CMR INSTITUTE OF TECHNOLOGY

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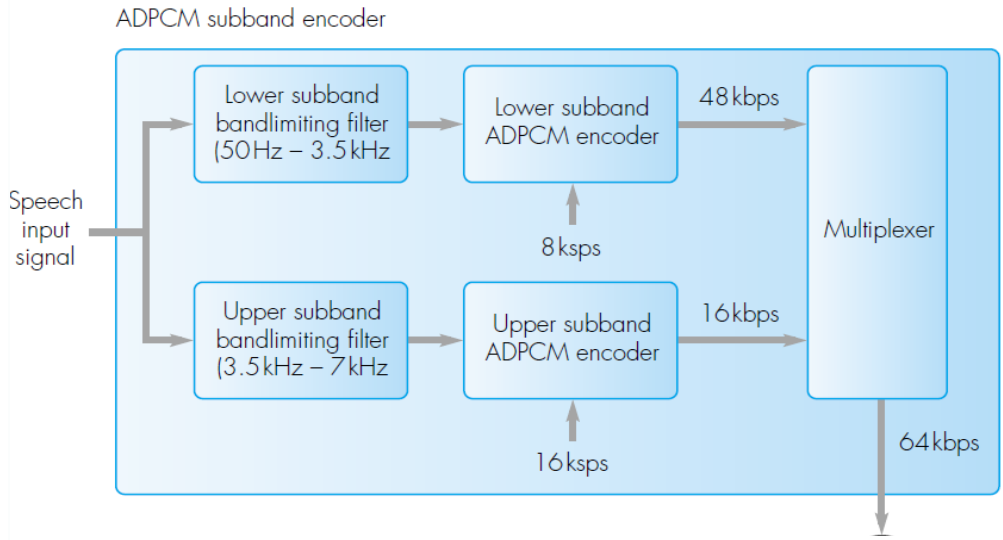
Internal Assesment Test - III

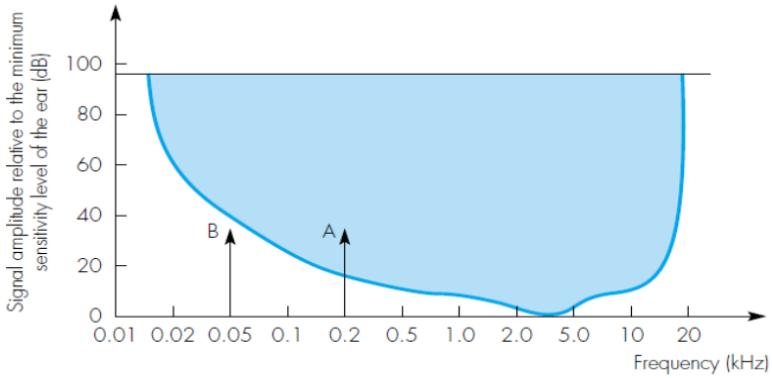
Sub:	Multimedia Communication							Code:	17EC741
Date:	27//01/2022	Duration:	90 mins	Max Marks:	50	Sem:	VII	Branch:	ECE-A,B,C,D

Answer Any FIVE FULL Questions

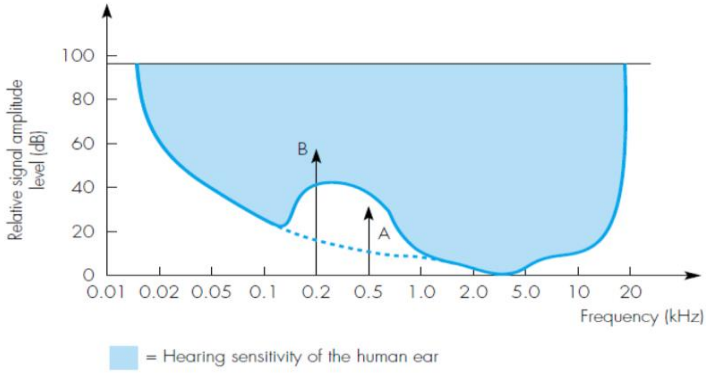
		Marks	OBE CO	OBE RBT
1.	Explain with a neat diagram, ADPCM sub-band encoder and decoder. ▪ Additional savings in bandwidth – or improved quality –obtained by ***varying the no. of bits used for the difference signal depending on its amplitude.*** ▪ So ***fewer bits to encode –*** transmitting ***smaller difference*** values for larger values. ▪ **ITU-T** Recommendations **– G.721 - *8-order predictor*** is used and the no. of bits used to quantize each difference value is varied – ***6-bits giving 32kbps***. OR ***5- bits giving 16kbps.*** ▪ **G.722 -** better sound quality with added complexity – technique added ***subband coding, giving wider bandwidth producing higher fidelity speech signal.- 6.9kHz from 3.4kHz.*** ▪ For higher BW – prior to sampling, input signal is divided into ***2 separate equal BW signal*** by passing through 2 filters.  ▪ Filter BW = 50Hz- 3.5kHZ – ***lower subband*** and 3.5kHz-7kHz – ***upper subband signal***. ▪ Each is sampled separately – different bit rate is used for each. Lower subband is encoded at ***48kbps*** & upper subband at ***16kbps***. ***64kbps*** output is produced by ***multiplexing*** both bitstreams.	[10]	CO2	L2
2.	Explain with relevant diagrams, sensitivity of the ear, frequency and temporal masking used in perceptual coding. ▪ Perceptual coding – used for **general audio, TV signal. – Lossy compression.** ▪ Perceptual coding uses ***Psychoacoustic model -*** It exploits a ***no. of the limitation of the human ear. / Imperfections of human ear.*** ▪ Perceptual audio coding is based *on 2 related characteristics* of the human ear: ▪ *Human ear sensitivity is not the same for all frequencies, varies with frequency.* ▪ *In the presence of multiple tones, a loud tone /noise can make a weaker tone inaudible - masking effect –* ***Frequency Masking.*** ▪ When the ear hears a loud sound, it takes a short but finite time before it can hear a	[10] [10] [5+5] [10]	CO2 CO4 CO2 CO2	L2 L2 L1 L1

quieter sound – ***Temporal Masking***

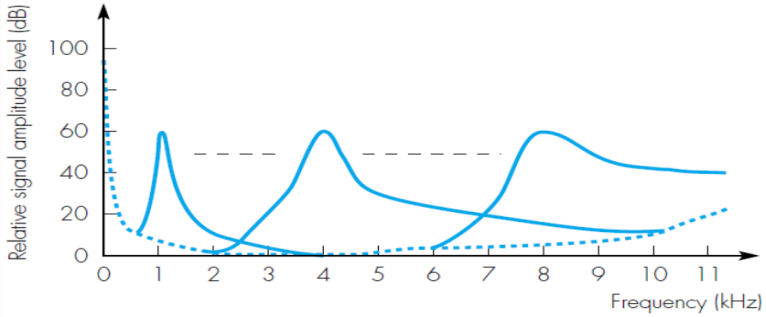
- ***Psychoacoustic model*** – used to *identify those signals* that suffer from both these effects, and are *eliminated* from the transmitted signal to reduce the amount of data transmitted.
- **Sensitivity of the ear –** *dynamic range of ear = loudest sound it can hear / quietest sound = 96dB.*
- For single frequency signal – ***perception threshold of ear****: i.e. minimum level of sensitivity* as a function of frequency is shown.
- Ear is most sensitive **2-5kHz** – signal in this band is the ***quietest the ear is sensitive to.***



- Vertical axis – amplitudes of all signals to be heard. A & B have same level, A is heard as it is above the hearing threshold and B is not.
- In ***presence of multiple frequency signal*** *- sensitivity of ear changes and varies with the relative amplitude of signal.*
- How sensitivity of ear changes in the vicinity of loud sound.
- B has larger amplitude than A, *ear sensitivity curve is distorted near B*.
- A is not heard even if it is above threshold.



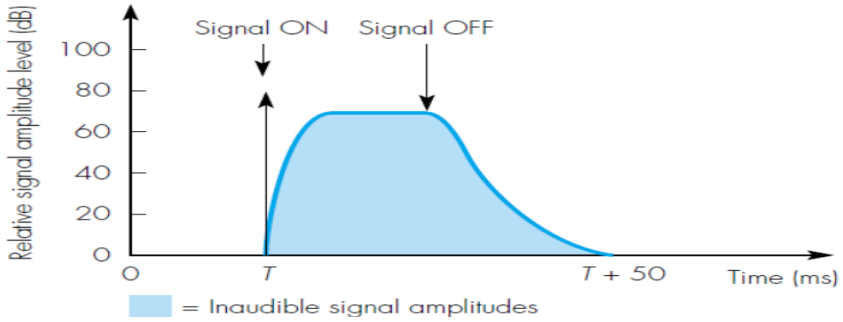
- Diagram shows masking effects for frequencies – 1, 4, 8kHz.
- Width of the masking curve – range of affected frequencies, increases with increase frequency.
- **Critical bandwidth** - width of each curve at a particular signal level for that frequency. For frequency **<** 500Hz, CBW =100Hz.



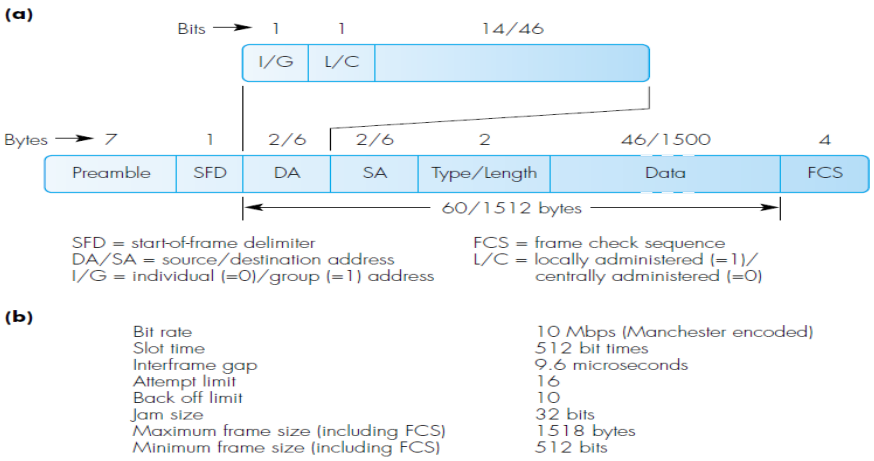
- **Temporal masking** - When the ear hears a loud sound, it takes a short but finite time before it can hear a quieter sound.

[10] | CO2 | L1

- Masking effect varies with frequency.
- Effect of temporal masking – after a loud sound ceases, signal *amplitude decays after a time period.*
- At this time signal ***amplitude less than decay envelope will not be heard*** and so need not be transmitted.



3. Draw and explain Ethernet frame format for data transmission.



(b)

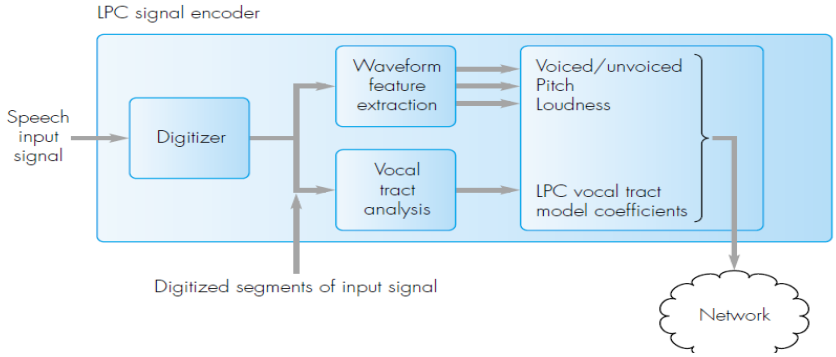
Bit rate	10 Mbps (Manchester encoded)
Slot time	512 bit times
Interframe gap	9.6 microseconds
Attempt limit	16
Back off limit	10
Jam size	32 bits
Maximum frame size (including FCS)	1518 bytes
Minimum frame size (including FCS)	512 bits

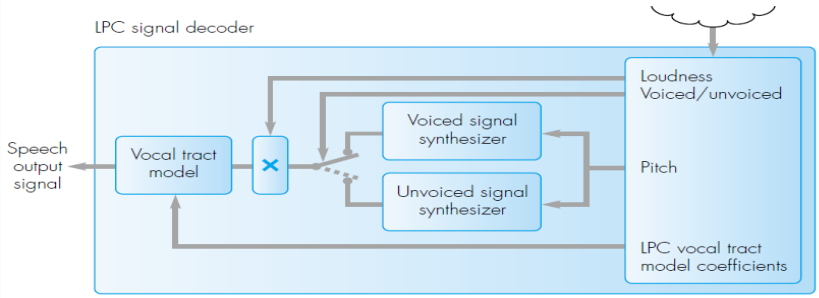
<u>Frame format:</u>
Preamble field- sent at the head of all frames, for synchronization of bits
Start of frame delimiter after preamble, single byte and informs the valid frame start.
Destination and source address – MAC address as used by MAC layer
First bit in the destination address specifies the address is individual or group address
Type of grouping is specified in second bit and can be locally or centrally administered
Group address is used for multicasting
Two byte type field indicates network layer protocol
Length field indicates number of bytes in the data field
Maximum size of data field – MTU (Maximum Transmission Unit)
FCS – Frame Check Sequence used for error detection

4. Describe the linear predictive coding for audio along with suitable diagram.

- DSP circuits help in analyzing the signal based on required **perceptual features**, then quantize and transmit.
- 3 feature which determine the perception of a signal by the ear are its:
  - **Pitch** – closely related to signal ***frequency*** – ear is more sensitive to 2-5kHz.
  - **Period** – signal ***duration.***
  - **Loudness** – amount of signal ***energy.***
- Origin of sound is also important – **vocal tract excitation parameters**.
  - **Voiced sounds** - generated through vocal chords – e.g. **– m, v,**
  - **Unvoiced sounds** – vocal chords are open – e.g. – **f, s**
  - Input speech is sampled and quantized at a defined rate.

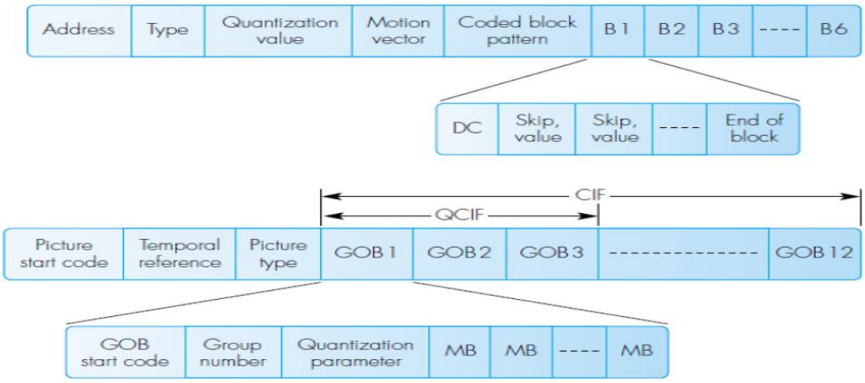
- A **block of digitized samples – segment** are analyzed to determine the various perceptual parameters.
- These are used with proper **model of vocal tract** to produce synthesized version of original speech.
- Destination uses them, together with a **sound synthesizer**, to regenerate a sound that is perceptually comparable with the source audio signal.
- Decoder uses the **current set of model coefficients + linear combination of previous set of model coefficients.**
- Low bit rate of **2.4kbps or 1.2kbps.**
- Output is highly **synthetic voice** used for **Military** applications.



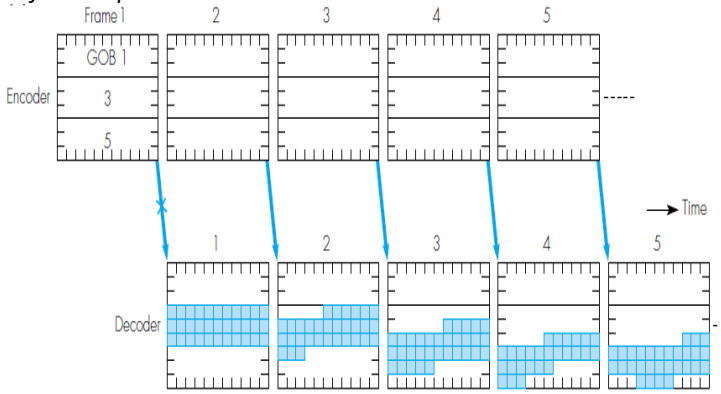


5. Write Notes on the following video compressions: (i) H.261 (ii) H.263

- **Application** - ***video telephony*** and ***video conferencing*** services over an ISDN. Transmission channels multiples of **64kbps – px64.**
- ***Digitization format –*** either
  - ***CIF - for video conferencing & QCIF – for video telephony***
  - ***Progressive scanning*** used with frame ***refresh rate = 30fps*** for CIF and ***15or 7.5fps*** for QCIF. ***CIF: Y=352x288, Cb=Cr=176x144 & QCIF: Y=176x144, Cb=Cr=88x72***
- H.261 encoding format – I- & P-frames are used with 3 P frames between each pair of I-frames.
- Each macroblock has an address for identification.

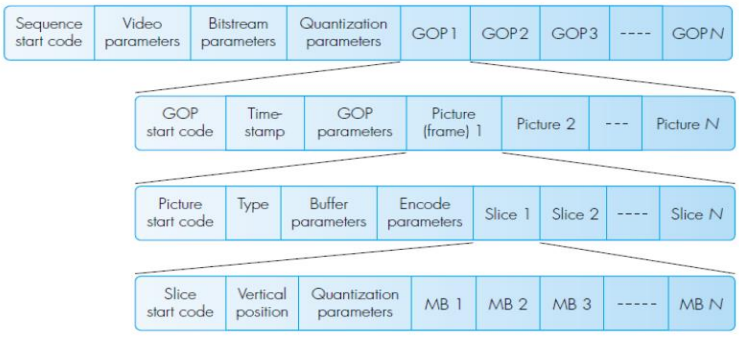


- ***Type field*** - macroblock is intra-coded or inter-coded.

- ***Quantization value*** -threshold value used to quantize all DCT coefficients.
- ***Motion vector*** - is the encoded vector.
- ***Coded block pattern*** **-** defines which of the *6 8x8 pixel block* make up macroblock are present.
- ***Picture start code-*** *Start of each new* video frame/picture.
- ***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 are also defined.
- Each GOB – Unique start code – **resynchronization marker**
- For bandwidth optimization variable bit rate of encoder is converted into constant 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 procedure is implemented for GOB.
- Used in video applications over **wireless** and public switched telephone networks (**PSTN**).
- Include - ***video telephony, videoconferencing, security surveillance, interactive game***.
- Low bit rates – ***28.8 – 56 kbps***.
- ***Digitization formats***
- ***QCIF: Y = 176x144, Cb = Cr = 88x72 & S-QCIF: Y = 128x96, Cb = Cr = 64x68***
- Progressive scanning with ***frame rate of 15 or 7.5 fps***.
- Frame types: ***I-frame, P-frame, B-frame***
- ***PB-frame*****:** because of the much-reduced encoding overhead.
- ***Mv*** associated with predicted macroblocks are restricted to defined area and the **search area is restricted to the edge of the frame.**
- To overcome this limitation, for those pixels of a potential close-match macroblock that *fall outside of the frame boundary edge pixels* are used.
- **Error resilience** - Cause error propagation.
- For PSTN, errors present in bitstream is difficult to find
- GOB(group of macroblocks) may contain errors is the macroblocks also.
- When error in GOB is detected, it *skips the remaining macroblocks* in the affected GOB and finds resynchronization marker.
- **Masking of error** – error concealment. Techniques used to avoid error propagation to other regions of the frame:
- *Error tracking*
- *Independent segment decoding*
- *Reference picture selection*
- 

6. What is MPEG coding in video compression? Differentiate between MPEG 1, MPEG 2 and MPEG 4.

- ***MPEG*** – Motion Pictures Expert Group – set of 3 standards formed by ISO – related to recording/transmission of integrated audio and video streams.
- 1. **MPEG-1**- ISO recommendation 11172.
  - *SIF – source intermediate digitization format.*
  - *Resolution:* ***352x288 pixels****.*
  - *For storing VHS-quality audio & video on CD-ROM at* ***bit rates =1.5Mbps.*** *Higher bit rates are also available.*
- 2. **MPEG-2** – ISO recommendation 13818
  - *For recording & transmission of studio quality audio*
  - *Video- The standard has 4 levels of video resolution*
  - ***LOW*** *– VHS quality video – CD quality audio –* ***bit rate =4Mbps***
- **MAIN** – ***4:2:0*** digitization format**–** ***720x576*** resolution.
  - *studio quality digital video with multiple CD quality audio channels.*
  - ***Bit rate = 15-20Mbps***
- **High 1440**- ***4:2:0*** digitization format–***1440x1152*** resolution.
  - *used for HDTV.*
  - ***Bit rate = 60Mbps****.*
- **High** – ***4:2:0*** digitization format - ***1920x1152*** resolution.
  - *used for wide screen HDTV.*
  - ***Bit rate = 80Mbps***
  - *3. MPEG-4 – similar to H.263. Very* ***low bit rate = 4.8-64kbps***
- USES SIF –source intermediate format
- Support two type spatial resolutions for two types of video source
  - NTSC
  - PAL
- Frame type: I,P,B-frame.
- Based on the H.261, there are two main differences:
  - ***Temporal*** *– time stamp inserted within the frame for decoder to synchronize more quickly in the event of macroblock corruption.*
  - ***No. of macroblocks*** *between two-time stamps****=22***
  - *B-frame increases time interval between I and P*
  -



- Support four levels and five profiles
- Four levels- *low, main, high1440, high*
- Simple, main, spatial resolution, quantization accuracy and high
- **MP@ML-** main profile at main level
- Used For digital television broadcasting
  - Resolution -***720x480 pixels*** at ***30Hz*** or ***720x576 pixels*** at ***25Hz***
  - Bit rate from ***4Mbps – 15Mbps***
  - Use interlaced scanning - frame divided into 2 fields
  - *Field mode*
  - *Frame mode*
- ***Interlaced scanning*** **-** each frame has 2 fields, alternate lines in each field.

- ***2 modes*** **-** field mode and frame mode depending on the video motion.
- For *larger movement* field mode is used for better compression- *live event*.
- For *smaller motion* frame mode is used- *studio based prog*.
- Motion estimation for encoding of macroblocks in ***P & B*** frames- 3 different modes are: ***field, frame & mixed***.
- ***Field mode*** – ***mv*** is computed using search window around the corresponding macroblock in preceding ***I or P fields.***
- ***B-frames*** - immediate succeeding ***P or I field***.
- ***Frame mode*** – ***mv*** in odd or even is encoded relative to that in preceding /succeeding odd /even field.
- ***Mixed mode - mv*** for both field & frame modes are computed and mean value selected.
- **3 standards with HDTV:**
  - **ATV –** *North America*.
  - **DIGITAL BROADCAST (DVB) –** *Europe, 4/3 aspect ratio.*
  - **Multiple sub-Nyquist sampling encoding (MUSE) -** Japan, Asia.

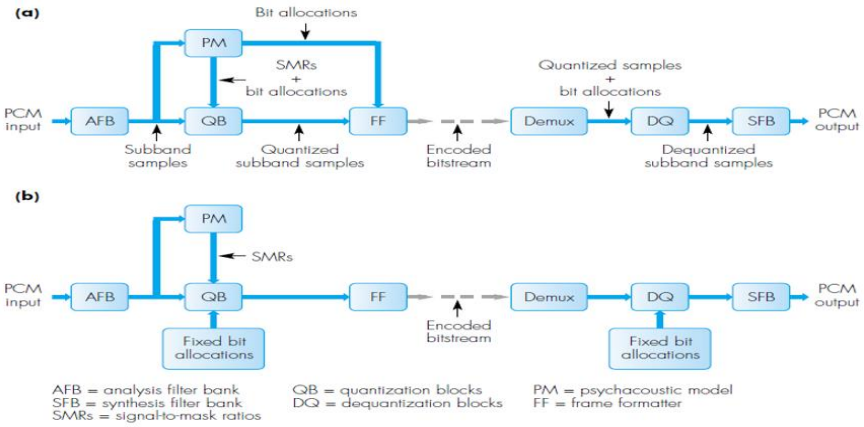
ITU-R HDTV specification - *Defines bitstream transmission over network, TV studio & International exchange prog. 16/9 aspect ratio, main profile at high level.*

HDTV(Grand Alliance) - *TV Manufactures defined ATV std.*

- ***Manipulate elements of picture*** **-** Alternate to H.263, supports low bitrate also.
- **Scene composition -** Content-based functionalities - each scene is defined in the form of background and one or more for ground objects -***Audio-visual object(AVOs)***
  - **Each AVO -** is defined as audio & video objects
  - **Object descriptor** - each audio & video objects origin description for manipulation.
  - **Binary format for scenes -** language used for modifying.
  - **Scene descriptor -** contains composition of a scene.
  - **Video object planes(VOPs) -** each video frame is segmented into no. of VOPs.

7. Differentiate between MPEG audio coder and DOLBY audio coder in detail with suitable diagrams.

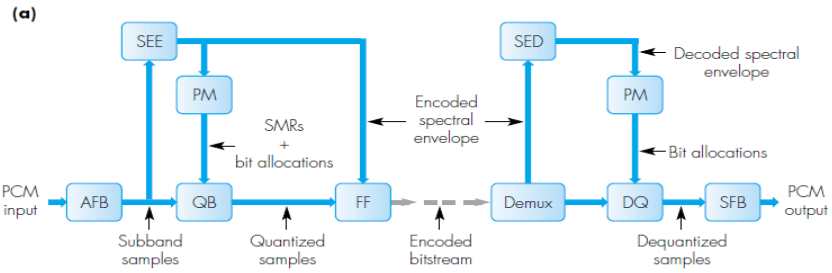
- **MPEG V.S Dolby AC-1**
  - **MPEG:**
    - *Advantage:* psychoacoustic model is required only in the encoder.
    - *Disadvantage:* a significant portion of each encoded frame contains bit allocation information - uses forward adaptive bit allocation mode and leads to inefficient use of available bitrate
  - **Dolby AC-1:**
    - Use a fixed bit allocation strategy for each sub-band which is then used by both the encoder and decoder.



- Dolby AC-2 standard - utilized in many applications including the compression

associated with the audio of many PC sound cards.

- Hybrid approach - used in the Dolby AC-3 standard used in MPEG audio standards including the audio associated with advanced television(ATV).



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