Kafrelsheikh University - Faculty of Engineering			
Course	**Communication systems**	**Date**	**3/6/2018**
Time	**3 Hours**	**Mark**	**85**
Students	**3^{rd} year Electronics and Electrical Communications**		

This exam measures ILOs no.: a.3, a.4, a. 8. a.18, a.25, b.2, b.7, b.15, c.1, c.14.

<u>Answer all the following questions:</u>

1- Drive the relationship between the viewing distance and the distance between adjacent pixels.
(10 Marks)
ANS:-

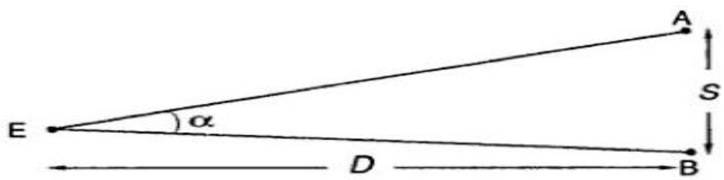


Fig. 1.2 *Angle Subtended at the Eye*

Let the two closely spaced but distinct objects be A and B.

Angle α subtended by A and B at the eye E = S/D radians

As 1 radian $= \frac{180}{\pi}$ degrees, angle $\alpha = \frac{180}{\pi} \times \frac{S}{D}$ degrees

For clear resolution, this angle should be = 1/60 degree.

$$\text{Hence} \quad \frac{180 \times S}{\pi \times D} = \frac{1}{60} \quad \text{or} \quad \frac{S}{D} = \frac{\pi}{180 \times 60} \qquad (1.1)$$

2- Calculate the viewing distance for a minimum distance of 0.073 inchs between two adjacent pixels, and then calculate the resolution required to achieve this viewing distance for a 50-inch screen TV. (Assume an aspect ratio of 4:3). (10 Marks)

ANS: $D = \frac{(180\times 60\ S)}{3.14} = 251\ inches$

For 50-inch tv screen we have Width=40 and height = 30.

The resolution required = $40\text{x}30/0.073^2$=226000.

3- Explain in detail the monochrome TV transmitter and receiver. (10 Marks)

Monochrome TV Transmitter

The basic elements of a monochrome TV transmitter are shown in Fig. 1.11. Description of their functions follows.

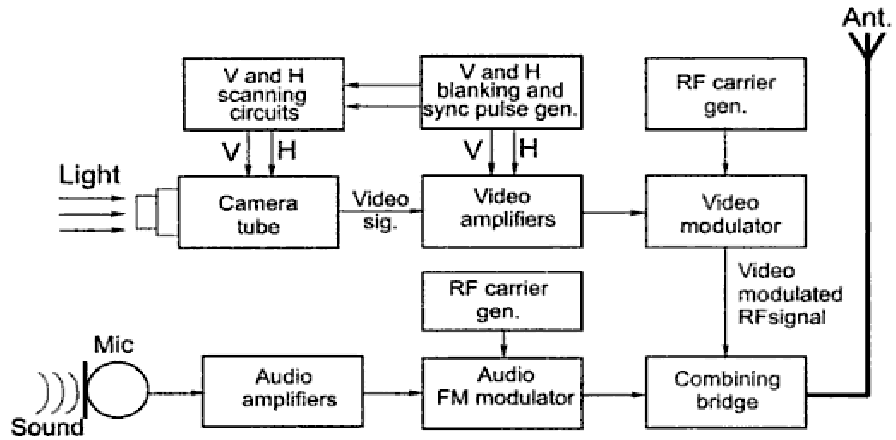


Fig. 1.11 *Basic Elements of a Monochrome TV Transmitting System*

Camera tube Converts intensity of light from a scene into electrical variations, called video signal by using a photosensitive target plate.

Scanning and sync circuits Electrical current is extracted from the photosensitive target of the camera tube with the help of a scanning beam which is produced by sawtooth currents through horizontal and vertical deflection coils.

Blanking and sync pulse generators The start of sawtooth or sweep current signal is triggered by pulses called sync pulses. The retrace is blanked by blanking pulses and these pulses are periodic and appear for the specified time by using a monostable multivibrator.

Video amplifiers Video signal along with blanking and sync pulses, called composite video signal is amplified by using wideband RC coupled amplifier circuits.

RF carrier generator and video modulator A radio frequency carrier of the channel frequency for video is generated and is amplitude modulated by the video signal. Modulation is of vestigial sideband (VSB) type AM to save the bandwidth. In this type of modulation, one sideband is vestiged to a small fraction of the whole band. In TV, lower sideband is vestiged and upper sideband and carrier are sent in full.

Microphone It converts sound pressure variations into electrical variations, called audio signals.

Audio amplifiers These amplify the weak audio signals.

RF carrier generator and audio modulator A radio frequency carrier is generated and is frequency modulated by the audio signals at low level. The radio frequency is then multiplied and amplified to the full level for transmission.

Combining bridge Video modulated and audio modulated signals pass through a combining bridge (also called diplexer circuit) to go to a common transmitting antenna. The bridge prevents audio modulated signals from going to video sections and vice versa to avoid overloading.

Transmitting antenna Video modulated signal and audio modulated signal are fed to the common transmitting antenna which radiates out the modulated signals into space in the form of electromagnetic waves. The antenna is omnidirectional in the horizontal plane.

Monochrome TV Receiver

Basic elements of a monochrome TV receiver are shown in Fig. 1.12.

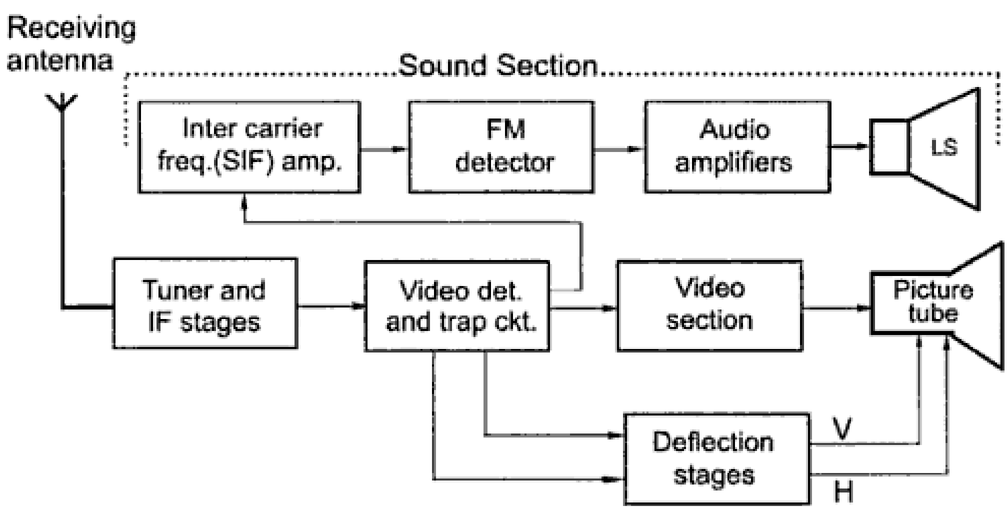


Fig. 1.12 *Basic Elements of a TV Receiver*

Receiving antenna The receiving antenna picks up signal from the electromagnetic waves travelling through space and the desired channel is selected by a tuned circuit.

Tuner and IF stages The receiver is a superhetrodyne receiver to achieve high selectivity and high gain. The signal is amplified by a radio frequency amplifier and then is mixed nonlinearly with the oscillations of higher frequency but of fixed amplitude, generated by a local oscillator. The output of the mixer consists of several intermodulation products (due to non-linear mixing), one of which is a signal having a frequency equal to the difference of frequencies of the two signals. The difference frequency is called *intermediate frequency*, which is selected and amplified.

Video detector and trap circuit The amplified IF goes to the video detector which recovers video signal from the modulated wave and feeds it to the video amplifier for amplification through the trap circuit, which prevents video signal from entering into audio channel.

Video amplifiers These are wideband RC coupled amplifiers. The amplified video signal goes to the picture tube.

Picture tube The video signal varies the strength of the electron beam. This beam strikes the phosphor dots on the fluorescent screen which glow, the intensity of glow being proportional to the intensity of the video signal.

Deflection stages The phosphor dots glow in quick succession from left to right and top to bottom with the help of scanning currents in the deflection coils. The synchronising pulses, recovered by the detector, trigger the scanning circuits which produce deflection currents, duly synchronised with the scanning currents used in the transmitter. The deflection currents go to the deflection coils to deflect the electron beam horizontally and vertically on the fluorescent screen to reproduce the picture.

Sound section It consists of sound IF (SIF) amplifier, FM detector, audio amplifier and loudspeaker. Difference of frequency between frequency modulated IF and video carrier IF is called *intercarrier frequency*, or second IF or sound IF. It is received from the video detector and passes to the SIF amplifier through trap circuit, which prevents SIF signal from going into video amplifier. FM detector detects the audio signal which is then amplified by audio amplifiers. The amplified signal goes to the loudspeaker which converts it into sound.

Thus the original sound is reproduced.

4- Channel 1 of a two-channel PAM system handles 0-10 kHz signals; the second channel handles 0-12 kHz signal. The two channels are sampled at equal intervals of time using very narrow pulses at the lowest frequency that is theoretically adequate. The sampled signals are time-multiplexed and passed through a low- pass filter before transmission. At the receiver that pulses in each of the two channels are passed through appropriate holding circuits (i.e. sample- and-hold) and low-pass filters. (15 Marks)

a- What is the minimum clock frequency of the PAM system? (5 Marks)

b- What is the minimum cutoff frequency of the LPF before transmission that will preserve the amplitude information on the output pulses? (5 Marks)

c- If the signal in channel 1 is sin 5000πt and in channel 2 is sin 10000πt. Sketch these signals; sketch the waves at the input to the first LPF, at the filter output, at the output of the sample-and-hold circuit and output of the LPF in channel 2. (5 Marks)

ANS: Solution

For channel 1: f_{m1}=10 kHz

Nyquist sampling rate $f_{N1}= 2f_{m1}= 20$ kHz

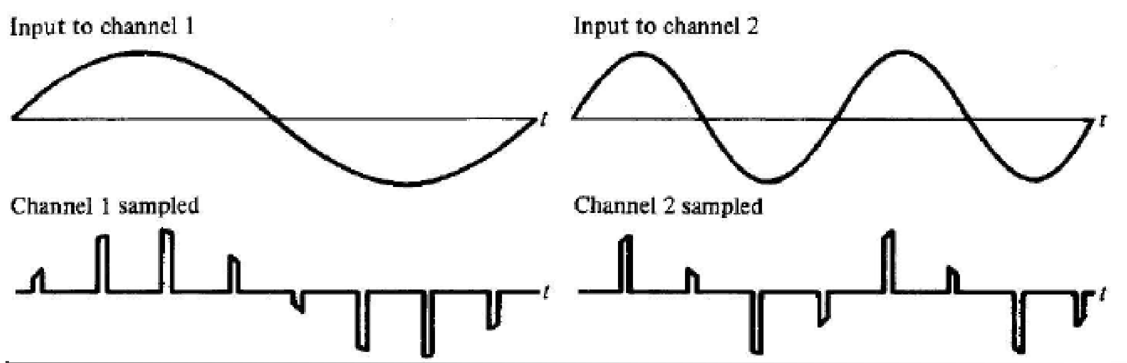
For channel 2: f_{m2}=12 kHz

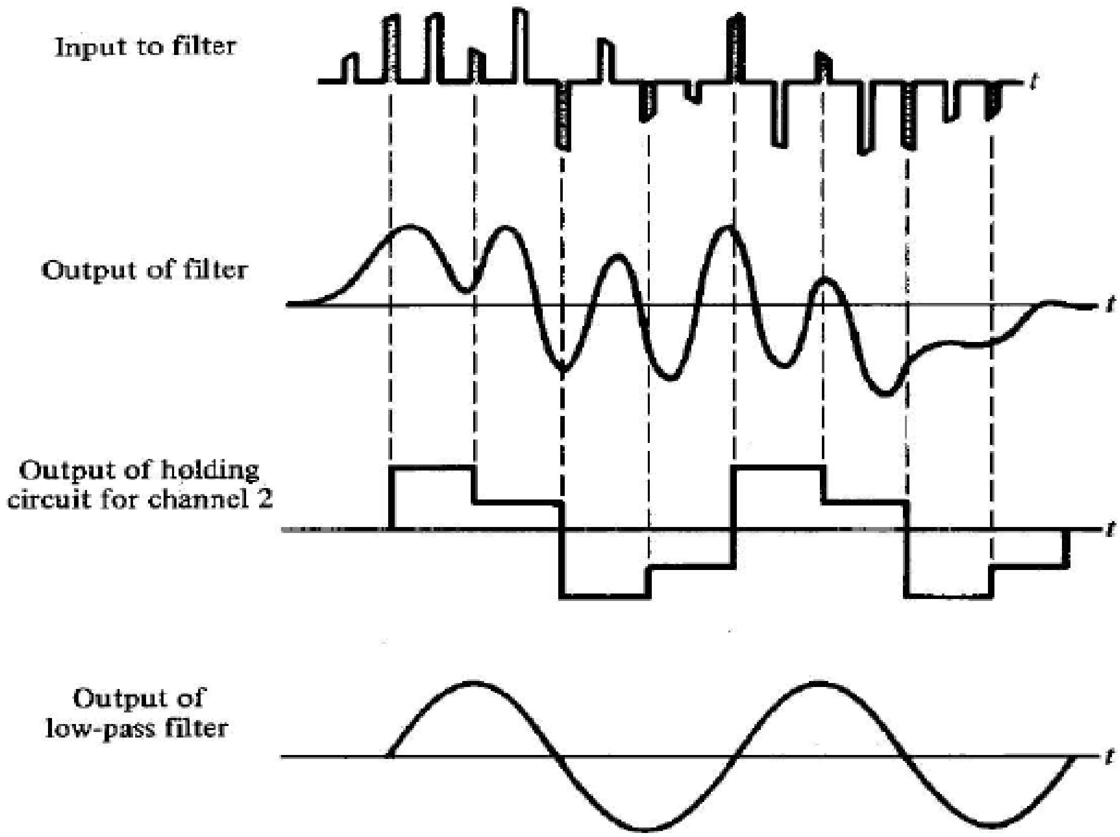
Nyquist sampling rate $f_{N2} = 2f_{m2} = 24kHz > 20$ kHz

Sampling rate f_s of the PAM system = 24 kHz for no aliasing

a) The minimum clock frequency f_x of the PAM system:
   No. of input PAM signals n= 2
   $T_x = T_s / n$
   $1/f_x = 1 / nf_s$ $f_x = nf_s = 2*24 = 48$ kHz

b) The minimum cutoff frequency of LPF:
   minimum cutoff frequency = B.W. of filter = f_{max}
   $B_x = f_x / 2 = 1/2T_x = 1/T_s = f_s = 24$ kHz
   Or
   f_{max} in multiplexed signal = $f_x / 2 = 24$ kHz
c) Channel 1: $f_1(t) = \sin 5000\pi t$
   Channel 2: $f_2(t) = \sin 10000\,\pi t$.
   $f_1 = 2500$ Hz < $f_2 = 5000$ Hz





5- **Explain in detail the Model of generic system of interactive services. (10 Marks)**

- **ANS:- The model shown in Figure 1.6 illustrates a generic system of interactive services. The downlink establishes communication between the network and the viewers; it can happen through broadcasting (open and available to all users/subscribers) or be individualized. The user is enabled to interact with the network by means of an uplink or return channel, expressing preferences and opinions. The return channel may be implemented by any communication technology which establishes a connection from the users to the networks.**

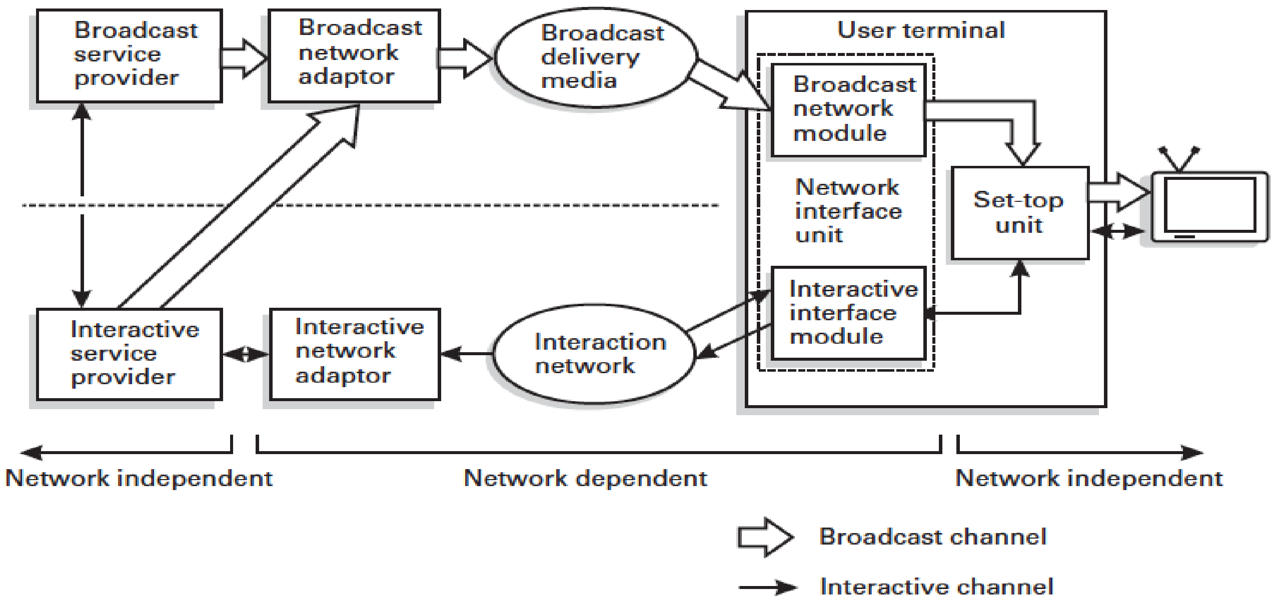


Figure 1.6 Model for a generic system of interactive services (Reimers, 2005b)

- **Figure 1.6 shows that two channels are established between the users and the supplier of the service. The network adapter provides the connectivity between the supplier of the service and the network, while the interface unit connects the network to the user. The supplier of the broadcast service distributes the MPEG-2 stream of transport from the unidirectional broadcast channel to the user's set-top box. The supplier of the interactive service offers an interactive channel for the bidirectional communication which is divided between the interaction route in the direct direction, to the downstream, and the return route in the reverse direction, to the upstream.**
- **In order to offer high-speed services to the user, the supplier of the interactive service can choose the broadcasting link to fit the data on the MPEG-2 transport stream. In that case, the broadcasting channel may contain data application or communication control, in such a way as to connect to the distribution network. The user may, for example, use a cable modem instead of a set-top box. A bidirectional control application and a communication channel will also be required from the different services suppliers, with the intention of obtaining synchronization.**

6- Explain in detail the adaptive PCM. (10 Marks)

ANS:- In the APCM quantizer, shown in Figure 2.11, the quantization step varies with time to accompany the signal amplitude variations. This adaptation is based on the previous signal samples. The objective of the adaptive technique is the reduction of the dynamic range of the signal, in order to obtain a reduction of the final transmission rate.

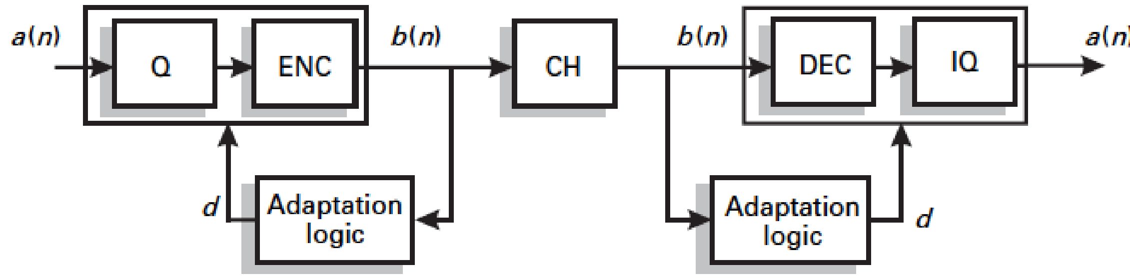


Figure 2.11 Flowchart of an APCM

7- **What are the differences between the Entropy encoding and source encoding? Give 3 examples for each coding system. (10 Marks)**

- **ANS:- Entropy encoding just manipulates bit streams without regard to what the bits mean. It is a general lossless, fully reversible technique, applicable to all data. We will illustrate it by means of three examples.**
- **Our first example of entropy encoding is run-length encoding. In many kinds of data, strings of repeated symbols (bits, numbers, etc.) are common. These can be replaced by a special marker not otherwise allowed in the data, followed by the symbol comprising the run, followed by how many times it occurred. If the special marker itself occurs in the data, it is duplicated (as in character stuffing).**
- **For example, consider the following string of decimal digits:**
- **315000000000000845871111111111116354674000000000000000000000065**
  If we now introduce A as the marker and use two-digit numbers for the repetition count, we can encode the above digit string as
- **315A01284587A11316354674A02265**
  Here run-length encoding has cut the data string in half.
- **Our second example of entropy encoding is statistical encoding. By this we mean using a short code to represent common symbols and long ones to represent infrequent ones. Morse code uses this principle, with E being . and Q being --.- and so on.**
- **Our third example of entropy encoding is CLUT (Color Look Up Table) encoding. Consider an image using RGB encoding with 3 bytes/pixel. In theory the image might contain as many as 2^{24} different color values. In practice, it will normally contain many fewer values, especially if the image is a cartoon or computer-generated drawing, rather than a photograph. Suppose that Only 256 color values are actually used. A factor of almost three compression can be achieved by building a 768-byte table listing the RGB values of the 256 colors actually used, and then representing each pixel by the index of its RGB value in the table.**
- **Now we come to source encoding, which takes advantage of properties of the data to produce more (usually lossy) compression. Here, too, we will illustrate the idea with three examples. Our first example is differential encoding, in which a sequence of values (e.g., audio samples) are encoded by representing each one as the difference from the previous value. Differential pulse code modulation is an example of this**

technique. It is lossy because the signal might jump so much between two consecutive values that the difference does not fit in the field provided for expressing differences, so at least one incorrect value will be recorded and some information lost.

- **Differential encoding is a kind of source encoding because it takes advantage of the property that large jumps between consecutive data points are unlikely. Not all sequences of numbers have this property. An example lacking this property is a computer-generated list of random telephone numbers to be used by telemarketers for bothering people during dinner. The difference between consecutive telephone numbers in the list will take as many bits to represent as the numbers themselves.**
- **Our second example of source encoding consists of transformations. By transforming signals from one domain to another, compression may become much easier. Consider, for example, the Fourier transformation in which a function of time is represented as a list of amplitudes. Given the exact values of all the amplitudes, the original function can be reconstructed perfectly. However, given only the values of the first, say, eight amplitudes rounded off to two decimal places, it may still be possible to reconstruct the signal so well that the listener cannot tell that some information has been lost. The gain is that transmitting eight amplitudes requires many fewer bits than transmitting the sampled waveform.**
- **Our third example of source encoding is vector quantization which is also directly applicable to image data. Here, the image is divided up into fixed-size rectangles. In addition to the image itself, we also need a table of rectangles of the same size as the image rectangles (possibly constructed from the image). This table is called the code book. Each rectangle is transmitted by looking it up in the code book and just sending the index instead of the rectangle. If the code book is created dynamically (i.e., per image), it must be transmitted, too. Clearly, if a small number of rectangles dominate the image, large savings in bandwidth are possible here.**

8- Explain in detail the MPEG-4 video encoding process. (10 Marks)

9- ANS:- Blocks of 8~8 or 16-16 pixels for the prediction and compensation of movement.

-¼ pixel precision. This feature implies a negligible increase in computational overhead. A more precise movement description implies a smaller amount of prediction error generating a video of better quality.

-Global Motion Compensation (GMC). This compensates for the global movement of an object using a small number of parameters. The GMC technique is based on the codification of the trajectory of the camera moves and encoding of the textures. Figure 3.4 illustrates this feature.

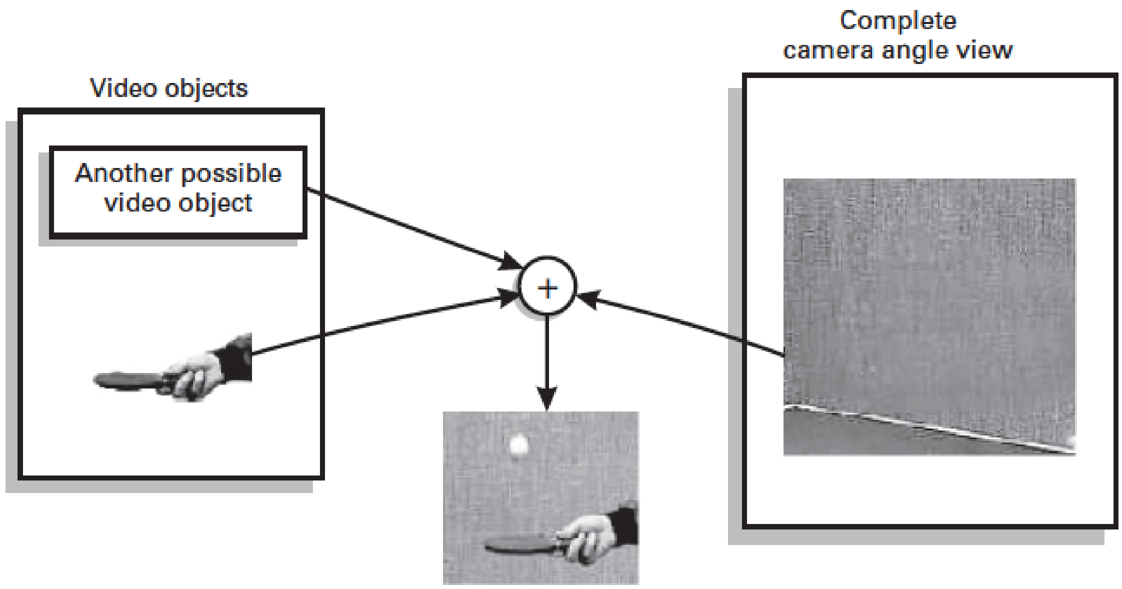


Figure 3.4 MPEG-4 encoding process using GMC

Good Luck and Best Wishes ***Dr. Ibrahim Elashry***