CMR Institute of Technology
Department of ECE
17EC741 – Multimedia Communication
Scheme & Solution for 4th IAT – Feb 2022

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Internal Assessment Test 4 – February 2022

Sub:	Multimedia Communication				Sub Code:	15EC741/ 17EC741	Branch:	ECE/TCE
Date:	03/02/2022	Duration:	90 min's	Max Marks: 50	Sem / Sec:	VII E		OBE

Answer any FIVE FULL Questions

	Question	MARKS	CO	RBT
1.	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]	CO5	L3
2.	Discuss multimedia operating system with respect to CPU management, memory management, I/O management, file system management	[10]	CO5	L2
3.	Write short notes on (i) Lempel Ziv coding (ii) Lempel Ziv Welsh coding	[10]	CO5	L1
4.	What is DMS and explain the main features of DMS with a block diagram	[10]	CO5	L1
5.	Explain how better sound quality can be obtained by using sub band DPCM with the help of block diagram of encoder and decoder/Explain with a neat diagram, ADPCM sub band encoder and decoder	[10]	CO2	L2
6.	Explain H.261 video compression standard with the help of macro block frame format and GOB structure	[10]	CO2	L2
7.	Explain with relevant diagrams, sensitivity of the ear, frequency and temporal masking used in perceptual coding	[10]	CO2	L2

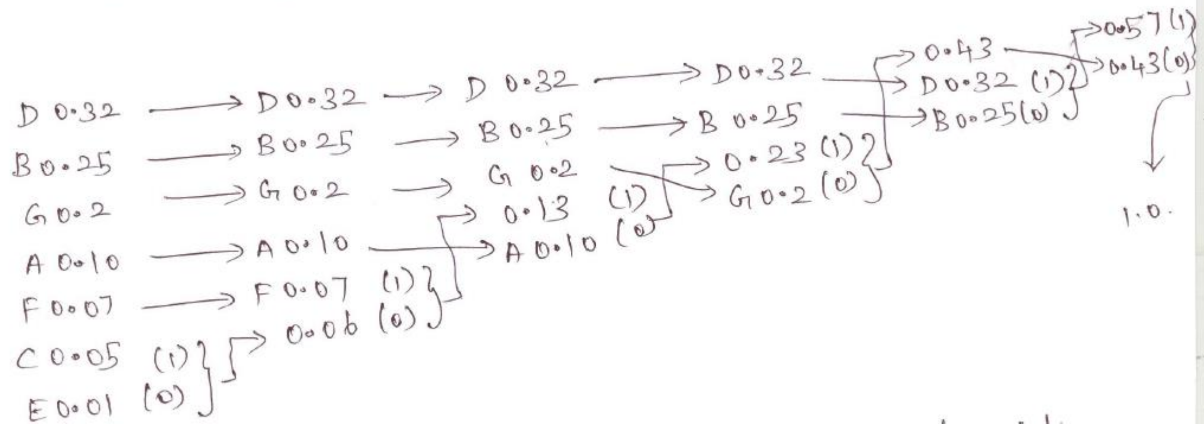
1.

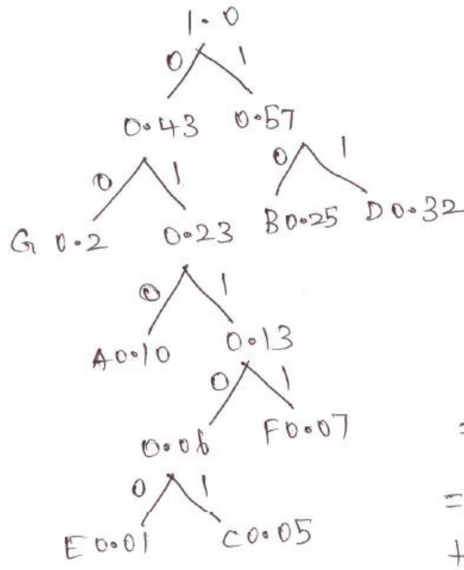
$$\text{Entropy } H = -\sum_{i=1}^{n} P_i \log_2 P_i$$

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

$$= -3.32[-0.1 - 0.15 - 0.065 - 0.158 - 0.02 - 0.080 - 0.139]$$

$$= -3.32[-0.712] = 2.36$$





Codewords

$A \rightarrow 010$

$B \rightarrow 10$

$C \rightarrow 01101$

$D \rightarrow 11$

$E \rightarrow 01100$

$F \rightarrow 0111$

$G \rightarrow 00$

Average no. of bits per codeword

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

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

$$= 0.3 + 0.5 + 0.25 + 0.64 + 0.05 + 0.28 + 0.4$$

$$= 2.42$$

2.

OSs manage computer resources, for example, CPU, memory, I/O devices, and so forth. They also hide the physical characteristics of the underlying hardware and provide an efficient and convenient environment for end-users. A multimedia OS extends the functionalities of OSs to accommodate multimedia data manipulation to provide an environment for real-time support. The major concerns include real-time support while simultaneously running traditional applications efficiently. The major concerns include real-time processing and QoS-based resource-management. A multimedia OS may be developed as an extension of a traditional OS or constructed using the microkernel architecture [4.22]. It should provide CPU management, memory management, I/O management and file system management.

CPU Management

Real-time processing can be achieved through efficient real-time scheduling. In the context of continuous media, a deadline can be the acceptable playback time of each frame. Therefore, it is a soft deadline and appears periodically. The challenges of multimedia scheduling are due to two conflicting goals: non-real-time processes and real-time processes. Non-real-time processes should not suffer from the execution of real-time processes, because multimedia applications equally depend on discrete and continuous media data. Real-time processes should be allowed to pre-empt non-real-time processes or other real-time processes with lower priorities.

The most important real-time scheduling approaches include Earliest Deadline First (EDF) and rate monitoring scheduling [4.23]. With EDF, each task is preemptive and is assigned a priority according to the deadline. The highest priority is assigned to the job with the earliest deadline, and tasks are executed in a priority order. When a new task arrives, the scheduler recomputes the priorities of all pending tasks and then reorganizes such that the order of the task being executed is preempted and the new task gets served immediately. The interrupted process is resumed later from the interruption point. Otherwise, the new task will be put in an appropriate position.

With rate-monotonic scheduling, each task is pre-empted and is assigned a priority according to the request rate. The highest priority is assigned to the job with the highest rate. In contrast to EDF, such assignments are performed only at the connection establishment time and are maintained through the lifetime of the connection. For preemptive periodic tasks, rate-monotonic scheduling is optimal in the sense that no other-static algorithm can schedule a task that the rate-monotonic algorithm cannot also schedule [4.24].

Comparing these two algorithms, EDF is more dynamic. It has to be executed frequently and thus incurs higher scheduling overhead. The advantage is that it can achieve processor utilization up to 100%. On the other hand, a rate-monotonic algorithm is static because the priority assignment is only calculated once. Because the priorities are assigned according to the request rate, more context switches occur in rate-monotonic scheduling than EDF. The worst-case upper bound of the process use is about 69% even though, on the average, the use is suitable for continuous media applications because it has no scheduling overhead and is optimal for periodic jobs.

Memory Management SB & PP

The memory manager allocates memory to processes. Continuous media data is typically very large in size and requires stringent timing requirements. One solution is to avoid swapping and to lock continuous media data in memory during the process [4.24]. This approach, however, may affect resource use. Other important practical implementation techniques include using scatter buffers and passing pointers. With scatter buffers or scatter loading, the address space of a process is loaded into possibly discontinuous regions of memory. This tends to be more space efficient than loading into a single continuous region, but may result in fragmentation. With passing pointers, objects are passed by reference rather than having to pass the objects themselves. This may result again in more efficient usage of memory space.

IO Management

The main function of the I/O subsystem is to transfer multimedia information between the main memory and the network adapter or multimedia peripherals (camera, loudspeaker, CD-ROM drive, microphone, disk, keyboard, monitor, and so forth). The important issues include device management, interrupt latency, and real-time transmission. Device management integrates all hardware components and provides a uniform interface for the control and management of these devices. Multimedia applications are I/O intensive. The continuous media frames will not frequently interrupt the kernel and lower the system throughput. There are three strategies to alleviate this problem: changing the internal structure of the kernel to make it highly preemptive, including a set of safe pre-emption points to the existing kernel or converting the current kernel to a user program and running on top of a microkernel [4.25]. Real-time I/O is necessary for most multimedia applications to ensure the continuity of each stream and the synchronization of multiple related streams. With advances in networking and communication technologies, it is possible to achieve network bandwidth well above a gigabit per second. The network I/O becomes the bottleneck that limits overall system performance. Therefore, the focus is to improve I/O system throughput.

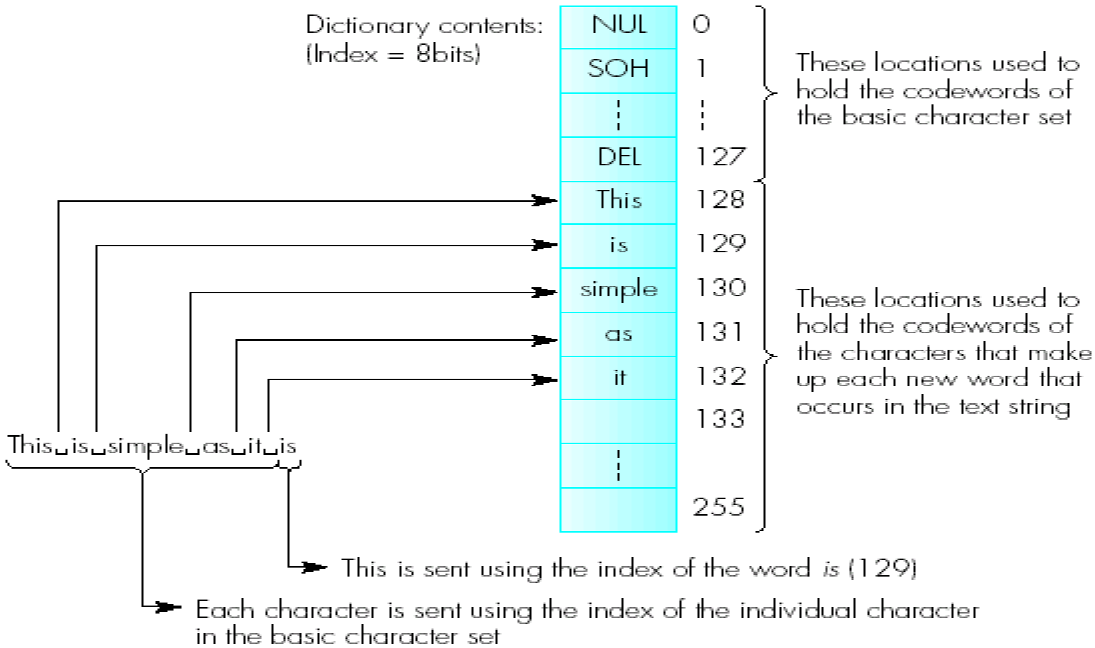
File System Management

File management is responsible for making efficient use of the storage capacity and for providing file access and control of stored data to the users. Multimedia file systems demand additional real-time requirements to handle continuous media streams.

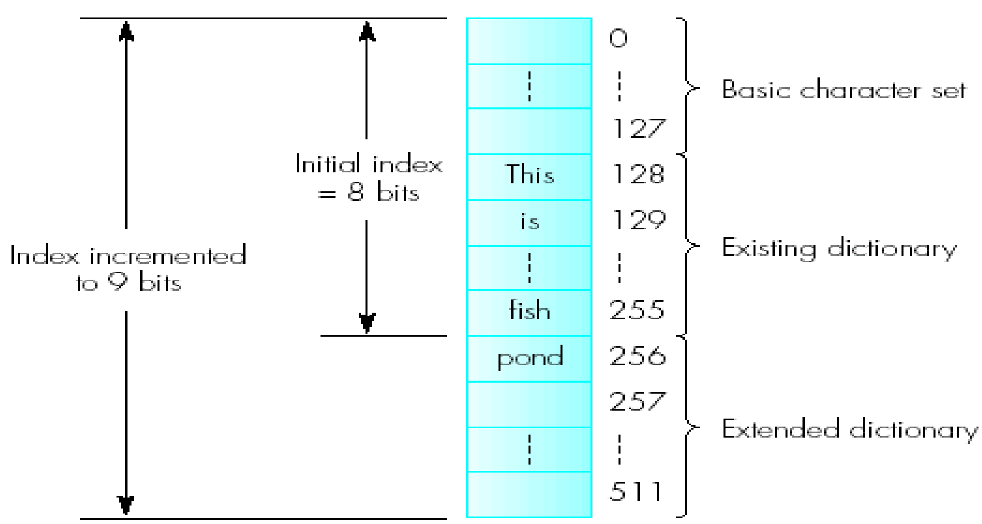
3. (i) Lempel Ziv Coding

- The LZ algorithm uses ***strings*** of characters instead of single characters
- For example for text transfer, a table containing all possible character strings are present in the encoder and the decoder
- As each word appears instead of sending the ASCII code, the encoder sends only the ***index*** of the word in the table
- This index value will be used by the decoder to reconstruct the text into its original form. This algorithm is also known as a ***dictionary-based*** compression

(ii) Lempel Ziv Welsh Coding



- The principle of the *Lempel-Ziv-Welsh* coding algorithm is for the encoder and decoder to build the ***contents of the dictionary dynamically*** as the text is being transferred
- Initially the decoder has only the character set – e.g ASCII. The remaining entries in the dictionary are *built dynamically* by the encoder and decoder
- Initially the encoder sends the index of the four characters T, H, I, S and sends the space character which will be detected as a non alphanumeric character
- It therefore transmits the character using its index as before but in addition interprets it as terminating the first word and this will be stored in the next free location in the dictionary
- Similar procedure is followed by both the encoder and decoder
- In applications with 128 characters initially the dictionary will start with 8 bits and 256 entries 128 for the characters and the rest 128 for the words



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A key issue in determining the level of compression that is achieved, is the **number of entries** in the dictionary since this determines the **number of bits** that are required for the index

4.

A DMS is an integrated communication, computing and information system that enables the processing, management, delivery and presentation of synchronized multimedia information that the quality of service guarantees [4.1, 4.2]. It integrates and manages the information communication and computing subsystems to realize multimedia applications. Such a system enhances human communications by exploiting both visual and aural senses and provides the ultimate flexibility in work and entertainment by allowing you to collaborate with remote participants, view movies on demand and access online digital libraries from the desktop. DMS will create an electronic world. Technological advances in computers, high-speed networks, data compression and consumer electronics—coupled with the availability of multimedia resource mechanism, and manipulation functions; the development of the relevant standards and the convergence of the computer, telecommunications, and digital TV industries—are accelerating the realization of such systems. An example of DMS is a number of multimedia PCs and/or workstations interconnected with continuous media servers usings the Internet that allow users to

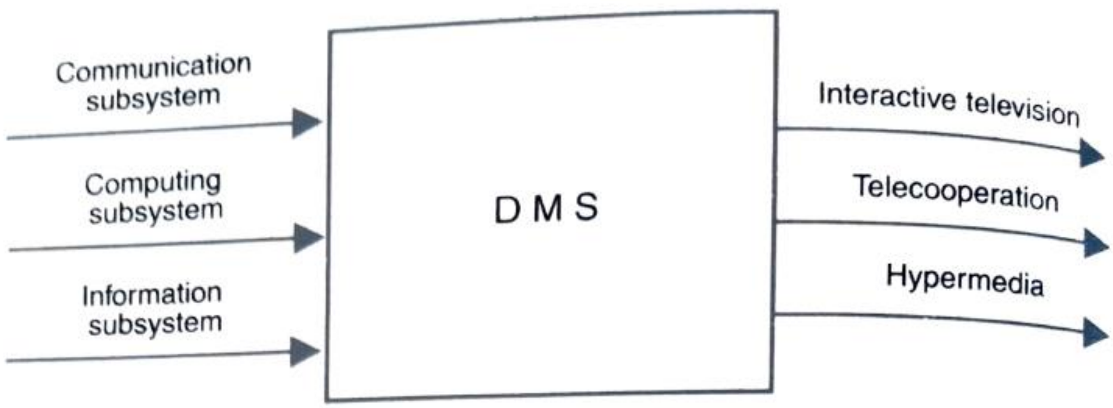


Figure 4.1 Block scheme of a summarized DMS.

retrieve, browse and manipulate video or audio. Besides constraints on bit error rates, packet-loss probabilities and delivery delays required in a point-to-point information delivery system, additional constraints are introduced in a DMS, such as the synchronization among multiple media streams from distributed sources to achieve a meaningful presentation. Figure 4.1 summarizes a DMS.

The inputs of the system consist of the factors that drive a DMS from concepts to reality, and the outputs consist of a wide range of distributed multimedia applications. The system inputs are a collection of the enabling technologies of the communication subsystem (for transmission), the computing subsystem (for processing) and information subsystem (for storage). The communication subsystem consists of the transmission medium and transport protocols. It connects the users with distributed multimedia resources and delivers multimedia materials with QoS guarantees, such as real-time delivery for video or audio data and error-free delivery for text data. The computing subsystem consists of a multimedia platform, operating system (OS), presentation and authoring tools and multimedia manipulation software. It allows users to manipulate the multimedia data. An authoring tool is specialized software that allows a producer or designer to design and assemble multimedia elements for a multimedia presentation. The information subsystem consists of the multimedia servers, information archives and multimedia database systems.

The outputs of the system can be broadly classified into three different types of distributed multimedia applications: Interactive Television (ITV), telecooperation and hypermedia. ITV allows subscribers to access video programs and interact with them. Services include home shopping, interactive video games (which can be classified as hypermedia applications), financial transactions, Video on Demand (VoD), news on demand and so forth. Telecooperation overcomes time and location restrictions and allows remote participants to join a group activity. Services include remote learning, telecommuting, teleservicing, teleoperation, multimedia emails, videophone, desktop conferencing, electronic meeting rooms, joint editing and group drawing. A hypermedia document is a multimedia document with links to other multimedia documents and it allows users to browse multimedia information in a consequential manner. Services include digital libraries, electronic encyclopedias, multimedia magazines, multimedia documents, information kiosks, computer-aided learning tools and the Web.

4.2 Main Features of a DMS

The main features of a DMS can be summarized as follows [4.3]:

- ***Technology integration***—Integrates information, communication and computing systems to form a unified digital processing environment.
- ***Multimedia integration***—Accommodates discrete data as well as continuous data in an integrated environment.
- ***Real-time performance***—Requires the storage systems processing systems and transmission systems to have real-time performance. Hence, huge storage volume, high network transmission rate and high CPU processing rate are required.
- ***Systemwide QoS support***—Supports diverse QoS requirements on an end-to-end basis along the data path from the sender, through the transport network and to the receiver.
- ***Interactivity***—Requires duplex communication between the user and the system and allows each user to control the information.
- ***Multimedia synchronization support***—Presents the playback continuity of media frames within a single continuous media stream, and temporal relationships among multiple related data objects.
- ***Standardization support***—Allows interoperability despite heterogeneity in the information content, presentation format, user interfaces, network protocols and consumer electronics.

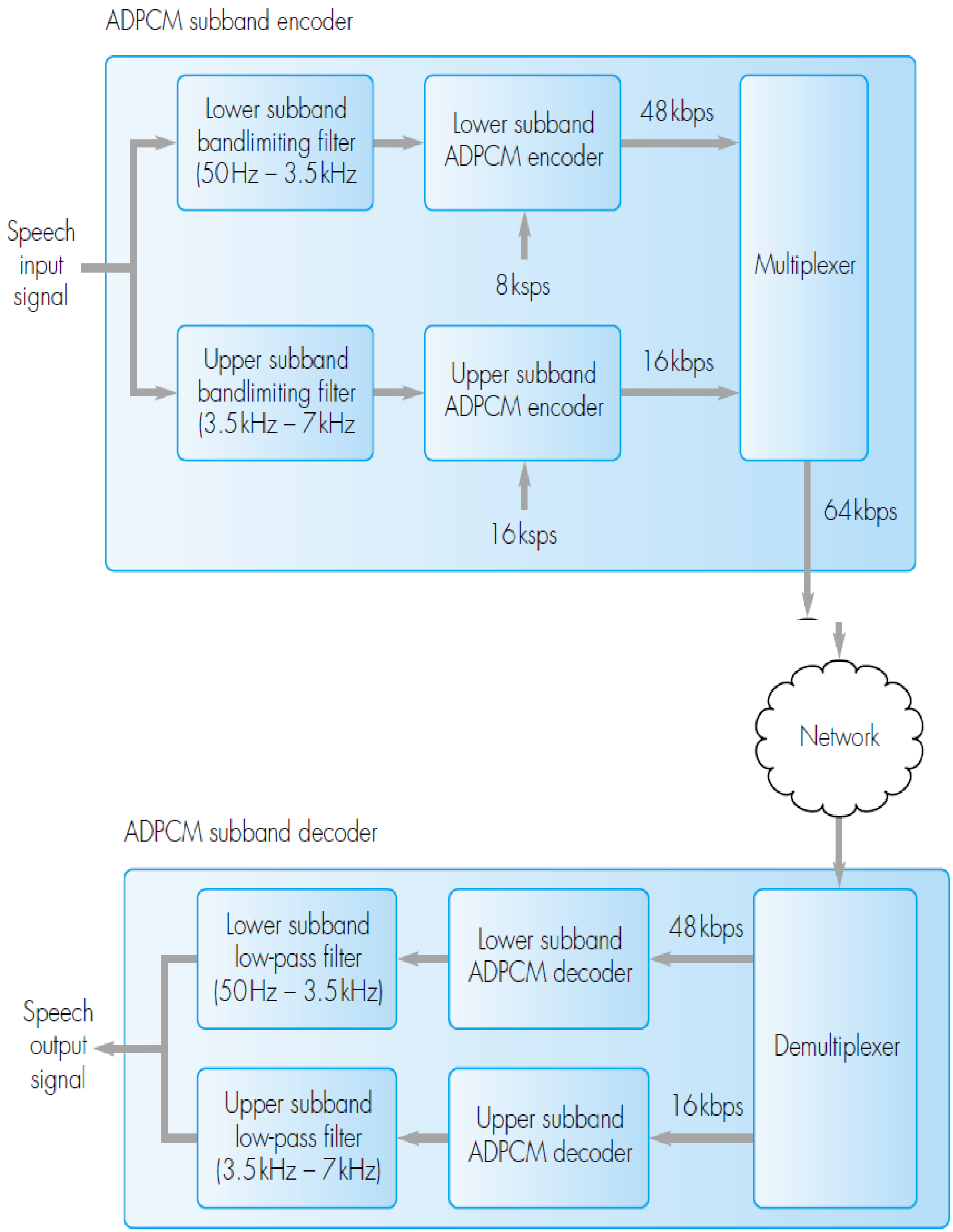
According to the different requirements imposed upon the information, communication and computing subsystems, distributed multimedia applications may be broadly classified into ITV, telecooperation and hypermedia. ITV requires a very high transmission rate and QoS guarantees. It demands point-to-point, switched connections as well as good customer services and excellent management for information sharing, billing and security. Telecooperation, such as videophone and desktop conferencing, allows lower picture quality and therefore has a lower bandwidth requirement. It requires powerful multimedia database systems rather than just continuous media servers. Sharing information among groups is the key to effective collaboration. Hypermedia systems may be treated as an application of database systems because they provide flexible access to multimedia information and a novel method to structure and manage data. Hypermedia applications require point-to-point and switched services.

5. **ADPCM:**

   Additional savings in bandwidth –or improved quality –can be obtained by varying the number of bits used for the difference signal depending on its amplitude

   A second ADPCM standard, which is G.722.It has added subband coding.better sound quality

   A third standard based on ADPCM is also available, this is defined in G.726.This also uses subband coding but with a speech bandwidth of 3.4kHz


ADPCM subband encoder
Speech input signal
Lower subband bandlimiting filter (50 Hz – 3.5 kHz
Upper subband bandlimiting filter (3.5 kHz – 7 kHz
Lower subband ADPCM encoder
Upper subband ADPCM encoder
8 ksps
16 ksps
48 kbps
16 kbps
Multiplexer
64 kbps
Network
ADPCM subband decoder
Demultiplexer
48 kbps
16 kbps
Lower subband ADPCM decoder
Upper subband ADPCM decoder
Lower subband low-pass filter (50 Hz – 3.5 kHz)
Upper subband low-pass filter (3.5 kHz – 7 kHz)
Speech output signal

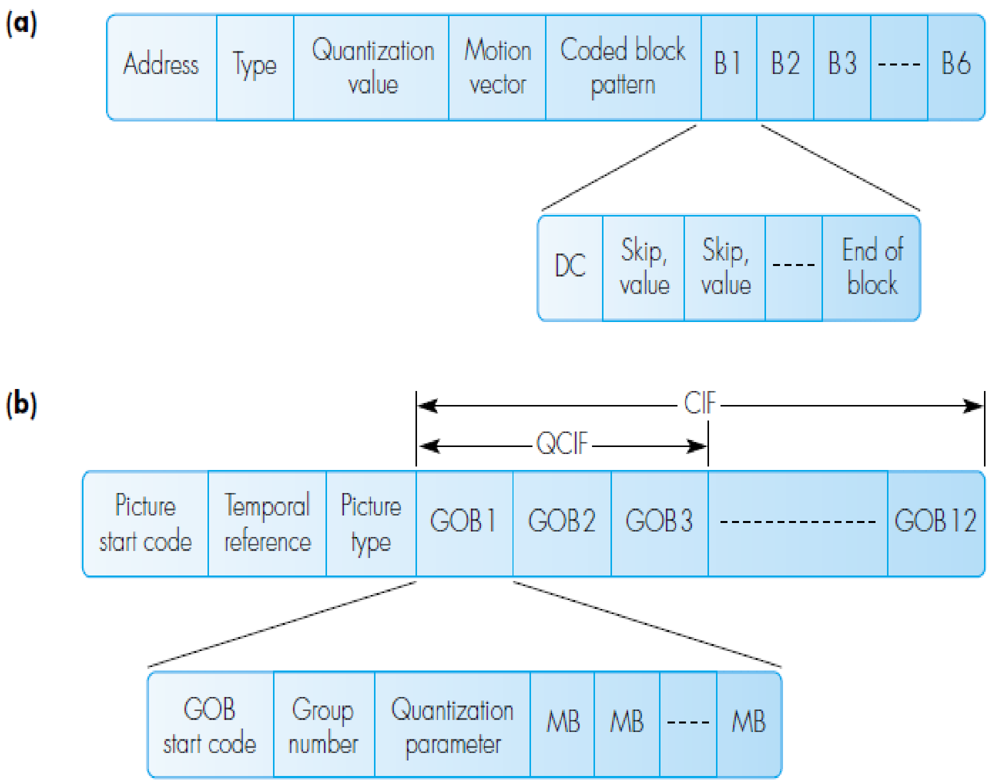
For higher signal bandwidth, before sampling the i/p signal is passed through filters.
The o/p of filters are lower band signal and upper sub band signal
It is sampled and encoded independently using ADPCM
Two bitstreams are multiplexed to produce transmitted signal and the decoder divides into separate stream for decoding

6. **H.261**

For the provision of video telephony and videoconferencing services over an ISDN
Transmission channels multiples of 64kbps
Digitization format used is either the common intermediate format (CIF) or the quarter CIF (QCIF)
Progressive scanning used with frame refresh rate of 30fps for CIF and 15or 7.5fps for QCIF

a. CIF:Y=352X288, Cb=Cr=176X144
b. QCIF:Y=176X144, Cb=Cr=88X72

I Frame and pframes are used with 3 p frames between each pair of I frames
Each macroblock has an address for identification
Type field indicates the macroblock is intracoded or intercoded
Quantization value is threshold value and mv is the encoded vector
Coded block pattern defines which of six 8x8 pixel block make up macroblock and the JPEG encoded DCT COEFFICIENTS are given in each block
Picture start code- Start of each video frame
Temporal ref field- time stamp to synchronize video block with the associated audio block of the same time stamp
Picture type field- type of frame (I or P frame)
GOB- GROUP OF MACROBLOCKS (size is chosen such that CIF and QCIF has integral number of GOBs)
EACH GOB – Unique start code – resynchronization marker
Each GOB also has group no.
For bandwidth optimization variable bit rate of encoder is converted into const bit rate
By passing through FIFO buffer
Feedback is provided to quantizer
o/p of the buffer is defined by the transmission bit rate, two threshold values are defined low and high
If contents of buffer is below the low threshold ,quantization threshold is reduced and the o/p rate is increased, if it is above high threshold then the threshold is increased and the o/p rate is reduced
Control proceedure is implemented for GOB


(a)
Address
Type
Quantization value
Motion vector
Coded block pattern
B1
B2
B3

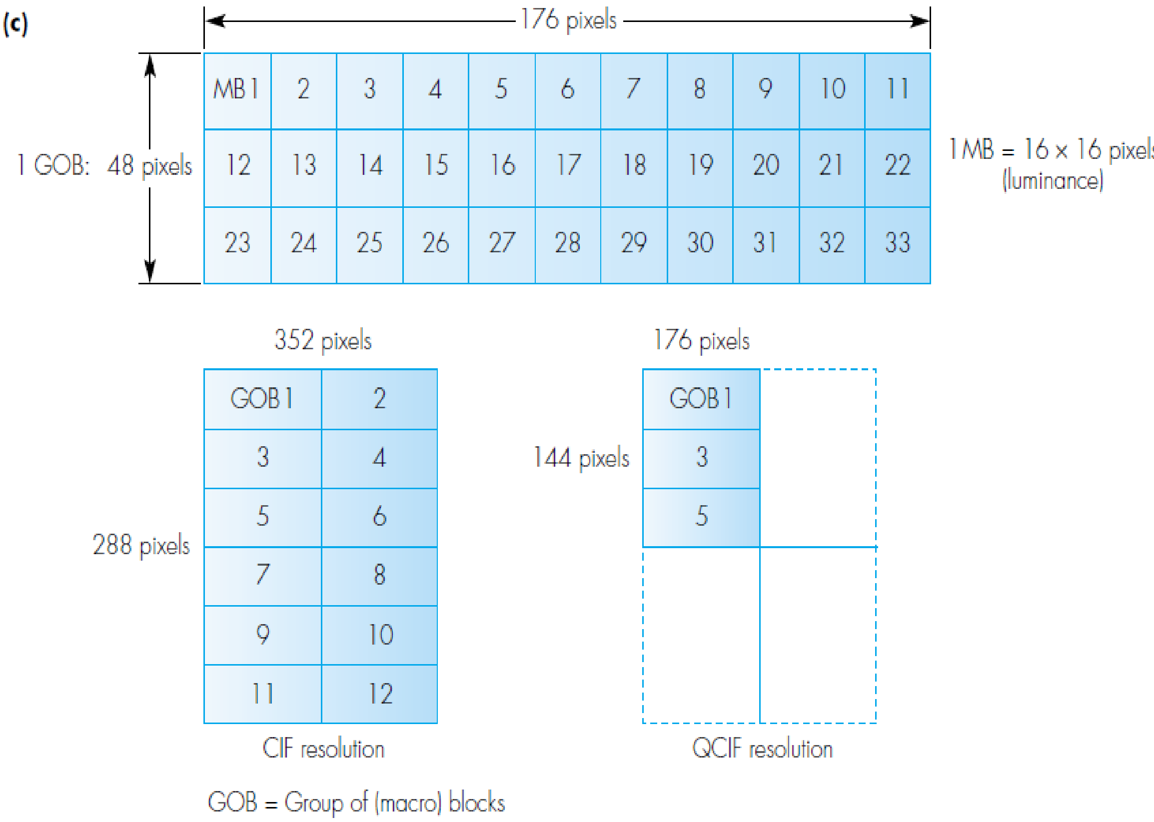
B6
DC
Skip, value
Skip, value

End of block
(b)
CIF
QCIF
Picture start code
Temporal reference
Picture type
GOB1
GOB2
GOB3

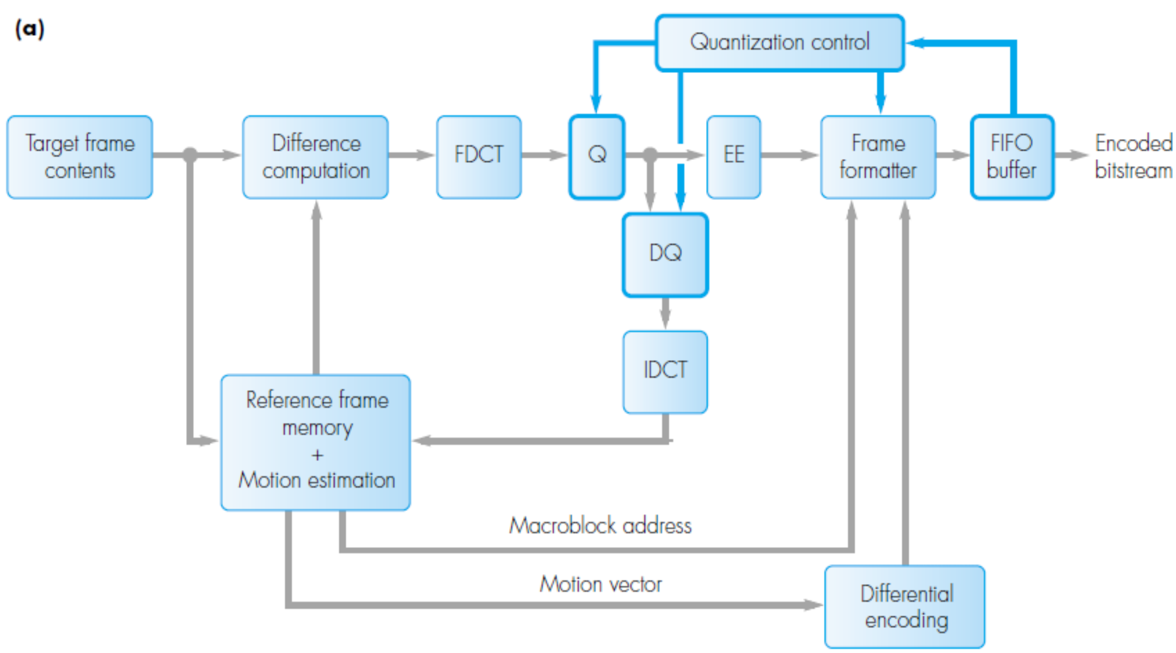
GOB12
GOB start code
Group number
Quantization parameter
MB
MB

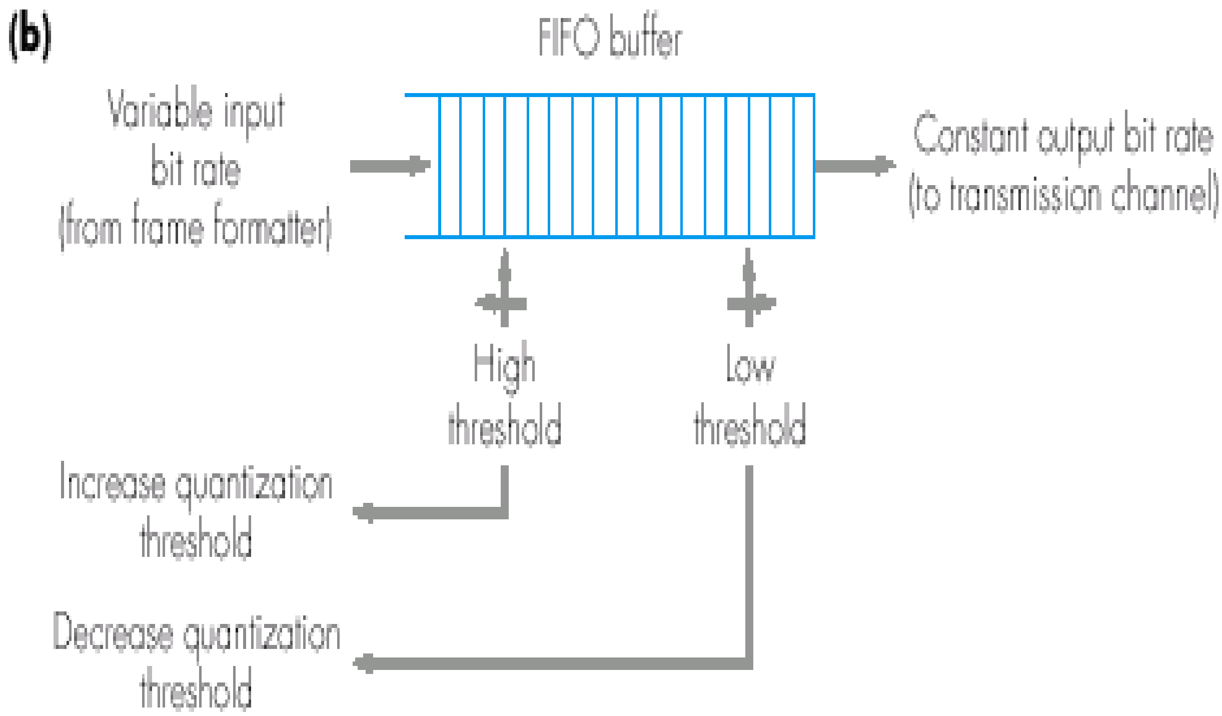
MB

(c)


176 pixels
MB 1
2
3
4
5
6
7
8
9
10
11
12
13
14
15
16
17
18
19
20
21
22
23
24
25
26
27
28
29
30
31
32
33
1 GOB: 48 pixels
1 MB = 16 × 16 pixels (luminance)
352 pixels
GOB 1
2
3
4
5
6
7
8
9
10
11
12
288 pixels
CIF resolution
176 pixels
144 pixels
GOB 1
3
5
QCIF resolution
GOB = Group of (macro) blocks

(a)

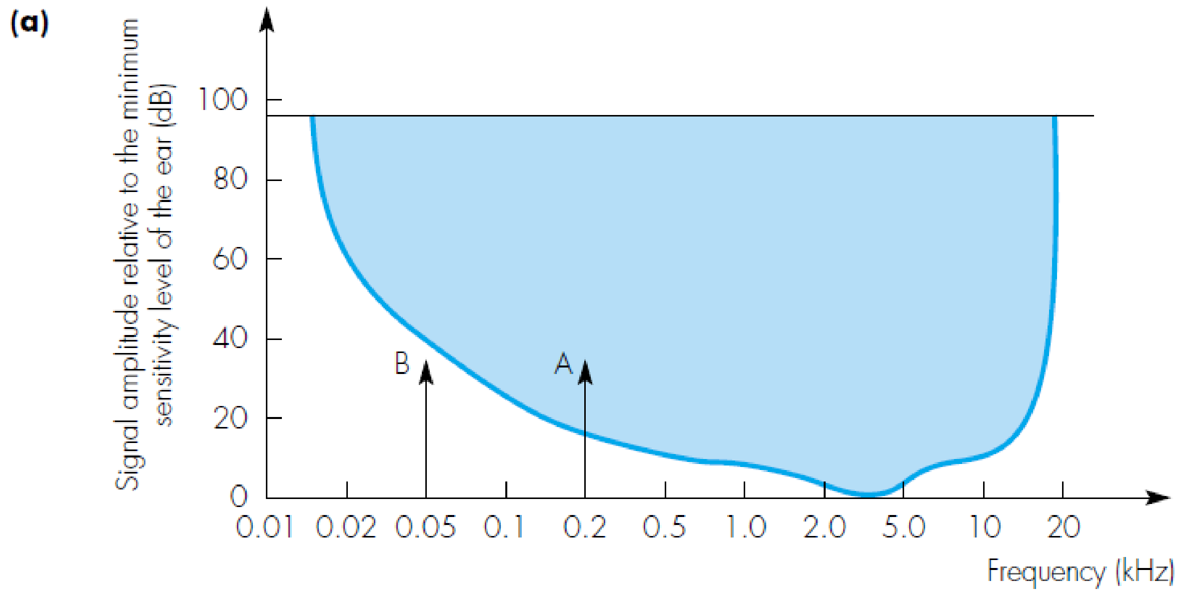

Quantization control
Target frame contents
Difference computation
FDCT
Q
EE
Frame formatter
FIFO buffer
Encoded bitstream
DQ
IDCT
Reference frame memory + Motion estimation
Macroblock address
Motion vector
Differential encoding



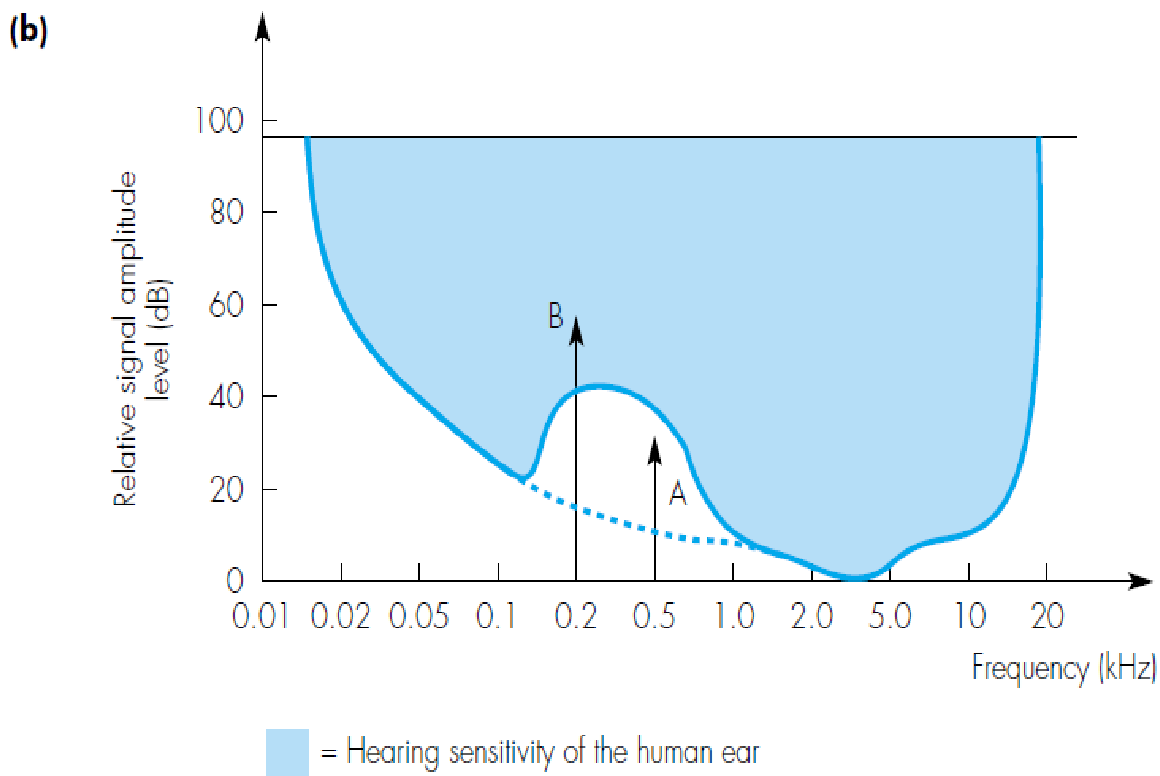
7. **Perceptual Coding**

   Perceptual encoders have been designed for the compression of general audio
   Perceptual coding since its role is to exploit a number of the limitation of the human ear.
   Sensitivity of the ear

   a. A strong signal may reduce the level of sensitivity of the ear to other signals which are near to it in frequency
   b. The model used is Psychoaccoustic
      The Sensitivity of the ear varies with the frequency of the signal,the perception threshold of the ear – that is, its minimum level of sensitivity-as a function of frequency is show in figure 4.5(a)
      Most sensitive to signals in the range 2-5kHz
      Fig a- vertical axis – amplitude of all signals to be heard , A and B have same level , A is heard due to threshold level



Shown 4.5(b) shows how the the sensitivity of the ear changes in the vicinity of a loud signal
Frequency masking- when multiple signals present , strong signal s reduce the sensitivity level of ear to other signals in the nearer freq



(b) frequency masking

- The masking effect also varies with frequency as show in figure 4.6
- Critical bandwidth- width of each curve at a particular signal level for that freq . For freq less than 500 hz critical bw is 100 hz and greater than 500 it changes .

- Temporal masking:
  - When the ear hears a loud sound,it takes a short but finite time before it can hear a quieter sound
  - Masking effect varies with freq-fig 4.6
  - SHOWN IN 4.7 –effect of temporal masking – signal amplitude decays after a time period after the loud sound ceases and at this time signal amplitude less than decay envelope will not be heard.
  - The masking effect also varies with frequency as show in figure 4.6
  - Critical bandwidth
- Temporal masking:
  - When the ear hears a loud sound,it takes a short but finite time before it can hear a quieter sound
  - SHOW 4.7

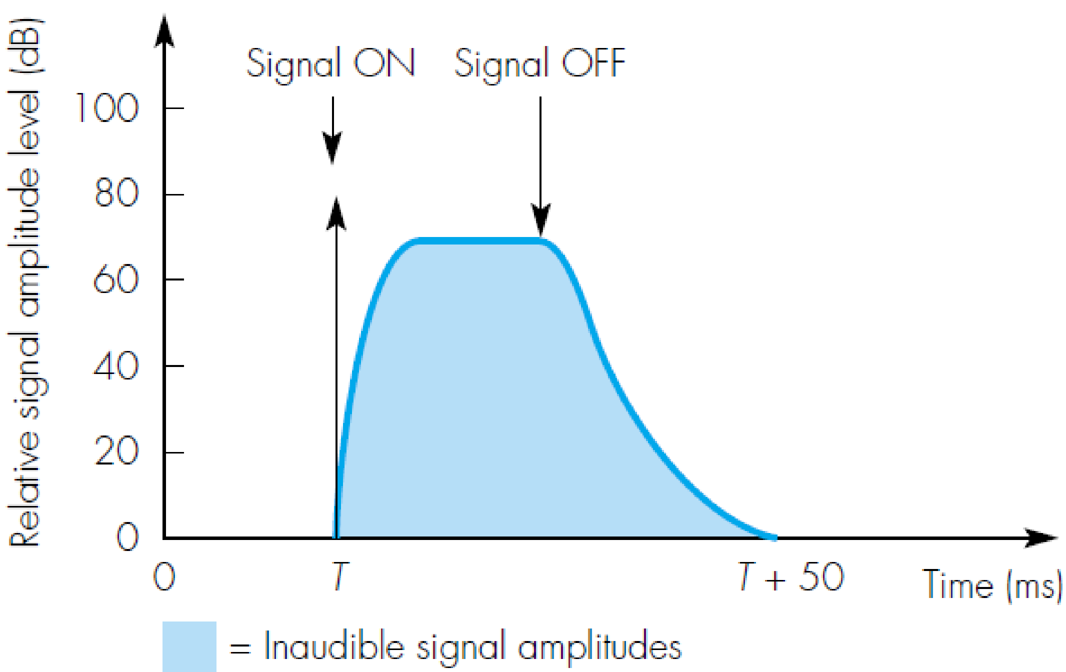


Figure 4.7 Temporal masking caused by a loud signal.