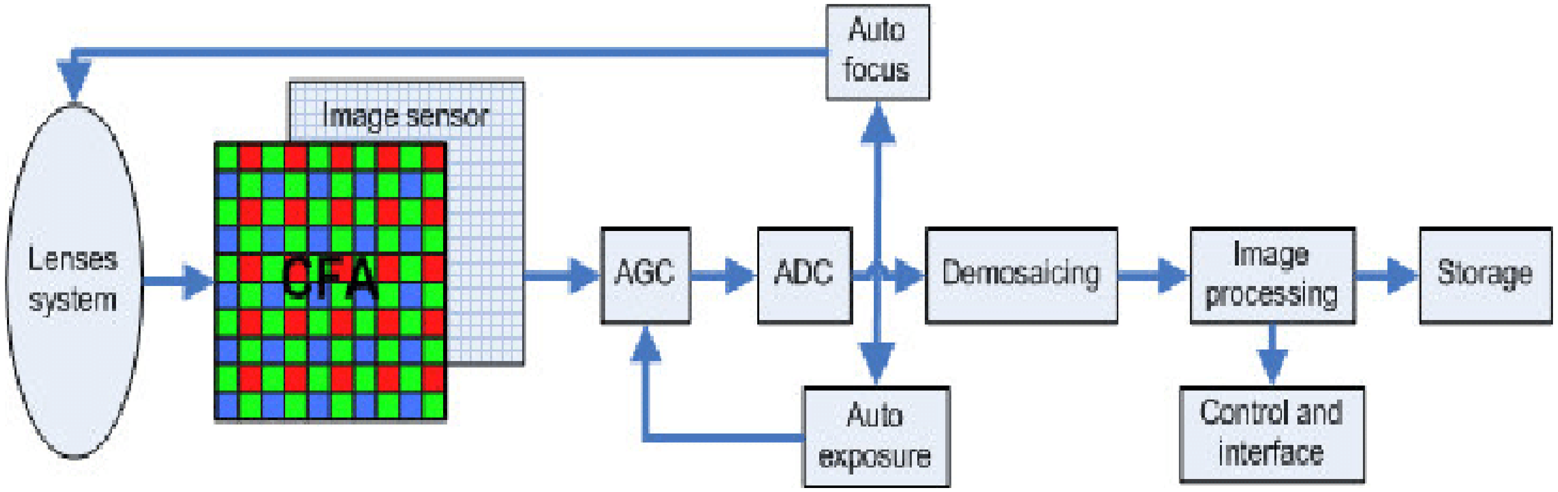
Unit 6

ADVANCED A/V EQUIPMENTS AND MEDIA FORMATS

Digital Camera and Camcorder



General Block diagram of Digital Camera

Explanation of Block Diagram

- Lens System – An Optical lens is used because the entire photo-site is not light sensitive. Only the photodiode is. As the light-sensitive area is so small in comparison to the size of the photo-site, the overall sensitivity of the chip is reduced. So manufacturers use micro-lenses to direct photons that would normally hit non-sensitive areas and otherwise go undetected, to the photodiode.
- Color Filter Array (CFA) – It is a mosaic of tiny color filters placed over the pixel sensors of an image sensor to capture color information.
- Image Sensor (CCD, CMOS) - The Image sensor receives incident light (photons) that is focused through a lens or other optics. Depending on whether the sensor is CCD or CMOS, it will transfer information to the next stage as either a voltage or a digital signal.
- Automatic Gain Control (AGC) – It is a camera function that increases the average gain when the image is too dark. AGC is basically a from of amplification where the camera will automatically boost the image received so that objects can be seen more clearly.

Explanation of Block Diagram

- ADC - ADC stands for Analog to Digital Converter and refers to the digital camera's ability to capture an image and convert it into a digital file.
- Auto focus - Autofocus (AF) is the feature of a camera that tries to ensure that your chosen subject is sharp within the photo. Sensors detect how far away the subject is from the camera, and this information is relayed to the lens, which then uses an electronic motor to adjusts the focal distance of the lens.
- Auto Exposure - Automatic exposure is an automated digital camera system that sets the aperture and shutter speed, based on the external lighting conditions for the photo. The camera measures the light in the frame and then automatically locks the camera's settings to ensure proper exposure.
- Demosaicing - A demosaicing algorithm is a digital image process which calculates the “missing” color values for each pixel in a raw capture and it is used to reconstruct a full color image from the incomplete color samples output from an image sensor overlaid with a color filter array (CFA). It is also known as CFA interpolation or color reconstruction.

- Image processing – It is a built-in image processing systems for applications such as computer vision, and multimedia and surveillance. It is use to reduce the noise in the image and do certain enhancements and adjustments .
- Control and interface –

  I. Mode dial: On most cameras this is a round dial on top of the camera. This is the shooting mode dial. By rotating it we can choose different modes like capture, record, panorama, etc.

  II. Shutter button: we press this button to prefocus the camera and take a picture.

  III. ISO Settings : We use this feature to change the ISO setting of the camera. The ISO determines how sensitive the sensor is to light. You use higher ISO settings to take pictures in low light conditions.

  IV. Aperture setting: The aperture determines how much light enters the camera. When you choose Aperture Priority as the shooting mode, you use a dial to change the aperture, and the camera automatically selects the shutter speed to properly expose the image.

V. Exposure compensation setting: This is used to increase or decrease the exposure. You increase or decrease the exposure when the camera gets it wrong.

VI. Histogram display: This option displays a graph that shows you the distribution of pixels from the lightest parts of the image to the darkest parts of the image. If you notice a spike on the right side of the histogram, your image is overexposed. If you see a spike on the left side of the histogram, part of the shadows are pure black, and no details are visible.

VII. Shutter speed setting: The shutter speed setting comes into play when you shoot in Shutter Priority mode. After choosing Shutter Priority for the shooting mode, you use a dial to change the shutter speed, and the camera automatically selects the correct f/stop to properly expose the image.

VIII. White balance: If the camera gets confused due to multiple light sources, the whites have a color cast to them and may have a green, orange, or blue tint. You can rectify this problem by choosing a preset white balance or by manually setting the white balance.

IX. Flash control: If your camera has a built-in flash unit, you push this button to pop the flash unit up and enable it. You can use flash to light the scene or add additional light known as fill flash.

X. LCD panel: This panel shows you all the current settings. When you change a setting such as the shutter speed or ISO setting, the panel updates to show you the new settings. If your camera doesn't have an LCD panel, these settings are visible in most camera viewfinders.

- Storage - Digital cameras use a number of storage systems. These are like reusable, digital film, and they use a card reader to transfer the data to a computer. Many involve fixed or removable flash memory. Digital camera manufacturers often develop their own proprietary flash memory devices, including Smart Media cards, CompactFlash cards and Memory Sticks. Some other removable storage devices include:

  I. Floppy disks

  II. Hard disks, or micro drives

  III. Writeable CDs and DVDs

Operation of Digital Camera

- **A digital camera** takes light and focuses it via micro-lens onto a sensor made out of silicon. It is made up of a grid (or an *array)* of tiny photosensors or photo-sites used to record the incoming pattern of light. Each photo-site is usually called a pixel, a contraction of "picture element".
- Each photo-site on a CCD or CMOS chip is composed of a light-sensitive area made of crystal silicon in a photodiode which absorbs photons and releases electrons through the photoelectric effect. The electrons are stored in a 'well' as an electrical charge that is accumulated over the length of the exposure. The charge that is generated is proportional to the number of photons that hit the sensor.
- This electric charge is then transferred and converted to an analog voltage that is amplified and then sent to an Analog to Digital Converter where it is digitized (turned into a number).

Video Editing Techniques - Linear and Non Linear

Linear Video Editing

- **Linear video editing** is a process of selecting, arranging and modifying images and sound in a pre-determined, ordered sequence – from start to finish. Linear editing is most commonly used when working with videotape. Unlike film, videotape cannot be physically cut into pieces to be spliced together to create a new order. Instead, the editor must dub or record each desired video clip onto a master tape.

  - Advantages –

I. It is simple and inexpensive. There are very few complications with formats, hardware conflicts, etc.

II. For some jobs linear editing is better. For example, if all you want to do is add two sections of video together, it is a lot quicker and easier to edit tape-to-tape than to capture and edit on a hard drive.

III. Learning linear editing skills increases your knowledge base and versatility. According to many professional editors, those who learn linear editing first tend to become better all-round editors.

- Disadvantages –

I. it is not possible to insert or delete scenes from the master tape without re-copying all the subsequent scenes. As each piece of video clip must be laid down in real time, you would not be able to go back to make a change without re-editing everything after the change.

II. Because of the overdubbing that has to take place if you want to replace a current clip with a new one, the two clips must be of the exact same length. If the new clip is too short, the tail end of the old clip will still appear on the master tape. If it's too long, then it'll roll into the next scene. The solution is to either make the new clip fit to the current one, or rebuild the project from the edit to the end, both of which is not very pleasant. Meanwhile, all that overdubbing also causes the image quality to degrade.

Non-Linear Video Editing

- **The nonlinear video editing** method is a way of random access editing, which means instant access to whatever clip you want, whenever you want it. So instead of going in a set order, you are able to work on any segment of the project at any time, in any order you want.
- In nonlinear video editing, the original source files are not lost or modified during editing. This is done through an edit decision list (EDL), which records the decisions of the editor and can also be interchanged with other editing tools.
- As such, many variations of the original source files can exit without needing to store many different copies, allowing for very flexible editing. It is also easy to change cuts and undo previous decisions simply by editing the EDL, without having to have the actual film data duplicated.
- Loss of video quality is also avoided due to not having to repeatedly re-encode the data when different effects are applied.

- Advantages –

I. It allows you access to any frame, scene, or even groups of scenes at any time. Also, as the original video footage is kept intact when editing, you are able to return to the original take whenever you like.

II. Nonlinear video editing systems offers the flexibility of editing. You can change your mind a hundred times over and changes can also be made a hundred times over without having to start all over again with each change.

III. It is also possible to edit both standard definition (SD) and high definition (HD) broadcast quality videos very quickly on normal PCs which do not have the power to do the full processing of the huge full quality high resolution data in real-time.

- Disadvantages –

I. The biggest downside to nonlinear video editing is the cost. While the dedicated hardware and software doesn’t cost much, the computers and hard drives do, from two to five times more than the gear. As such, the average price for a basic nonlinear video editing package can come in between $5,000 and $10,000. For stand-alone systems that approach broadcast quality, the amount you pay may be twice that.

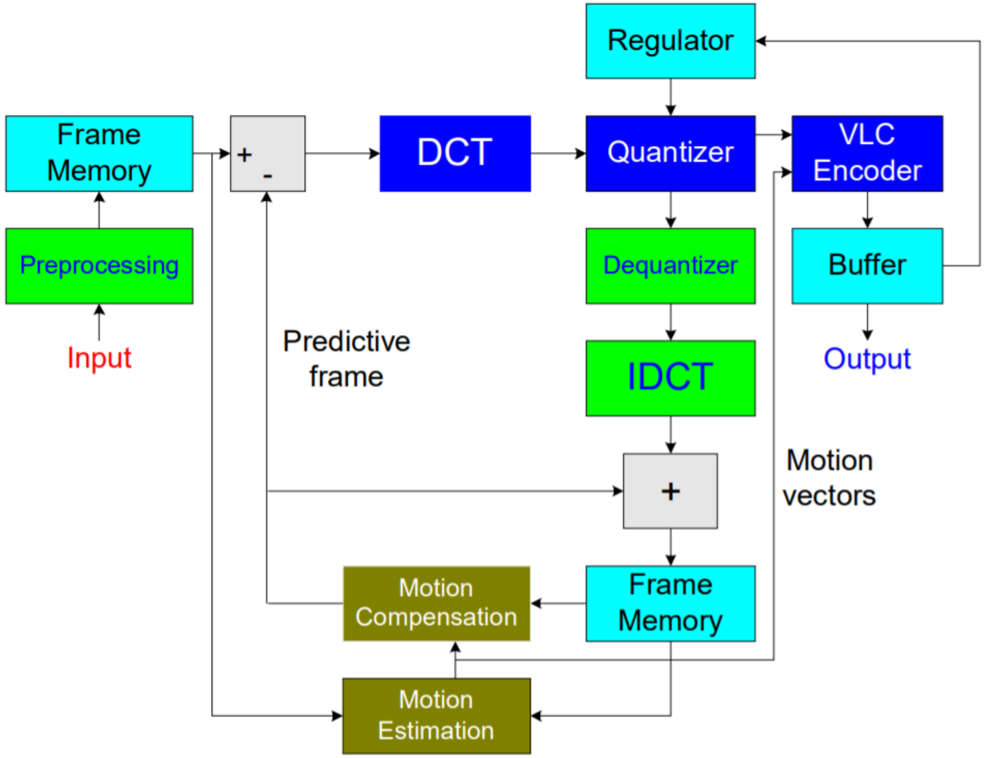


Fig. 2. Block diagram of MPEG encoder for video part.

Block Diagram of MPEG Encoder

Audio Formats

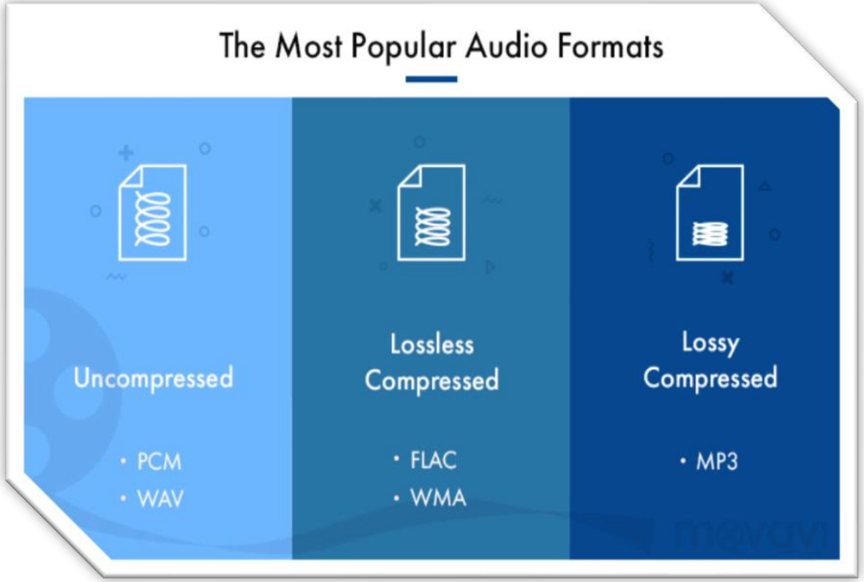
- What Is an Audio Format?
  - An audio format is a two-part structure that consists of a container and a codec. A container is a special file in which multimedia data, in this case audio, is compressed and stored. A codec is a piece of software used to perform the data compression and to decode the data during playback. In other words, a container is where the audio files are stored, while a codec is the tool that makes them smaller and easier to store. Together, they make up what is called the format.

- There are three major groups of audio file formats:

  I. Uncompressed audio is audio without any compression applied to it. This includes audio recorded in PCM or WAV form. Examples of lossless formats includes WMA Lossless or FLAC

  II. Lossless audio compression is where audio is compressed without losing any information or degrading the quality at all.

  III. Lossy audio compression attempts to apply to discard as much 'irrelevant' data as possible from the original audio, thereby producing a file much smaller than the original that sounds almost identical. This results in a much smaller file size then lossless or uncompressed audio. Lossy audio formats include AC3, DTS, AAC, MPEG-1/2/3, Vorbis, and Real Audio.



WAV – Waveform Audio File

- A waveform audio file, also known as a wave file, or simply WAV after its extension, is a common type of sound file. Microsoft and IBM introduced the format in 1991 for use in the Microsoft Windows 3.1 operating system (OS).
- The format uses containers to store audio data, track numbers, sample rate, and bit rate. This file format is based on the Resource Interchange File Format (RIFF), which stores audio files in indexed “chunks” and “sub-chunks.”
- The WAV file had two very big things going for it when introduced. First, it could digitize sounds 100% faithful to the original source because it is a lossless format. “Lossless” means that the file format does not compromise audio quality even when it holds compressed data. Second, the format is very easy to edit and manipulate with software.
- The most common WAV audio format is uncompressed audio in the linear pulse-code modulation (LPCM) format. LPCM is also the standard audio coding format for audio CDs, which store two-channel LPCM audio sampled at 44,100 Hz with 16 bits per sample. Since LPCM is uncompressed and retains all of the samples of an audio track, professional users or audio experts may use the WAV format with LPCM audio for maximum audio quality.
- As it is a lossless format, its file occupies large space. One four-minute song could easily consume over 35 megabytes (MB) of space when saved as a WAV. The WAV format is limited to files that are less than 4 GB, Although this is equivalent to about 6.8 hours of CD-quality audio (44.1 kHz, 16-bit stereo), it is sometimes necessary to exceed this limit, especially when greater sampling rates, bit resolutions or channel count are required. The W64 format was therefore created.

MP3 – MPEG-1 Audio Layer III

- MPEG-1 Audio Layer-3 format offers a very high rate of compression for audio files (about a 12:1 ratio) while preserving the original level of sound quality to the ear. Because of its high quality at small size, mp3 has exploded in popularity, and many sites offer mp3 files for download (most are offering these files in violation of copyright).
- Digital audio is typically created by taking 44,100, 16-bit samples per second (Hz) of the analog audio signal, this means hat one second of CD-quality sound requires 1.4 million bits (about 176K bytes) of data. Using a knowledge of how people actually perceive sound, the developers of MP3 devised a compression algorithm that reduces data about sound that most listeners cannot perceive.
- MP3 is currently the most powerful algorithm in a series of audio encoding standards developed under the sponsorship of the Motion Picture Experts Group (MPEG) and formalized by the International Organization for Standardization (ISO).
- Bitrate is the product of the sample rate and number of bits per sample used to encode the music. CD audio is 44100 samples per second. The number of bits per sample also depends on the number of audio channels. CD is stereo and 16 bits per channel. So, multiplying 44100 by 32 gives 1411200—the bitrate of uncompressed CD digital audio.
- MP3 was designed to encode this 1411 kbit/s data at 320 kbit/s or less.
- When using MPEG-2 instead of MPEG-1, MP3 supports only lower sampling rates (16000, 22050 or 24000 samples per second) and offers choices of bitrate as low as 8 kbit/s but no higher than 160 kbit/s. By lowering the sampling rate, MPEG-2 layer III removes all frequencies above half the new sampling rate that may have been present in the source audio.

WMA – Windows Media Audio