Notes

3

FUNDAMENTALS OF SOUND TECHNOLOGY

3.1 INTRODUCTION

In this lesson, you will learn about different technologies i.e. analog and digital used for processing of sound (audio) signals, and differentiate between them. You will learn about how the respective technology is used, to capture, process and reproduce the signals, i.e. about the analog to digital conversion and digital to analog conversion.

We will also discuss why, today, digital technology is more widely used than analog technology.

3.2 OBJECTIVES

After reading this lesson you will able to

- explain the structure of analog and digital signal;
- explain the process of capturing sound both in analog mode and digital mode;
- explain why digital technology is used more widely than analog technology;
- recognize the process of analog to digital conversion and vice versa.

3.3 ANALOG SIGNAL

In order to record, reproduce, or transmit sound, it first needs to be converted into an electrical signal. The beginning of this process requires a microphone. A microphone has a thin diaphragm that is suspended in or attached (depending on the type of microphone) to a magnetic field. The diaphragm moves back and forth in reaction to the sound waves that pass through it, and that movement within the magnetic field creates a small electrical signal, which is an electrical representation of the compressions and rarefactions of the sound wave. The signal is transmitted from the microphone along its cable to be amplified. Microphones generate only a small amount of signal (measured in volts), which is further amplified by using an amplifier (which you will learn in further chapters)

3.4 DIGITAL SIGNAL

A digital signal refers to an electrical signal that is converted into a pattern of bits. Unlike an analog signal, which is a continuous signal that contains time-varying quantities, a digital signal has a discrete value at each sampling point. The precision of the signal is determined by how many samples are recorded per unit of time. For example, the figure 3.1 below shows an analog pattern (represented as the curve) alongside a digital pattern (represented as the discrete lines).

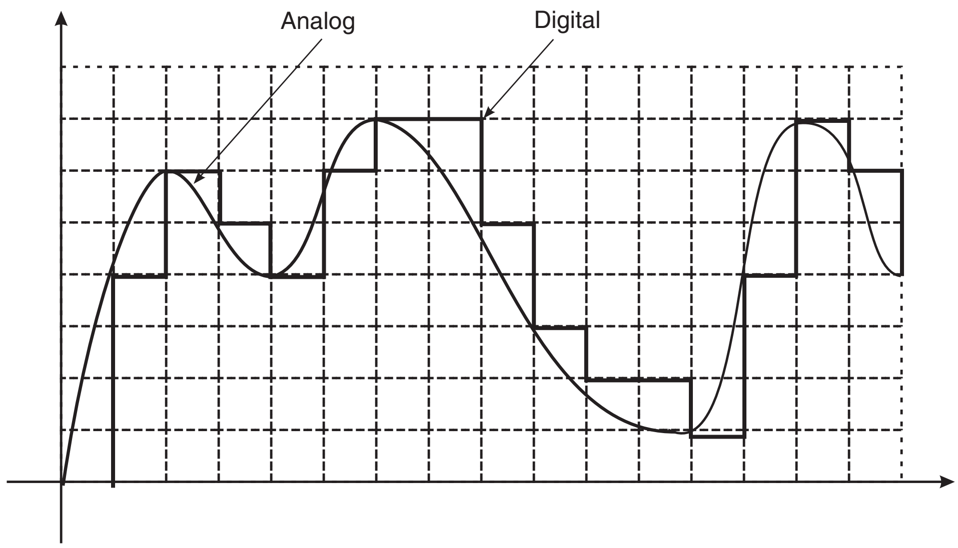


Fig. 3.1: Analog signal and coresponding digital level at different points

A digital signal is easily represented by a computer because each sample can be defined with a series of bits that are either in the state 1 (on) or 0 (off).

Digital signals can be compressed and can include additional information for error correction.

3.5 ANALOG V/S DIGITAL SIGNAL

Sound is recorded by converting continuous variations in sound pressure into corresponding variations in electrical voltage using microphone, this varying voltage is then converted into varying pattern of magnetization (by recording head) on tape or alternatively into a pattern of light and dark areas on an optical soundtrack on film.

In case of digital recording it converts the electrical wave form from a microphone into a series of binary numbers, each of which represents the amplitude of the signal at a sampling time.

Digital audio has advantages of benefitting from the developments in the computer industry, and is particularly beneficial because the size of that industry results in scope for mass production (and therefore cost savings). Now-a-days it is common for sound to be recorded, processed and edited on relatively low cost desktop equipment.

Any analog signal like sound electrical signal temperature, pressure etc., has infinite values between two limits.

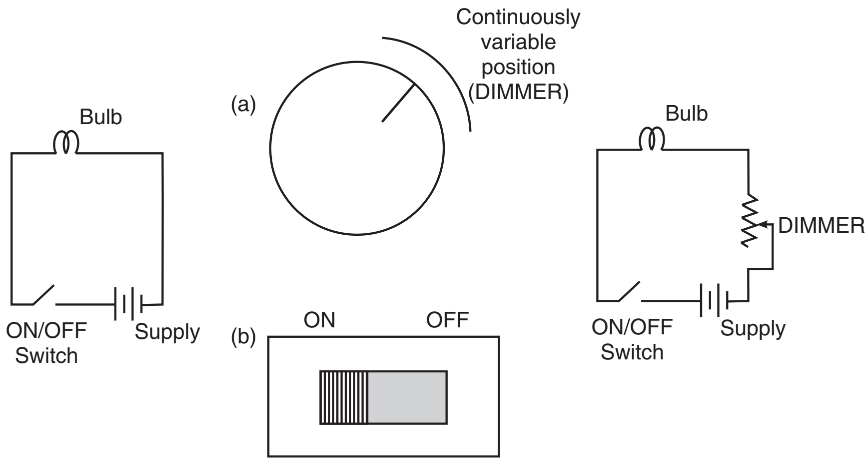


Fig. 3.2: Analog and digital signal examples

Where as in digital, signals have only two distinct values or state i.e. either on/off or 0/1, whenever we light up on electrical lamp (Fig. 3.2 L.H.S), through a simple electrical switch it has only two positions if ON/OFF assign for light and dark, if the same bulb is connected through a dimmer in the place of switch Fig. 3.3 R.H.S), then the fall/rise of brightness seen by human eyes smoothly because the brightness has several intermediate values, which is continuous in nature.

Electrically the analog signal is represented as a varying voltage or current like a sine wave below (a).

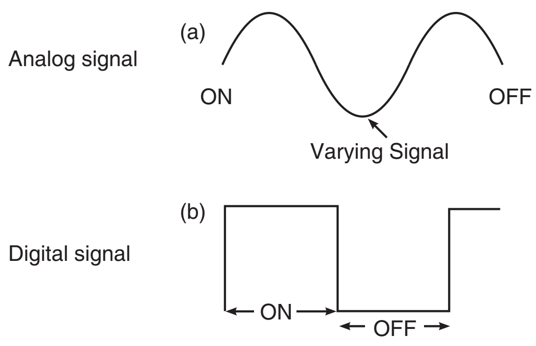


Fig. 3.3

Where as a digital signal is represented by a square wave shown in above figure (b) above. You can see it has two distinct values only either '0' or '1'.

In any system, if the output is similar to the input that means the system is analogue system.

Where as in digital system, the output wave is not similar to the input.

3.6 ADVANTAGES OF DIGITAL SIGNAL OVER ANALOG SIGNAL

(i) Analog signals are prone to be affected by noise, whereas the noise added to digital signal doesn't matter. Because state '1' will be detected as state '1' and state '0' will be detected as state '0', despite addition of noise.

(ii) Analog signal can be recorded and played back but recording and playback process deteriorate the signal in terms of addition of noise, distortion, and change of frequency response.

(iii) In digital signal, the recording and play back process does not add up noise, distortion and change of frequency response.

(iv) With falling rates of computer, components for storage of digital signal has become quite cheap. Whereas analog signals are still being recorded on expensive magnetic medium or optical medium.

(v) Signals once converted into digital form, are much easier to store, manipulate and transport.

3.7 ADVANTAGES OF DIGITAL TECHNOLOGY OVER ANALOG TECHNOLOGY

The quality of digital audio is independent from the medium and depends only on the conversion process. The conversion of audio from analog to digital domain, provides the following advantages.

(i) In Analog system the errors caused by noise, distortion and jitter (due to long cables) cannot be removed fully. Whereas in digital system, these errors can be removed by easier and cheaper means.

(ii) Deterioration due to flutter, print through, drop out noise, alignment errors (change in angle of head) do not occur in digital systems.

(iii) In Digital technology, data can be copied infinitely without generation loss.

(iv) In Digital technology, the data can be accessed instantly whereas Analog technology makes this process complicated and lengthy. Hence Digital technology borrowed RAM (Random Access Memory) and Hard disk technology from computer industry to make DDS (Digital Data Storage).

(v) Editing in digital domain is easier than tape edit.

(vi) Digital audio broadcasting can be carried out in the digital domain with less interference, fading and multipath reception problem as compared to Analog broadcast. Hence the allotted bandwidth can be used more efficiently.

(vii) Maintenance is easier in Digital systems because digital equipment can have self-diagnosis program, built in the system, to point out its own failure or error.

(viii) In Analog – recording, editing and playback is linear in nature.

(ix) In Digital – recording, editing and playback is non-linear in nature. This saves time and enhances creativity.

(x) The cost of Digital equipment is much less than that of analog equipment and the size is also smaller due to integrated circuits & cheap technology.

INTEXT QUESTIONS 3.1

1. Sound signal is first converted into .... signal before processing

   (a) Voltage (b) Electrical

   (c) Magnetic (d) None of the above

2. Digital signal is stored in which form?
3. Digital recording converts the electrical wave into a series of decimal numbers. True or False.
4. Digital audio has advantages for mass production. True or False.
5. Is it true that an analog signal has infinite values between two limits?

3.8 ANALOG TO DIGITALCONVERSION (A/D CONVERSION)

Analog to digital conversion is a process by which an analog signal is converted to a series of binary digits which represents its value at different sampled point.

First the analog audio signal, which is a time varying continuous electrical voltage or current is passed through an A/D converter (i.e. Analog to Digital Converter). In this process the audio signal is sampled many thousands time per second and converted into a series of samples, which are the snapshots of audio signal taken at the time of each sampling and each such sample is represented by a number. See figure 3.4 below.

Notes

A sample and hold circuit can be used for obtaining these samples

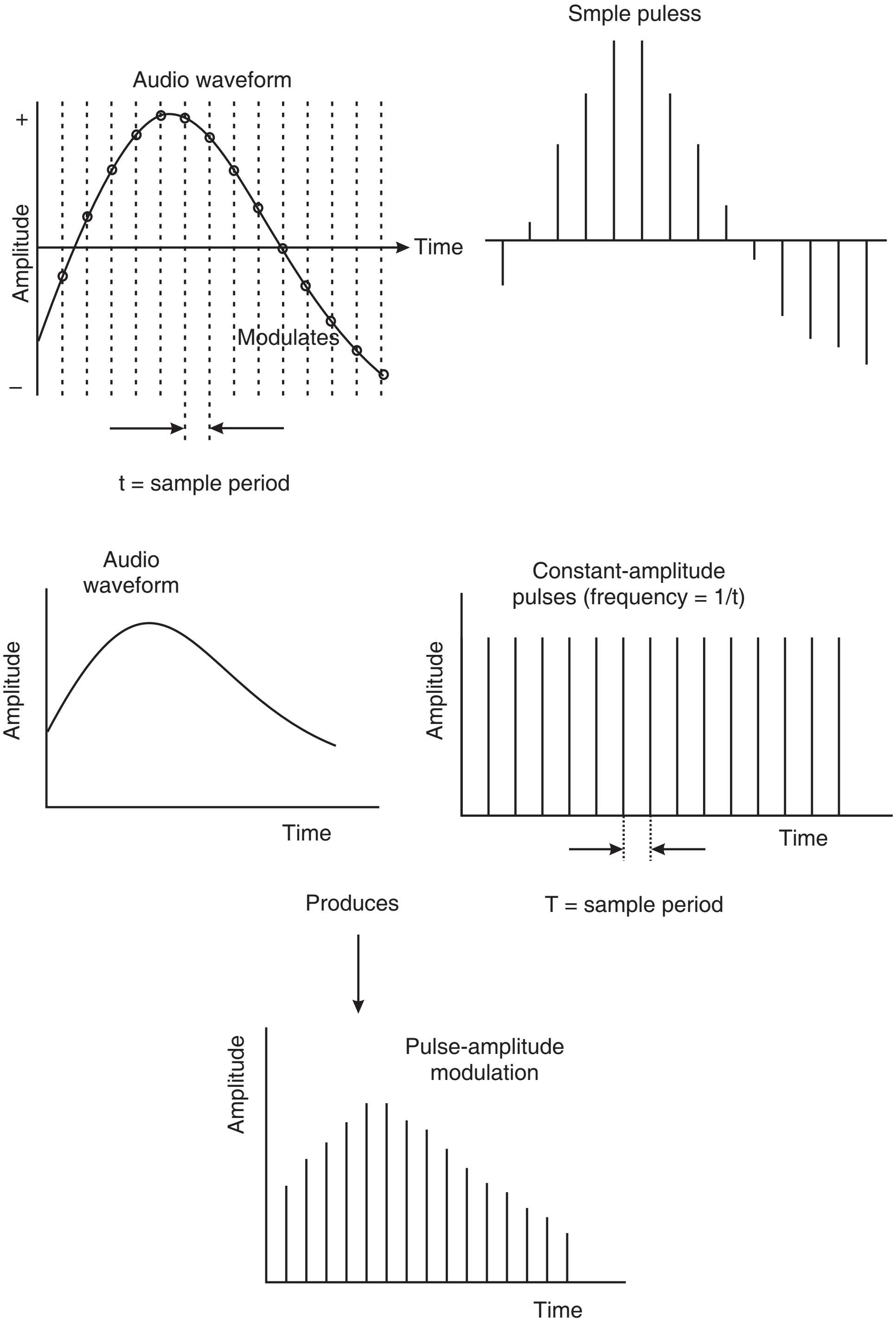


Fig. 3.4: Sampling of analog signal and corresponding pulse levels

The sample pulses represents the instantaneous amplitude signals at each point in time interval, the samples can be considered as "still frames" or "snap shots" of the continuous audio signal, which when put together serially in a sequence for a continuous pulse forms of the audio signal. In order to represent the details of the audio signal it is necessary to take large number of samples per second.

In order to convert analog signal into digital signal it is necessary to measure its amplitude at specific point in time, is called "Sampling" the process of assigning a binary digital value to each measurement called quantization.

Do you know?

The mathematical theory by "Shannon" says at least two samples per audio cycle must be taken to convey necessary information about the signal or in the other words sampling frequency must be at least twice as high as the highest audible sequences in practice.

The quality of A/D conversion is determined by

1. Sampling rate
2. Quantization
3. Dynamic range/Bit depth
4. Dithering

3.8.1 Sampling Rate/Frequency

The choice of sampling frequency determines the maximum audio bandwidth available. Sampling frequency is at least twice the highest audio frequency to be sampled. Since, the audio frequency band extends up to 20Khz, implying the need for a sampling frequency of just over 40Khz for high quality audio work. There are in fact two standard sampling frequency between 40 and 50 kHz, the compact disc rate is 44.1khz and so called "professional" rate is of 48 kHz. These are both allowed in original AES standard (Audio Engineering Society) which sets sampling frequencies for digital audio equipment.

Audio Sampling frequencies

Frequency (kHz)	Application
8	Telephone (speech quality) standard
32	Used some broadcast coding system e.g. NICAM, DAT Long Play mode
44.1	CD Sampling frequency, AES secondary rate.
48	AES Primary rate for professional application basic rate for blue-ray disk (which no longer specifies 44.1Khz as an option)

96	This is optional for DVD-Video, DVD Audio & Blue ray disks for AES secondary rate for high bandwidth application
192	Four times the basic standard rates. Optional in DVD-Audio. 192Khz is the highest sampling frequency allowed on Blu-Ray Audio disks.

3.8.2 Quantization

After sampling, the the sampled/ modulated pulse chain is quantized. In quantizing the range of sample amplitude is mapped into a scale of stepped binary values.

The quantization determines which of a fixed number of quantizing intervals (size of Q) represented. This is done so that each sample amplitude can be represented by a unique binary number in pulse code modulation (PCM).

PCM: It is the form of modulation in which signals are represented as a sequence of sampled and quantized binary data words.

In linear quantizing, each quantizing step represents an equal increment of a signal voltage. Most high quality audio systems use linear quantizing.

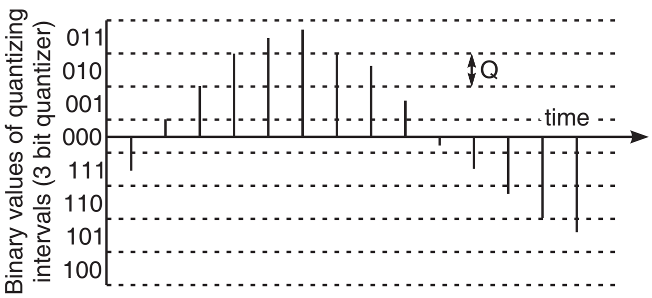
Quantizing error: it is an inevitable side effect in the process of A/D conversion and the degree of error depends on the quantizing scale, higher the scale lower the quantizing error.

- 4 Bit scale offers 16 possible steps (more error)
- 8 Bit scale offers 256 possible steps
- 16 bit scale offers 65536 possible steps. (lesser error)

The quantized output of an A/D convertor, can be represented in either serial or parallel form

3.8.2.1 Quantization Resolution

The quantizing error may be considered as unwanted signal added to the wanted signal as shown in Fig. 3.5 and Fig 3.6 below.



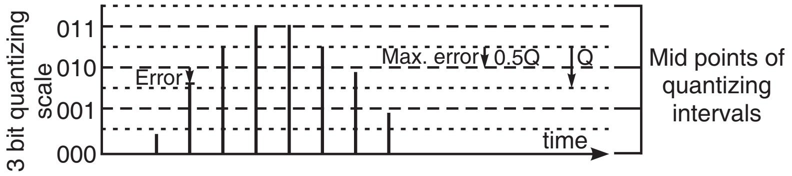


Fig. 3.5 : Quantization errors

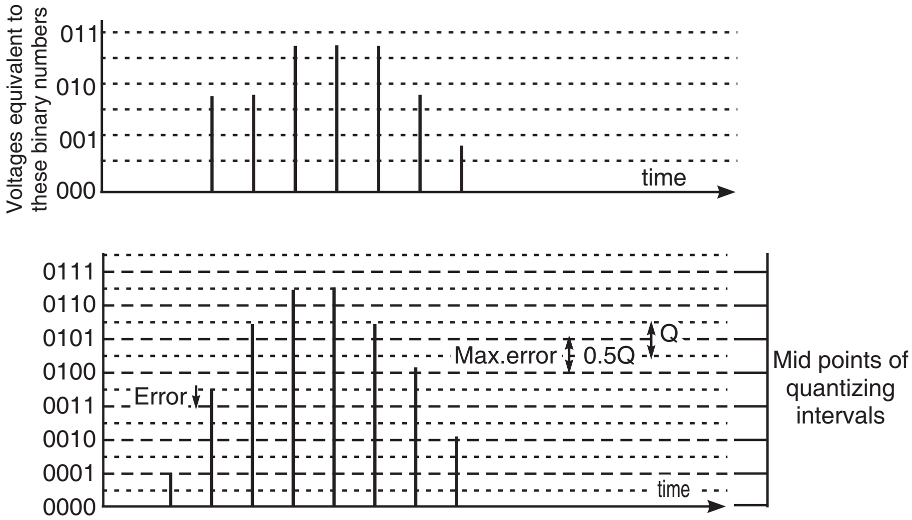


Fig. 3.6: Higher level Quantization and lesser error

The error is classified either as distortion or noise, depending on their characteristics and the nature of the quantizing error signal.

The bit rate of the digital signal is therefore is the multiplication of sampling frequency with the quantization bits used. For example if a audio signal is sampled at 48 kHz and 16 bit quantization is used, then the bit rate of digital signal will be 48*16= 768 Kbps(kilo bits per second). If compressed by four it will become 192 Kbps for a mono(single channel signal) and 192*2= 384Kbps for stereo signal.

3.8.3 Dynamic Range/Bit Depth

The human hearing capabilities should be regarded as the standard against which the quality of digital systems is measured, since it could be argued that the only distortion and noise that matter are those that can be heard, Louis Fielder and Elizabeth Cohen work out to establish dynamic range requirements for high quality digital audio systems, Fielder was able to show, what was likely to be heard at different frequencies in terms of noise and distortion and where the limiting elements might be in a typical recording chain, they determined a dynamic range requirement of 122 dB for natural reproduction, taking into account microphone performance and the limitations of consumer loudspeakers. This requirement dropped to 115dB for consumer systems.

When there are no more bits available to represent a higher level signal at this point the waveform will be hard clipped and will become highly distorted. The effect too is very different from that encountered in analog tape recorders which tend to produce gradually more distortion as the recording level increases. Digital recorders remain relatively undistorted as the recording level rises until the overload point is reached at which point very bad distortion occurs. The number of bits per sample therefore dictates the signal to noise ratio of a linear PCM digital audio system 16 Bit Linear PCM was considered as normal for high quality audio application for many years. This is the CD standard and is capable of offering a good S/N ratio range over 90db, but it fails to reach the psychoacoustic idea of 122db for subjectively noise free reproduction in professional systems.

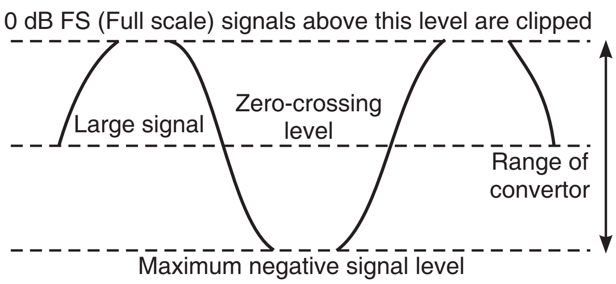


Fig. 3.7: Clipping of signal

For professional recording purpose and avoidinf clippin (Fig. 3.7) we may need a certain amount of head room for distortion free recording/reproduction.

Headroom

Unused dynamic range above the normal peak recording level which can be used in unforeseen circumstances such as when a signal overshoot its expected level this can be particularly necessary in live recording conditions where one is never quite sure what is going to happen with recording level.

This is another reason why many professionals feel that a resolution of greater than 16 Bit is desirable for original recording. Twenty and 24 bit format are become very popular for this reason.

Quantizing Resolution

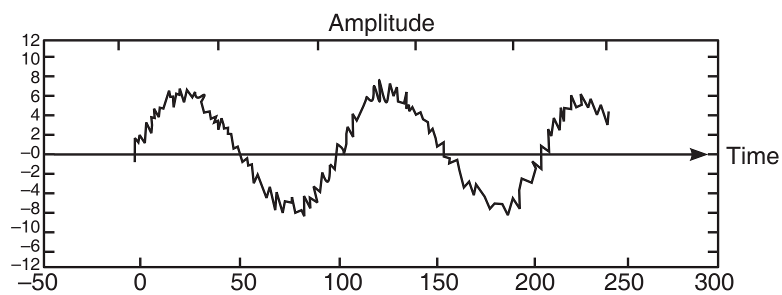
The table shows some commonly encountered quantizing resolutions and then application

Notes

Bits per sample	**Approximate dynamic range with dither (db)**	**Application**
8	44	Low moderate quality for older PC Internal sound generation and some older multimedia application.
12	80	Original EIAJ format PCCM adaptors such as Sony PCM-100
16	92	CD-standard, DAT standard commonly used high quality resolution for consumer media, some professional recorders and multimedia pcs. Usually twos compliment binary numbers.
20	116	High quality professional audio recording and mastering applications
24	146	Maximum resolution of most recent professional recording system also of AES 3 digital interface, dynamic range exceeds psycho-acoustic requirements

3.8.4 Dither

Dither in A/D conversion has the effect of linearzing a normal convertor (in other words it effectively makes each quantizing interval the same size) and turns quantizing distortion into a random, noise line signal at all times. (Fig. 3.8 below)



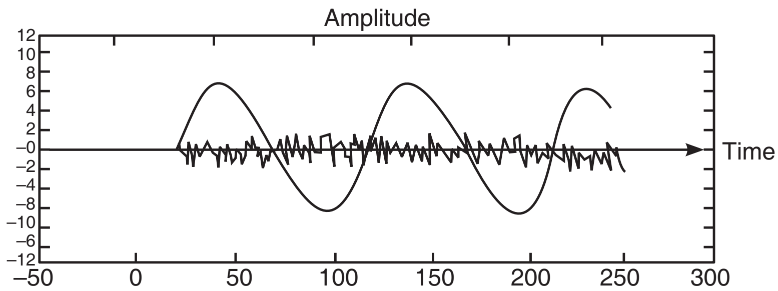


Fig. 3.8: Dithering or linearization

This is used for number of reasons

1. It allows signal to be faded smoothly down without the sudden disappearance.
2. It allows the signal to be reconstructed even when their level is below the noise floor of the system.
3. The white noise at very low level is less subjectively annoying than distortion.

Undithered audio signal begin to sound "grainy" and distorted as the signal level falls. Quiescent hiss will disappear if dither is switched off making a system seem quieter but a small amount of continuous hiss is considered preferable to low level distortion. The resolution of modern high resolution convertors is such that the noise floor is normally inaudible in any case.

Quiescent Hiss – The hiss generated from the machine during standby mode.

White Noise – This is a noise in which all frequencies have same amplitude or in same level.

3.9 DIGITAL TO ANALOG CONVERSION (D/A CONVERSION)

In D/A conversion, the audio sample words are converted back into stair case like chain of voltage levels corresponding to the sample values. This is achieved simple D to A convertors, by using the states of bits to turn current sources on or off, making up the required pulse amplitude by the combination of outputs of each of these sources.

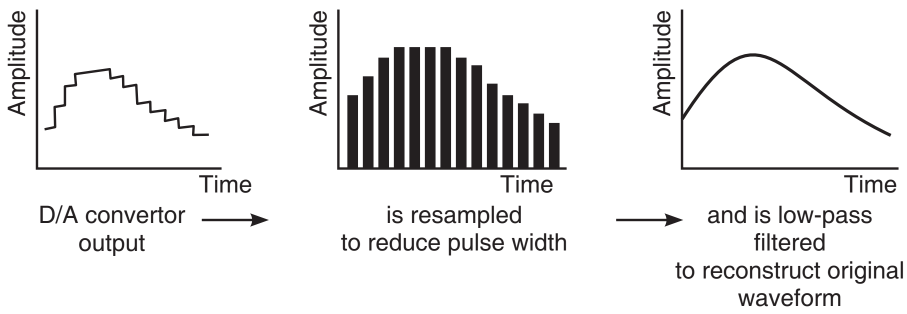


Fig. 3.9: Digital to analog conversion

These staircases are then resampled to reduce the width of the pulses before they are passed through a low pass reconstruction filter whose cut off frequency is half the sampling frequency. The effect of the reconstruction filter is to join up the sample points to make a smooth wave form as shown in figure 3.9 above.

Re sampling is necessary to avoid any discontinuities in signal amplitude at the sample boundaries, otherwise the averaging effect of the filter would result in a reduction, in the amplitude of high frequency audio signal.

INTEXT QUESTIONS 3.2

Write down the correct option from the ones given below:

1. Converting analog signal into digital signal by measuring amplitude at specific points is called

   (a) Quantization (b) Dithering

   (c) Sampling

2. The process of assigning a binary digital value of each sampled measurement is called

   (a) Sampling (b) Quantization

   (c) Dithering (d) Dynamic range

3. Sampling frequency determines at least .... the highest audio frequency.

   (a) Once (b) Twice

   (c) Thrice

4. What is the sampling rate for a Compact Disc?

5. Match the following:

Sample rate	Application
48	Telephone
32	DVD – Audio
96	CD – Audio
8	Broadcast coding
44.1	Basic Blu ray disk

6. According to Fielder what is the dynamic range for natural reproduction?
7. Dither allows a signal to be faded smoothly without the sudden disappearance .True or false
8. What is quiescent hiss?
9. What is white noise?

3.10 WHAT HAVE YOU LEARNT

After reading the lesson, we have learnt about the different types of signals—analog and digital. We saw, how sound is first converted into an electrical signal when it enters a microphone.

We then proceeded to learn about the advantages of digital signal over analog signal. How digital signal lends itself to compression and therefore occupies less space.

The process of analog to digital conversion and vice-versa were two other points covered in this lesson. :

3.11 TERMINAL QUESTIONS

1. Briefly define analog and digital signal? What are the main differences between them?
2. What are the advantages of digital signal over analog signal?
3. What are the advantages of digital technology over analog technology?
4. Explain analog to digital (A/D) conversion.
5. Briefly describe the following terms.
   (a) Sampling Rate

(b) Quantizing

(c) Dithering

6. Explain dynamic range and list a few applications?

7. Explain digital to analog conversion.

3.12 ANSWERS TO INTEXT QUESTIONS

3.1

1. Electrical
2. Binary
3. False
4. True
5. Yes, true.

3.2

1. Sampling
2. Quantization
3. Twice
4. 44,100 Hz (or 44.1kHz)
5.

Sample rate	Application
48	Basic Blu ray disk
32	Broadcast Coding
96	DVD - Audio
8	Telephone
44.1	CD - Audio

6. 122 dB

7. True.

8. The hiss generated from the machine during standby or sleep mode.

9. The noise in which all frequencies have same amplitude or in other wards in same level