BASICS OF AUDIO & VIDEO

Module 3

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Types of magnetic recording

1. Analog recording
2. Digital recording
3. Magneto_optical recording
4. Domain propagation memory

1.Analog recording

- Works on concept of remnant magnetization.

Remanence or **remanent magnetization** or residual**magnetism** is the **magnetization** left behind in a ferromagnetic material (such as iron) after an external magnetic field is removed. It is also the measure of that **magnetization.**

- The magnitude of applied field influences remnant magnetization of gvn material .
- Tape in magnetic material form_demagnetized in its blank form
- During recording_tape runs at const speed.
- Tape magnetized by writing head with current_propotional to signal it receives
- When magnetization distribution is read out by read_write head_original sound reproduced.
- Audio video recording by analog…now turned out digital recording

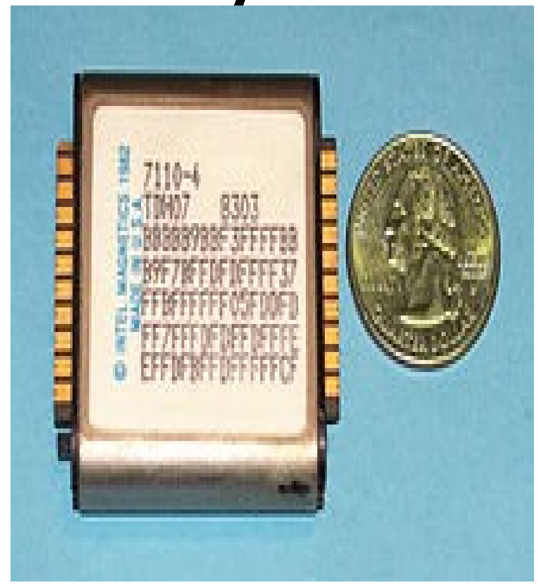
2.Digital recording

- Unlike analog_magnetization distribution created
- Works with 2 stable magnetized states.
- +Ms and –Ms digital recording done with magnetic tapes

3.Magneto-optical recording

- Optical technology used.media read or written optically
- During write _magnetic medium, is heated by laser beam
- Magnetization is switched by creating s small magnetic field
- Reading process_using magneto-optical Kerr effect._no poularity

4.Domain propagation memory



- Reffered as bubble memory
- intel 7110 magnetic-bubble memory module
- **Bubble memory** is a type of non-volatile computer memory that uses a thin film of a magnetic material to hold small magnetized areas, known as *bubbles* or *domains*, each storing one bit of data.

- The basic idea is to control domain wall motion in a magnetic medium that is free of microstructure.
- Bubble refers to a stable cylindrical domain.
- Data is then recorded by the presence/absence of a bubble domain.
- Domain propagation memory has high insensitivity to shock and vibration, so its application is usually in space and aeronautics.

Technical specification

1.Access method

- Magnetic storage media classified as either sequential access memory(start from beginning of tape,one by one) or random access memory(index maintained at beginning_no need to travel all tracks)_fastest
- The access time can be defined as the average time needed to gain access to stored records.
- In the case of magnetic wire, the read/write head only covers a very small part of the recording surface at any given time. Accessing different parts of the wire involves winding the wire forward or backward until the point of interest is found
- Discs and tapes have many parallel tracks across the width of the media and the read/write heads take time to switch between tracks and to scan within tracks. Different spots on the storage media take different amounts of time to access. For a hard disk this time is typically less than 10 ms, but tapes might take as much as 100 s.

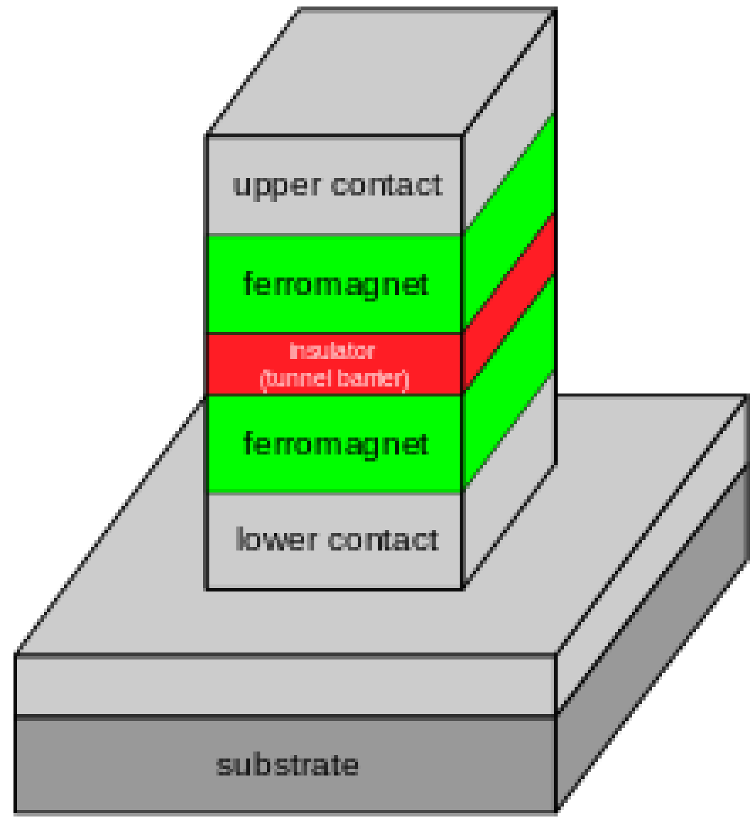
Technical specification
2.Current usage

- As of 2011, common uses of magnetic storage media are for computer data mass storage on hard disks and the recording of analog audio and video works on analog tape.
- Since much of audio and video production is moving to digital systems
- Digital tape and tape libraries are popular for the high capacity data storage of archives and backups.
- Floppy disks with older computer systems and software.
- Magnetic storage used in bank cheques (MICR) and credit/debit cards (mag stripes).

Future

- A new type of magnetic storage, called magnetoresistive random-access memory or MRAM, is being produced that stores data in magnetic bits based on the tunnel magnetoresistance (TMR) effect.
- **Magnetoresistive random-access memory (MRAM)** is a non-volatile random-access memory technology available today that began its development in mid-1980s.[1
- **Tunnel magnetoresistance** (TMR) is a magnetoresistive effect that occurs in a **magnetic tunnel junction** (MTJ), which is a component consisting of two ferromagnets separated by a thin insulator.

Magnetic tunnel junction



- Its advantage is non-volatility, low power usage, and good shock robustness.
- Developments is progressing on thermal-assisted switching (TAS) and spin-transfer torque (STT).
- MRAM offers a smaller magnitude of storage density and capacity orders than hard disk drive(HDD)
- Use in media players

DC and AC bias (Tape bias)

- **Tape bias** is the term for two techniques, **AC bias** and DC bias, that improve the fidelity of analogue tape recorders.
- DC bias is the addition of a direct current to the audio signal that is being recorded.
- AC bias is the addition of an inaudible high-frequency signal (generally from 40 to 150 kHz) to the audio signal. Most contemporary tape recorders use AC bias.
- When recording, magnetic tape has a nonlinear response as determined by its coercivity. Without bias, this response results in poor performance especially at low signal levels.
- Bias increases the signal quality of most audio recordings significantly by pushing the signal into more linear zones of the tape's magnetic transfer function.

DC bias

- The earliest magnetic recording systems simply applied the input signal w/o modifications to a recording head, resulting in recordings with poor low-frequency response and high distortion.
- With the addition of a suitable direct current to the signal, a DC bias, was found to reduce distortion by operating the tape substantially within its linear-response region.
- The principal disadvantage of DC bias was that it left the tape with a net magnetization, which generated significant noise on replay because of the grain of the tape particles.
- Some early DC-bias systems used a permanent magnet that was placed near the record head. It had to be swung out of the way for replay. DC bias was replaced by AC bias but was later re-adopted by some very low-cost cassette recorders.

AC bias

- The reduction in distortion and noise provided by AC bias was rediscovered in 1940 by Walter Weber
- during the Second World War uk company had produced a steel wire recorder equippedwith AC bias.
- in 1950sproduced magnetic tape,based tape recorderusing German wartime technology.

How AC BIAS works

- During recording process, tape passes the the tape head.
- As it goes past the edge at the end of gap, the oscillating magnetic field comes in to play.
- due to the applied AC bias is rapidly reduced to the average magnetic field of the much slower-changing audio signal, and the tape particles are therefore left in this magnetic condition.
- A **tape head** is a type of transducer used in tape recorders to convert electrical signals to magnetic fluctuations and vice versa.
- The non-linearity of the magnetic particles in the tape coating is overcome by having the AC bias field greater by at least an order of magnitude , which saturates these particles in both magnetic directions while they pass the gap in the recording head.

tape head

- The AC bias level is quite critical and, after being adjusted for a particular tape formulation with a specific recording machine, is usually left unchanged..
- The mechanism is similar to the demagnetizing signal which is used to erase the tape except that the desired audio signal is retained on the tape during the recording process.
- The large AC bias acts as a demagnetizing signal which decays exponentially as the tape moves beyond the head, while the audio signal is the residual field that remains imprinted on the magnetic medium

Application(tape bias)

- system gives the minimal distortion (which is the highest bias). There is also a level at which the high-frequency response is at maximum (lowest bias).
- cassette recorders are always set up for minimal distortion.
- in particular compact-cassette recorders have the bias set at a compromise level (usually a little higher) to give good frequency response and acceptably low distortion.

- Inventions
- Bang & Olufsen invented and patented the so-called Dolby HX PRO (Headroom eXtension) principle for combining bias control with the Dolby system for better high-frequency response in cassette recorders.
- Tandberg invented the cross-field recording system for tape recorders where a separate head with the bias was used. Adding bias to the recorded signal in the one head had tended to limit the system's high-frequency response, due to interaction between bias and signal.
- The cross-field system produced less interference from the bias signal. This allowed extended high-frequency performance compared to mixing the two signals in the recording head, but mechanical tolerances for cross-field are tight. The system required frequent readjustment and was largely abandoned.
- Japanese manufacturer Akai, however, persisted with cross-field bias and successfully marketed portable and mains-operated machines featuring the cross-field system.

- Different amplitudes of bias field are optimal for different types of tape, so most recorders offer a bias setting switch on the control panel, or, in the case of the compact audio cassette, may switch automatically according to cutouts on the cassette shell.
- Ferric-based tapes require the lowest bias field,
- chrome-based tapes (including the pseudo-chromes) requiring a higher level
- metal-particle tapes requires even more.
- Metal-evaporated tape accepts the highest level of bias, but it is mostly used for digital recording (which does not use bias, as the non-linearity is not a major problem).
- The same is valid for a combination cassette tape, the FeCr variant, on which a thicker ferric layer was covered by a thinner chrome layer. The idea behind this was that at lower frequencies and higher head currents the ferric layer would be more deeply magnetized, while at higher frequencies only the top Cr layer was active. In practice, this didn't work well, and some claimed[who?] that this thin chrome layer was quickly polished off in heavy use

Frequency Response

- **Frequency response** is the quantitative measure of the output spectrum of a system or device in response to a stimulus,
- It is used to characterize the dynamics of the system.
- It is a measure of magnitude and phase of the output as a function of frequency, in comparison to the input.
- In simplest terms, if a sine wave is injected into a system at a given frequency, a linear system will respond at that same frequency with a certain magnitude and a certain phase angle relative to the input. Also for a linear system, doubling the amplitude of the input will double the amplitude of the output. In addition, if the system is time-invariant (so LTI), then the frequency response also will not vary with time.

Two applications of frequency response analysis are related but have different objectives.

- Obj 1)For an audio system, the objective may be to reproduce the input signal with no distortion.

That would require a uniform (flat) magnitude of response up to the bandwidth limitation of the system, with the signal delayed by precisely the same amount of time at all frequencies. That amount of time could be seconds, or weeks or months in the case of recorded media.

- Obj 2)In contrast, for a feedback apparatus used to control a dynamic system, the objective is to give the closed-loop system improved response as compared to the uncompensated system.
- The feedback generally needs to respond to system dynamics within a very small number of cycles of oscillation (usually less than one full cycle), and with a definite phase angle relative to the commanded control input.
- For feedback of sufficient amplification, getting the phase angle wrong can lead to instability for an open-loop stable system, or failure to stabilize a system that is open-loop unstable.

Estimation

- Estimating the frequency response for a physical system generally involves exciting the system with an input signal, measuring both input and output time histories, and comparing the two through a process such as the Fast Fourier Transform (FFT).
- One thing to keep in mind for the analysis is that the frequency content of the input signal must cover the frequency range of interest because the results will not be valid for the portion of the frequency range not covered.
- The frequency response of a system can be measured by applying a *test signal*, for example:
- applying an impulse to the system and measuring its response (see impulse response)
- sweeping a constant-amplitude pure tone through the bandwidth of interest and measuring the output level and phase shift relative to the input
- calculating the impulse response of an input signal and output signal of the system by deconvolution ,after applying a signal with a wide frequnecy spectrum.

- The frequency response is characterized by the *magnitude* of the system's response, typically measured in decibels (dB) or as a decimal, and the *phase*, measured in radians or degrees, versus frequency in radians/sec or Hertz (Hz).

Nonlinear frequency response[edit

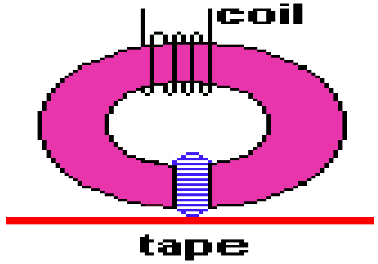
- If the system under investigation is nonlinear then applying purely linear frequency domain analysis will not reveal all the nonlinear characteristics.
- To overcome these limitations, generalized frequency response functions and nonlinear output frequency response functions have been defined that allow the user to analyze complex nonlinear dynamic effects.[7]
- The nonlinear frequency response methods reveal complex resonance, inter modulation, and energy transfer effects that cannot be seen using a purely linear analysis and are becoming increasingly important in a nonlinear world.

Applications

- in electronics this stimulus would be an input signal.[8] In the audible range it is usually referred to in connection with electronic amplifiers, microphones and loudspeakers.).
- Frequency response requirements differ depending on the application.[9] In high fidelity audio, an amplifier requires a frequency response of at least 20–20,000 Hz, with a tolerance as tight as ±0.1 dB in the mid-range frequencies around 1000 Hz, however, in telephony, a frequency response of 400–4,000 Hz, with a tolerance of ±1 dB is sufficient for intelligibility of speech.[9]
- Frequency response curves are often used to indicate the accuracy of electronic components or systems.[8] .
- Once a frequency response has been measured (e.g., as an impulse response), provided the system is linear and time-invariant, its characteristic can be approximated with arbitrary accuracy by a digital filter. Similarly, if a system is demonstrated to have a poor frequency response, a digital or analog filter can be applied to the signals prior to their reproduction to compensate for these deficiencies.

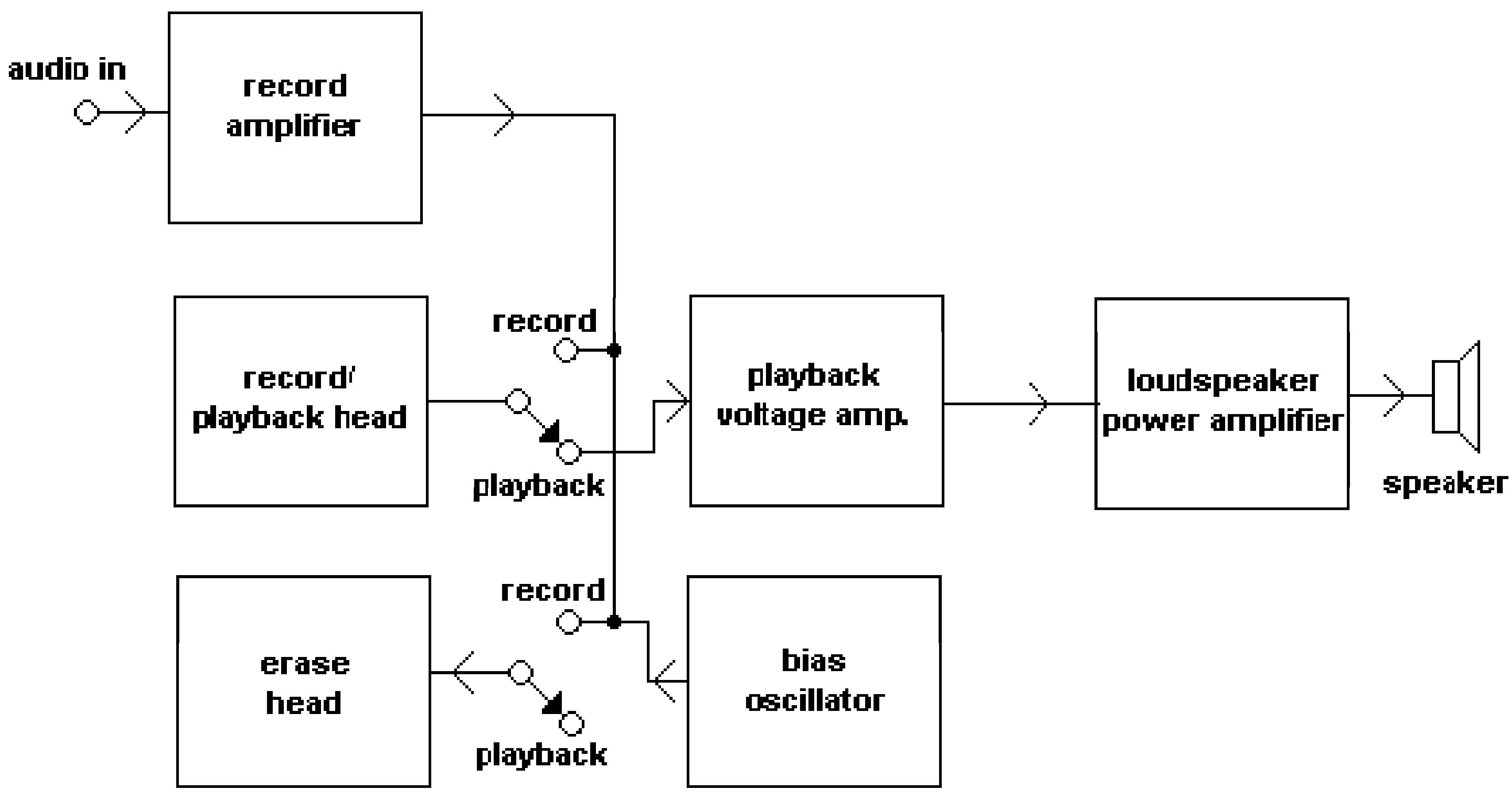
- The form of a frequency response curve is very important for anti-jamming protection of radars, communications and other systems.

Block Diagrams
Tape Recorder Tutorial



- The tape head consists of a ring of soft magnetic material, called the core, with a small gap in it.
  A coil is wound around the core. The tape travels over the gap in the core.
- During recording, an audio signal causes current to flow through the coil producing a magnetic field in the gap, as shown by the blue lines of force in the diagram.
- As the audio signal varies in amplitude and frequency so does the magnetic field.
- The tape consists of a plastic film coated with a material that is magnetised by the field as it passes over the gap.
- As the magnetic field varies in strength so does the magnetism stored on the tape.
- During playback the tape passes over the same head. (it is called the record/playback head).0
- This time the magnetism stored on the tape induces a voltage in the head coil.
- This voltage is amplified and used to drive a loudspeaker.

Amplifier



audio in
record amplifier
record/ playback head
record
playback
playback voltage amp.
loudspeaker power amplifier
speaker
erase head
record
playback
bias oscillator

3.2.6 concept of multi_track sequencing

- Music sequencer: any device that is used to record ,edit ,play back music . a tape recorder plays audio, a sequencer plays MIDI data. A sequence is a multi-track, linear, time-ordered recording of MIDI musical data, which a sequencer can play at various speeds, rewind, shuttle to particular points, record into, or copy to a file for storage..
- It should be able to handle information in different format like cv/gate, MIDI , open sound control(OSC),DWA(digital audio workstations ,plug ins for audio and automation data
- DAW are software programs specifically created for the purpose of recording , editing and producing audio files.
- DAW interface is similar to that of multi-track tape recorders.
- The layout of all DAWs follow a standard with transport controls like play ,rewind ,record etc.. track controls and a mixer and have a waveform display.
- Single –track DAWs will display one sound , whether mono or stereo ,at a time.

- DAWs with multi-track functionality have the capability to handle simultaneous procedures on more than one track.
- DAWs will have the controls for each track that help to adjust the overall volume and stetero balance of the sound for each track individually.
- A DAW can process the sound on a track by bringing some other software or software plugins
- DAWs have practically replaced the traditional tape-based studio setup becausetjey have acquired the capability of these setups.
- One drawback of DAW is poor interoperability with other DAWs.
- DAWs provoide an option to ‘undo’ a previous operation,which is not available in analod DAWs

- Some form of automation done with the help of envelopes would be a common DAWs feature.
- Curve –based interactive graphs or procedural line segment-based graphs are called envelopes.

3.3. Digital recording

- In **digital recording**, audio signals picked up by a microphone or other transducer or video signals picked up by a camera or similar device are converted into a stream of discrete numbers, representing the changes over time in air pressure for audio, and chroma and luminancevalues for video, then recorded to a storage device.
- To play back a digital sound recording, the numbers are retrieved and converted back into their original analog waveforms so that they can be heard through a loudspeaker.
- To play back a digital video recording, the numbers are retrieved and converted back into their original analog waveforms so that they can be viewed on a video monitor, television or other display.

Recording Process

- The analog signal is transmitted from the input device to an analog-to-digital converter (ADC).
- The ADC converts this signal by repeatedly measuring the momentary level of the analog (audio) wave and then assigning a binary number with a given quantity of bits (word length) to each measuring point.
- The frequency at which the ADC measures the level of the analog wave is called the sample rate or sampling rate.
- A digital audio sample with a given word length represents the audio level at one moment.
- The longer the word length the more precise the representation of the original audio wave level.
- The higher the sampling rate the higher the upper audio frequency of the digitized audio signal.
- The ADC outputs a sequence of digital audio samples that make up a continuous stream of 0s and 1s.
- These binary numbers are stored on recording media such as a hard drive, optical drive or in solid state memory.

Playback

- The sequence of numbers is transmitted from storage into a digital-to-analog converter (DAC), which converts the numbers back to an analog signal by sticking together the level information stored in each digital sample, thus rebuilding the original analog wave form.
- This signal is amplified and transmitted to the loudspeakers or video screen.

Recording of bits

- Even after getting the signal converted to bits, it is still difficult to record; the hardest part is finding a scheme that can record the bits fast enough to keep up with the signal. For example, to record two channels of audio at 44.1 kHz sample rate with a 16 bit word size, the recording software has to handle 1,411,200 bits per second.
- **Techniques to record to commercial media**
- For digital cassettes, the read/write head moves as well as the tape in order to maintain a high enough speed to keep the bits at a manageable size.
- For optical disc recording technologies such as CDs or DVDs, a laser is used to burn microscopic holes into the dye layer of the medium. A weaker laser is used to read these signals. This works because the metallic substrate of the disc is reflective, and the unburned dye prevents reflection while the holes in the dye permit it, allowing digital data to be represented.

Concerns with digital audio recording

1.Word size

- The number of bits used to represent a sampled audio wave (the *word size*) directly affects the resulting noise in a recording after intentionally added dither, or the distortion of an undithered signal[44].
- The number of possible voltage levels at the output is simply the number of levels that may be represented by the largest possible digital number. There are no "in between" values allowed. I
- if there are more bits in each sample the waveform is more accurately traced, because each additional bit doubles the number of possible values. The distortion is roughly the percentage that the least significant bit represents out of the average value. Distortion (as a percentage) in digital systems increases as signal levels decrease, which is the opposite of the behavior of analog systems.[45]

2.Sample rate

- The sample rate is just as important a consideration as the word size. If the sample rate is too low, the sampled signal cannot be reconstructed to the original sound signal.
- To overcome aliasing, the sound signal (or other signal) must be sampled at a rate at least twice that of the highest frequency component in the signal. This is known as the Nyquist-Shannon sampling theorem.

3.Error rectification

- One of the advantages of digital recording over analog recording is its resistance to errors.

3.3.1 Basics of digital coding

- Audio Codec

An **audio codec** is a codec (a device or computer program capable of encoding or decoding a digital data stream) that encodes or decodes audio

- In software, an audio codec is a computer program implementing an algorithm that compresses and decompresses digital audio data according to a given audio file or streaming media audio coding format.
- The objective of the algorithm is to represent the high-fidelity audio signal with minimum number of bits while retaining quality. This can effectively reduce the storage space and the bandwidth required for transmission of the stored audio file. Most software codecs are implemented as libraries which interface to one or more multimedia players.

- In hardware, audio codec refers to a single device that encodes analog audio as digital signals and decodes digital back into analog.
- In other words, it contains both an analog-to-digital converter (ADC) and digital-to-analog converter (DAC) running off the same clock signal. This is used in sound cards that support both audio in and out, for instance. Hardware audio codecs send and receive digital data using buses .

Audio Coding Format

- An **audio coding format**[1] (or sometimes **audio compression format**) is a content representation format for storage or transmission of digital audio (such as in digital television, digital radio and in audio and video files).
- Examples of audio coding formats include MP3, AAC, Vorbis, FLAC, and Opus.
- A specific software or hardware implementation capable of audio compression and decompression to/from a specific audio coding format is called an audio codec;
- an example of an audio codec is LAME, which is one of several different codecs which implements encoding and decoding audio in the MP3 audio coding format in software.

- Some audio coding formats are documented by a detailed technical specification document known as an **audio coding specification**.
- Some such specifications are written and approved by standardization organizations as technical standards, and are thus known as an **audio coding standard**.
- The term "standard" is also sometimes used for *de facto* standards as well as formal standards.

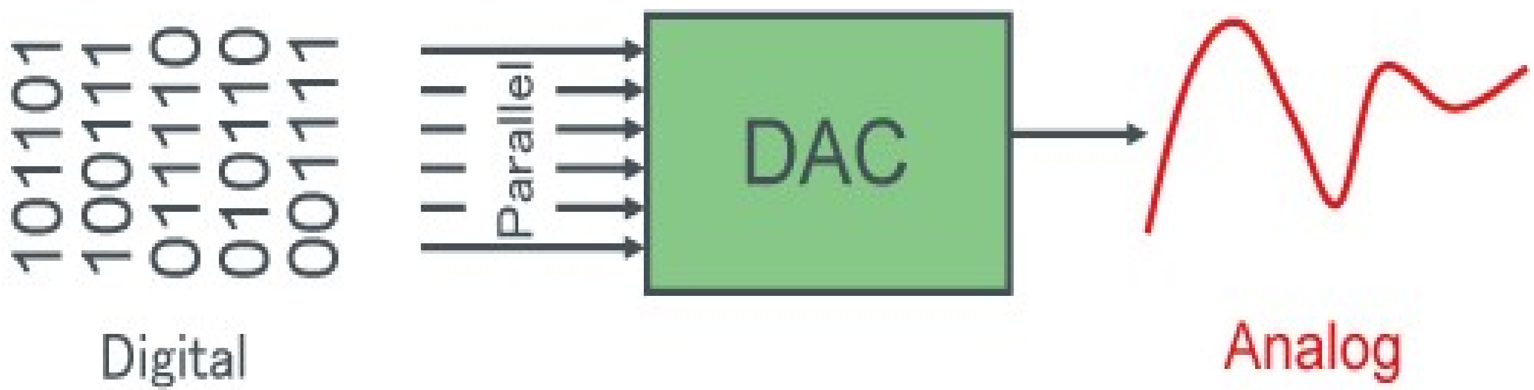
- Audio content encoded in a particular audio coding format is normally encapsulated within a container format.
- . The container also contains metadata such as title and other tags, and perhaps an index for fast seeking.[2
-] A notable exception is MP3 files, which are raw audio coding without a container format.
- De facto standards for adding metadata tags such as title and artist to MP3s, such as ID3, are hacks which work by appending the tags to the MP3, and then relying on the MP3 player to recognize the chunk as malformed audio coding and therefore skip it.
- In video files with audio, the encoded audio content is bundled with video (in a video coding format) inside a multimedia container format.

Lossless, lossy, and uncompressed audio coding formats

- A lossless audio coding format reduces the total data needed to represent a sound but can be de-coded to its original, uncompressed form. A lossy audio coding format additionally reduces the bit resolution of the sound on top of compression, which results in far less data at the cost of irretrievably lost information.
- Consumer audio is most often compressed using lossy audio codecs as the smaller size is far more convenient for distribution. Lossless audio coding formats such as FLAC and Apple Lossless are sometimes available, though at the cost of larger files.
- Uncompressed audio formats, such as pulse-code modulation (.wav), are also sometimes used.

3.3.2. basics of DA conversion

- **D/A Converters**
- D/A converters convert digital signals into analog format.



- Digital Data:Evenly spaced discontinuous valuesTemporally discrete, quantitatively discrete
- Analog Data (Natural Phenomena):Continuous range of valuesTemporally continuous, quantitatively continuous
- **A/D Converters**
- An A/D converter is a device that converts analog signals (usually voltage) obtained from environmental (physical) phenomena into digital format
- Conversion involves a series of steps, including sampling, quantization, and coding.

A/D and D/A Requirements

- Electrically sophisticated and high-speed processing are performed digitally in CPUs and DSPs.
- Natural phenomena are converted to digital signals using an A/D converter for digital signal processing, then converted back to analog signals via a D/A converter.
- Advancements in Microfabrication Technology→Signal Processing Digitization
→A/D and D/A Converters Required

A/D Converter Applications

- Digital Audio:Digital audio workstations, sound recording, pulse-code modulation
- Digital signal processing:TV tuner cards, microcontrollers, digital storage oscilloscopesScientific instruments:
- Digital imaging systems, radar systems, temperature sensors

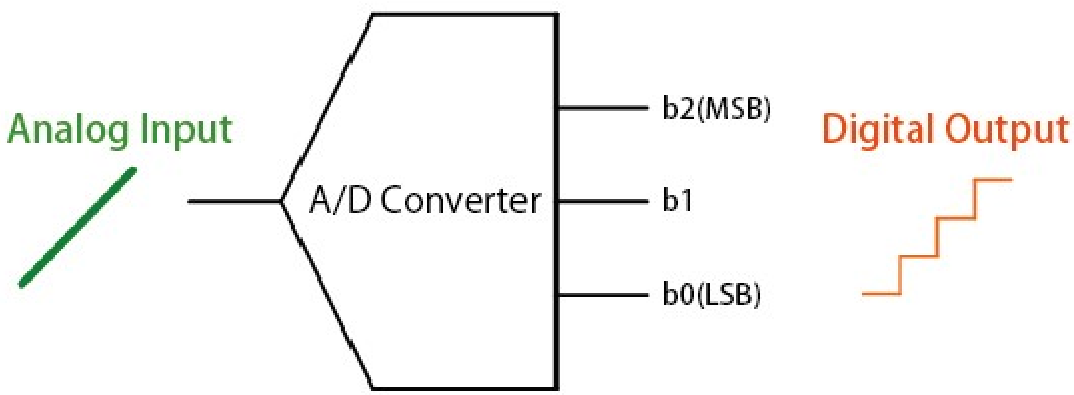
D/A Converter Applications

- Digital Audio ： CD, MD, 1-bit Audio
- Digital Video ： DVD, Digital Still Camera
- Communication Equipment ： Smartphones, FAX, ADSl equipment
- PCs ： Audio, video cardsMeasurement instruments ： Programmable power supplies, etc.

Basic Operation of a D/A Converter

- A D/A converter takes a precise number (most commonly a fixed-point binary number) and converts it into a physical quantity (example: voltage or pressure). D/A converters are often used to convert finite-precision time series data to a continually varying physical signal.
- An ideal D/A converter takes abstract numbers from a sequence of impulses that are then processed by using a form of interpolation to fill in data between impulses. A conventional D/A converter puts the numbers into a piecewise constant function made up of a sequence of rectangular functions that is modeled with the zero-order hold.
- A D/A converter reconstructs original signals so that its bandwidth meets certain requirements. With digital sampling comes quantization errors that create low-level noise which gets added to the reconstructed signal. The minimum analog signal amplitude that can bring about a change in the digital signal is called the Least Significant Bit (LSB), while the (rounding) error that occurs between the analog and digital signals is referred to as quantization error.

Basic Operation of an A/D Converter

- Now, let's take a look at the basic operation of an A/D converter.
- 

- he A/D converter breaks up (samples) the amplitude of the analog signal at discrete intervals, which are then converted into digital values. The resolution of an analog to digital converter (indicating the number of discrete values it can produce over a range of analog values) is typically expressed by the number of bits. In the above case of a 3bit A/D converter, the upper value (b2) is referred to as the Most Significant Bit (MSB) and the lowest value (b0) the Least Significant Bit (LSB).

Analog Signal to Digital Signal Conversion Methods

- Sampling: Sampling is the process of taking amplitude values of the continuous analog signal at discrete time intervals (sampling period Ts).
  [Sampling Period Ts = 1/Fs (Sampling Frequency)]
  Sampling is performed using a Sample and Hold (S&H) circuit
- .Quantization: Quantization involves assigning a numerical value to each sampled amplitude value from a range of possible values covering the entire amplitude range (based on the number of bits).[Quantization error: Sampled Value - Quantized Value]
- Coding: Once the amplitude values have been quantized they are encoded into binary using an Encoder.

DAC based on 6 parameters

- Physical size
- Power consumption
- Resolution
- Speed
- Accuracy
- cost

DAC Types

- The pulse-width modulator:Used for electric motor speed control and dimming LED lamps
- Oversampling DACs or interpolating DACs such as those employing delta-sigma modulation, use a pulse density conversion technique with oversampling
- The binary-weighted DAC :this type of converter is usually limited to 8-bit resolution or less.

- The R-2R ladder DAC which is a binary-weighted DAC
- The successive approximation or cyclic DAC, which successively constructs the output during each cycle
- The successive approximation or cyclic DAC, which successively constructs the output during each cycle
- Hybrid DACs, which use a combination of the above techniques in a single converter

3.3.3 Basics of audio compression techniques and standards

- The purpose of audio data compression is to reduce the transmission bandwidth and the storage requirements of audio data.
- These are the algorithms embedded in software in the form of audio codecs.
- Less audible or meaningful sounds are eliminated through psychoacoustics in these algorithms so that the space requirement for storage and transmission is reduced.
- Methods like coding a, pattern recognition and linear predication are used to reduce redundancy and the amount of information used to represent the uncompressed data.

Loseless audio compression

- **Lossless compression** is a class of data compression algorithms that allows the original data to be perfectly reconstructed from the compressed data. By contrast, lossy compression permits reconstruction only of an approximation of the original data, though usually with improved compression rates (and therefore reduced file sizes).
- Lossless data compression is used in many applications. For example, it is used in the ZIP file format and in the GNU tool gzip. It is also often used as a component within lossy data compression technologies
- Lossless compression is used in cases where it is important that the original and the decompressed data be identical, or where deviations from the original data would be unfavourable.
- Typical examples are executable programs, text documents, and source code. Some image file formats, like PNG or GIF, use only lossless compression, while others like TIFF and MNG may use either lossless or lossy methods.

Lossless compression techniques

- Most lossless compression programs do two things in sequence: the first step generates a *statistical model* for the input data, and the second step uses this model to map input data to bit sequences in such a way that "probable" (e.g. frequently encountered) data will produce shorter output than "improbable" data.
- The primary encoding algorithms used to produce bit sequences are Huffman coding (also used by DEFLATE) and arithmetic coding. Arithmetic coding achieves compression rates close to the best possible for a particular statistical model, which is given by the information entropy, whereas Huffman compression is simpler and faster but produces poor results for models that deal with symbol probabilities close to 1.

Lossy audio compression

- In information technology, **lossy compression** or **irreversible compression** is the class of data encoding methods that uses inexact approximations and partial data discarding to represent the content.
- These techniques are used to reduce data size for storing, handling, and transmitting content.
- The different versions of the photo of the cat to the right show how higher degrees of approximation create coarser images as more details are removed.
- This is opposed to lossless data compression (reversible data compression) which does not degrade the data.
- The amount of data reduction possible using lossy compression is much higher than through lossless techniques.

)High compression (low quality) JPEG
2)Low compression (high quality) JPEG

Coding methods

- MDCT(modified discrete cosine transform) are widely used by lossy compression algorithms to identify ,in an audio signal ,what information is not relevant.
- A transform domain is the outcome of the conversion of time domain sampled waveforms using these transforms.
- After the transformation is doen in to a frequency domain,bits can be allocated to component frequencies based on their audibility.

- <u>The audibility of spectral components is calculated on the following factors:</u>

1. <u>The absolute threshold of hearing</u>: max and min limits of human hearing
2. <u>The principles of simultaneous masking</u>: One signal masking another separated by frequency
3. <u>Temporal masking</u>: One signal is masked by another signal separated by time
4. <u>Equal-loudness contours</u> may also be used to weight the perceptual importance of components.

- Source based coders are another type of lossy audio compressor like the linear predictive coding(LPC) that is used with speech
- These coders whiten the audio signal before quantization using a model of the sound’s generator like the human vocal tract in the case of LPC.
- A basic perceptual coding technique like in the LPC, shapes the coder’s quantization noise in to the spectrum of the target signal ,by using a linear predictor and reconstructing the audio signal thereby mask it.

- Lossy formats have been found to be better for distributing streaming audio or interactive applications as in cell phone applications
- The distinctiveness of these applications is that decompression must happen as the data flows and not after the entire data stream is transmitted.
- Codecs specifically designed for streaming data effectively , will be chosen for such applications

- Methods that are used in the encoding and decoding of the data can result in latency.
- To optimize on efficiency ,longer segments of the data are analyzed by some codecs for coding
- In such cases it will require a larger segment of data at a time when decoding.
- Latency of the algorithm is a significant factor in the quality of output as too long delays can have a degrading effect.

- Latency is the number of samples that are to be analyzed after which an audio block is processed

Speech encoding

- The model used for perception of music is different those that are for perception of speech.
- Sounds of human voice fall in a narrower range of frequencies and are less complex, encoding speech with a low bit rate can provide high quality results.
- When analog data is to be compressed, quantization needs to be done when it is digitized into numbers say integers.(called A/D conversion)
- This provides some visibility of the compromise required between high resolution where there are more analog intervals and high compression where a small number of integers are generated.
- Many speech compression methods apply a compromise form of quantization

- A combination two approaches ……

Sounds made by a single human voice are encoded

Most of the data in the signal is thrown away ,avoiding coding of the full frequency rqange of human hearing.

Must be able to reconstruct an ‘ intelligible ‘ voice

3.3.4 Digital tape recording systems

Concept of MIDI

Introduction to analog and digital Mixers

- Digital vs Annalog
- Propagation Delay
- Ease of Use
- Sound Quality
- Remote Control

• University Questions

1. Explain the need for A/D and D/A conversion in Digital Tape recording?(2 m 2018)
2. With neat sketches ,explain the need for AC bias in audio recording?(5 m 2018)
3. Explain the working of an eight bit Digital to analog converter.(5 m 2018)
4. Explain about MP3(2 m 2017)
5. Explain two noise reduction techniques(2 m 2017)
6. Discuss digital tape recording systems(15 m 2017)
7. Discuss magnetic recording on a tape and explain recorded wavelength , gap width and tape speed(15 m 2017)
8. What is DA conversion(2 m 2016)?
9. Distinguish analog and digital mixers.(5 m 2016)
10. Discuss AC and DC biasing of magnetic recording in tapes(5m 2016)
11. Discuss magnetic recording on tape and explain recorded wavelength ,gap width and tape speed(15 m 2016)