicates the end of the string for this block
- Significant levels of compression can be obtained by replacing long strings of binary digits by a string of much shorter codeword.

- The length of each codeword is a function of its relative frequency of occurrence
- Normally, a table of codeword is used with the set of codeword precomputed using the Huffman coding algorithm

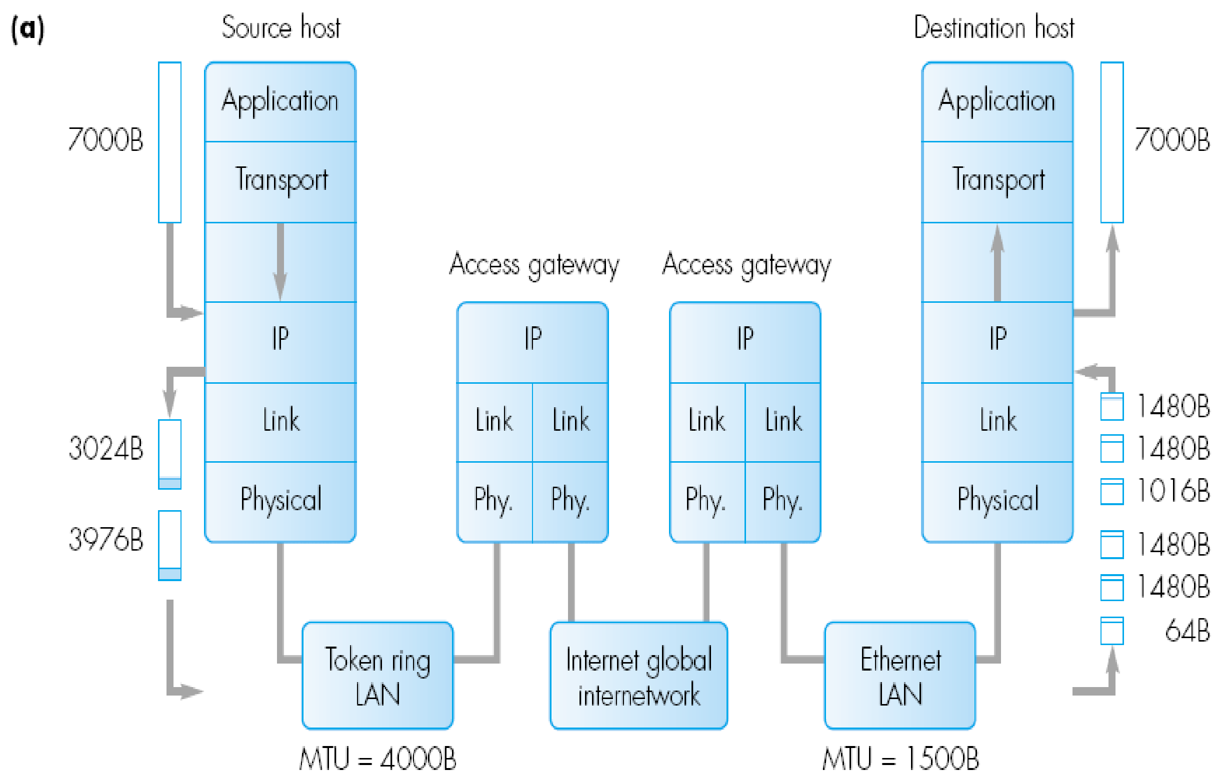
4. IP Adjunct Protocol (Diagram-3 M, Explanation-7 M)



- Address resolution protocol(ARP) and Reverse ARP(RARP)
- used by the IP in hosts that are attached to a broadcast LAN in order to determine the physical MAC addresses of the host or gateway given its IP address (ARP) and RARP the reverse function.
- Open shortest path first (OSPF) is an example of routing protocol used in the global internetwork. These are used to build up the contents of the routing table that is used to route packets.
- Internet Control message protocol(ICMP)
- used by the IP in a host or gateway to exchange error and other control messages with the IP in another host or gateway.
- Internet Group management protocol(IGMP)
- used with multicasting to enable a host to send a copy of datagram to the other host that are part of the same multicast group.

5. **Fragmentation and Reassembly (**Diagram-3 M, Explanation-5 M, Steps- 2 M)

- If the size of the packet is greater than the MTU of the destination access network, the IP in the destination gateway divides the information received in the packet into a number of smaller blocks known as fragments.
- Each fragment is then forwarded to the IP in the destination host in a separate packet the length of which is determined by the MTU of the access/intermediate network.
- The destination host IP then reassembles the fragments of information from each received packet to form the original submitted block of information.



Note: All values shown are the amounts of user data in each packet/frame in bytes

Consider the transport protocol in a host that is attached to a token ring LAN transferring a block of 7000bytes

MTU – 4000 BYTES FOR TOKEN RING LAN AND ETHERNET LAN- 1500 BYTES

HEADER OF EACH DATAGRAAM – 20BYTES

MAX USABLE DATA IN EACH TOKEN RING FRAME IS 4000-20= 3980 BYTES

EACH ETHERNET FRAME IS 1500-20= 1480 BYTES

ALL FRAGMENTS OF USER DAT MUST BE IN MULTIPLES OF 8 HENCE MAX USER DATA IN EACH PACKET TRANSFERRED OVER TOKEN RING IS 3976 BYTES

BUT FOR ETHERNET 1480 IS NOT CHANGED AS IT IS DIVISIBLE BY8

TRANSFER OF BLOCK OF 7000 BYTES OVER TOKEN RING REQUIRES TWO DATAGRAM ONE 3976 AND OTHER 7000-3976= 3024 BYTES

(b) packet header fields for token ring LAN;

(b) Token ring LAN:	(i)	(ii)
Identification	20	20
Total length	7000	7000
Fragment offset	0	497
(User data)	3976	3024
M-bit	1	0

(c) Ethernet LAN.

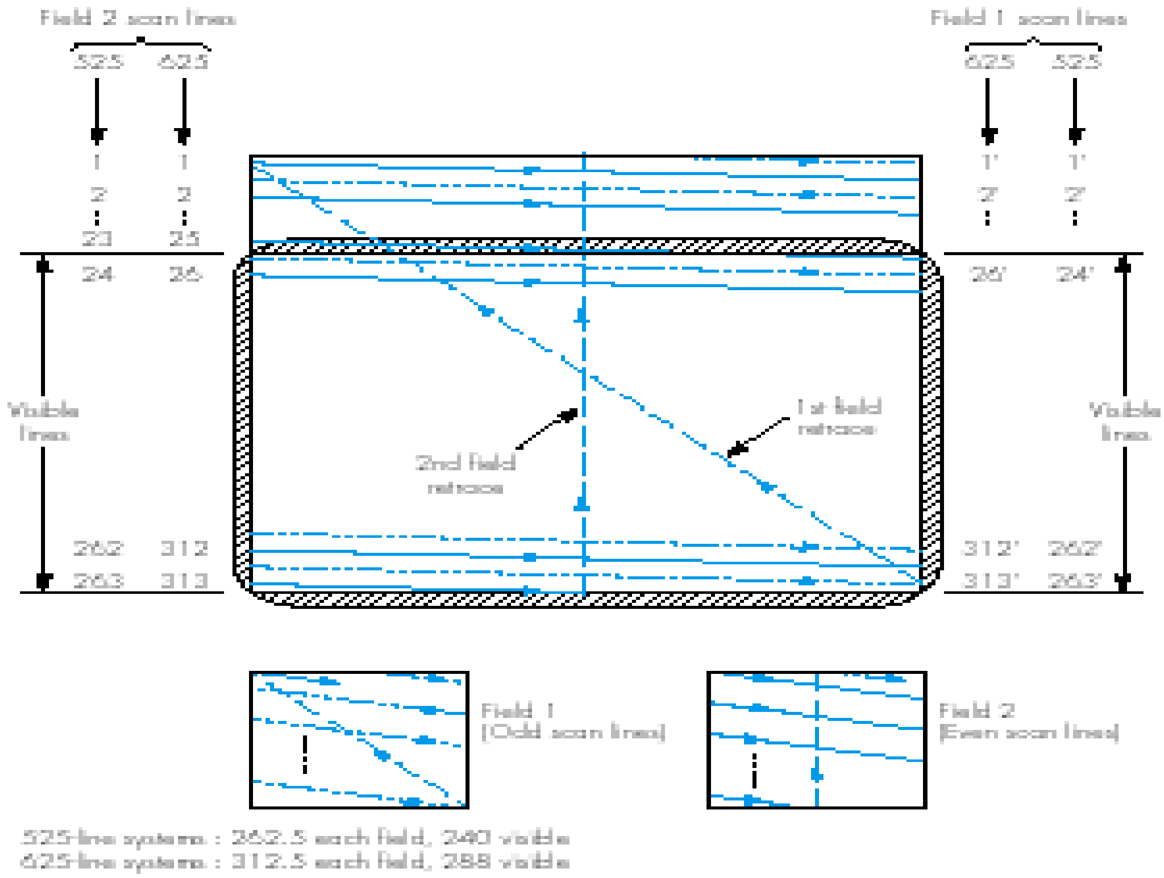
(c) Ethernet LAN:	(i)	(ii)	(iii)	(iv)	(v)	(vi)
Identification	20	20	20	20	20	20
Total length	7000	7000	7000	7000	7000	7000
Fragment offset	0	185	370	497	682	867
(User data)	1480	1480	1016	1480	1480	64
M-bit	1	1	1	1	1	0

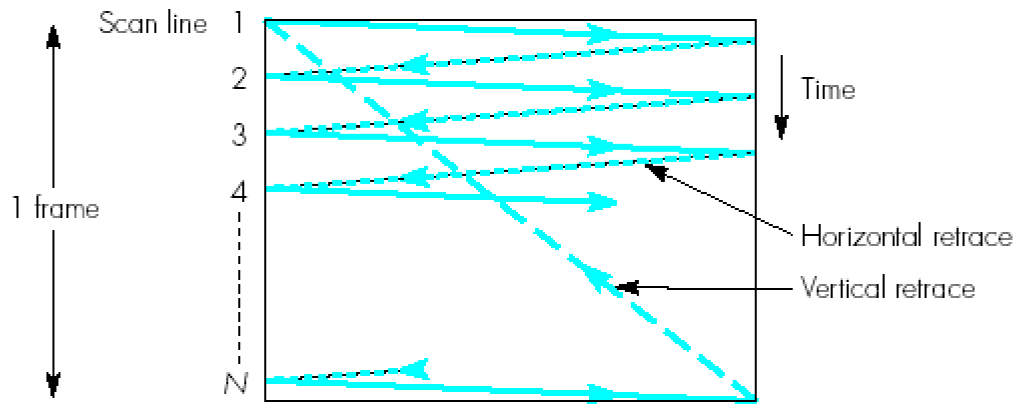
6. (a) **Interlaced Scanning (** Diagram-3 M,Explanation-2 M)

The picture tubes used in most television sets operate using what is known as a **raster-scan**; this involves a finely-focussed electron beam being scanned over the complete screen. *Progressive scanning* is performed by repeating the scanning operation that starts at the top left corner of the screen and ends at the bottom right corner follows by the beam being *deflected back* again to the top left corner. This is known as Non-Interlaced Scanning.

- It is necessary to use a minimum refresh rate of 50 times per second to avoid flicker. A refresh rate of 25 times per second is sufficient. The image/picture associated with each frame is transmitted in two halves and each is known as a Field. The first comprising only the odd scan lines and the second the even scan lines. The two fields are then integrated together in the television receiver using a technique known as interlaced scanning.

Figure 2.19 Interlaced scanning principles.





N = 525 (NTSC) and 625 (PAL/CCIR/SECAM)
Frame refresh rate = 60 times per second (NTSC)
= 50 times per second (PAL/CCIR/SECAM)

(b)

Assuming the bandwidth of a speech signal is from 50 Hz through to 10 kHz and that of a music signal is from 15 Hz through to 20 kHz, derive the bit rate that is generated by the digitization procedure in each case assuming the Nyquist sampling rate is used with 12 bits per sample for the speech signal and 16 bits per sample for the music signal. Derive the memory required to store a 10 minute passage of stereophonic music.

Answer:

(i) Bit rates: Nyquist sampling rate = $2 f_{max}$

Speech: Nyquist rate = 2×10 kHz = 20 kHz or 20 ksps
Hence with 12 bits per sample, bit rate generated
= $20\text{k} \times 12 = 240$ kbps

Music: Nyquist rate = 2×20 kHz = 40 kHz or 40 ksps
Hence bit rate generated = $40\text{k} \times 16 = 640$ kbps (mono)
or $2 \times 640\text{k} = 1280$ kbps (stereo)

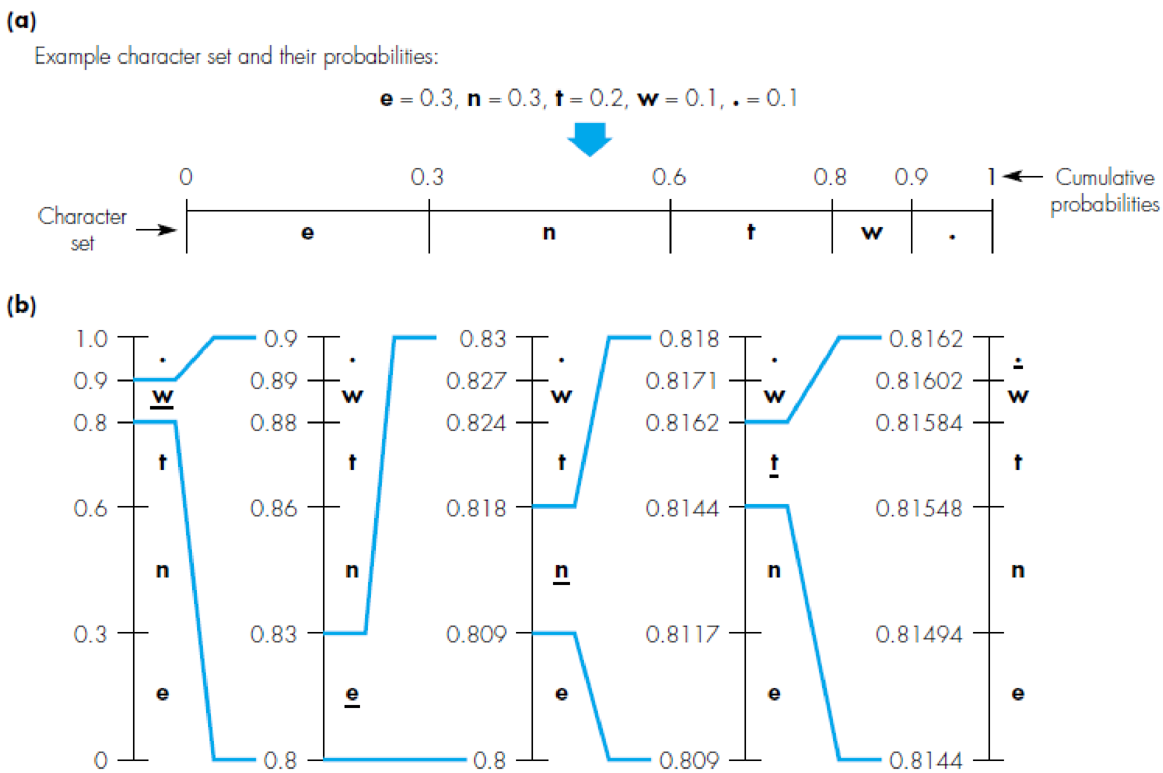
(ii) Memory required: Memory required = bit rate (bps) × time (s)/8 bytes
Hence at 1280 kbps and 600 s,

$$\text{Memory required} = \frac{1280 \times 10^3 \times 600}{8} = 96 \text{ Mbytes}$$

Calculation of Bit rates for Speech and Music (4 M); Memory requirement – (1 M)

7. **Arithmetic Coding**

Arithmetic coding is a coding method that usually outperforms Huffman coding. Huffman coding assigns each symbol a codeword which has a integral bit length. Arithmetic coding can treat the whole message as one unit.



Encoded version of the character string **went.** is a single codeword in the range 0.816 02 ≤ codeword < 0.8162

Definition of Arithmetic coding- 2 M, Problem with steps- 8M