INTRODUCTION TO MUSIC PRODUCTION – WEEK3

ANALOG TO DIGITAL CONVERSION PROCESS - BY AKASH KODIGANTI

Audio signals which mean, the harmony of waveforms that travel in space(analog) with an orderly sequence of arrangement, so as to produce a unified and continuous propagation of waves.

Converting these audio signals to Digital form greatly improvised in this 20th century.

Gramophone

Magnetic tape cassette

Digital Music

Let's get into the current way of conversion process i.e., Digital Music.

Steps:

1. Sampling
2. Quantization
3. Encoding

1.SAMPLING – by Shannon

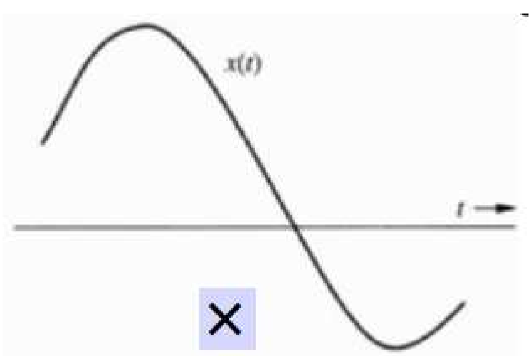
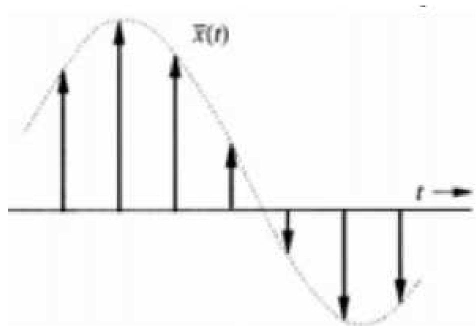


Fig1. Original clip of an audio signal

Why Sampling?

Because we can't process those signals and can't generate sounds. It is the computers and digital devices that makes up ADC.

For computer to analyze in Zeros & Ones sampling has been introduced.

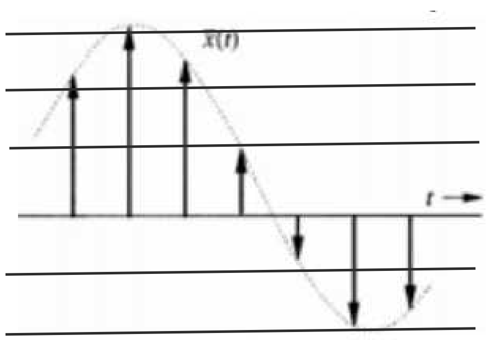


Sampling a signal provides 'samples' with respect to Horizontal axis.

Computer operations are performed easily after sampling a signal. After completing computer operations, a sound file need to be reconstructed from the samples.

Fig2. After sampling the above signal

2.Quantization – by Shannon



The signal is then projected to quantization where only specified levels on Vertical axis are allowed to load the volume (or strength) of a signal.

Rounding/truncation are the methods used which leaves off a quantization error due to lossy-compression.

Fig3. Quantization of the signal

3.Encoding – by Shannon

Encoding is a process in which the quantized levels are given a zero-one notation

Voltage Level	bits	Voltage Level	bits	Voltage Level	bits
2 volts	0	1 volt	00	0.5 volts	000
4 volts	1	2 volts	01	1 volt	001
		3 volts	10	1.5 volts	010
		4 volts	11	2 volts	011
				2.5 volts	100
				3 volts	101
				3.5 volts	110
				4 volts	111

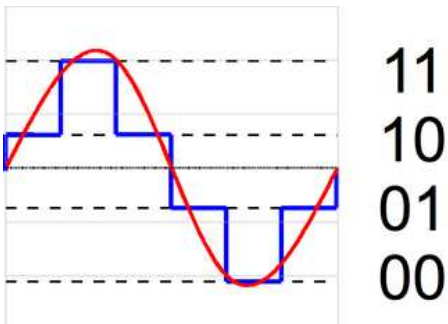
Fig4 One-bit encoder, Two-bit encoder & Three-bit encoder

The above table shows the number of ways we can encode a signal of our interest. Like in our example i.e., the signal presented in Fig1 has quantized to 6 levels in Fig2.

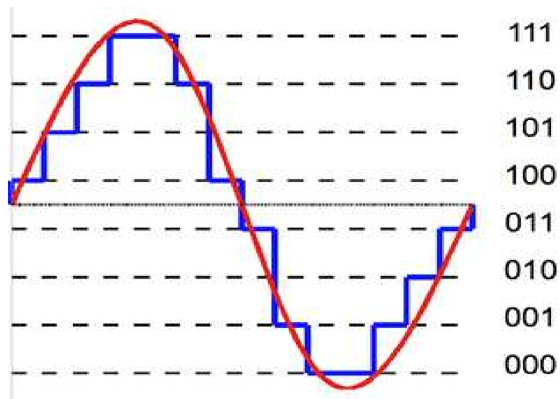
Hence, in this context we need at least 6 values to be encoded which works with Three-bit encoder.

Using 3 bit i.e., this, provides us to encode 8 levels of which we use 6 (not a big deal).

For better understanding images are provided below:



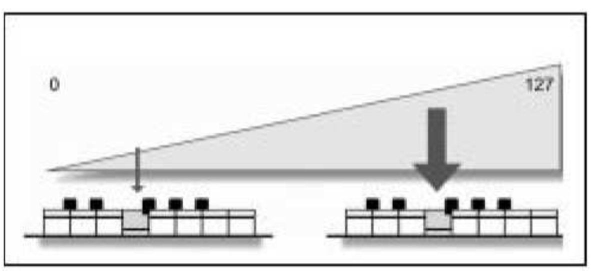
Two-bit encoder



Three-bit encoder

The above is **'2 power N'** logic presented by our course.

Note:

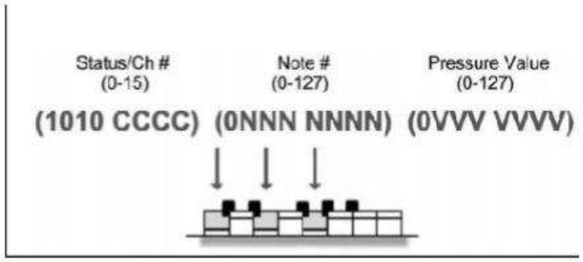


Pressure variation in pressing key

The measures on vertical axis (**under quantization)** depends on how forcedly you are pressing the keys on keyboard.

If 7bits are allocated to note down the level, like in the Fig4 lower pressure on key makes a low measure (say 20) and higher is around 100. (Since 2^(7) =128 i.e., 0 to 127)

Fig4. Levels used for encoding volume



Polyphonic key pressing

The polyphonic key takes values such as Volume level, note frequency (musically saying pitch).

Basically, 4 parameters namely, ADSR – Attack, Delay, Sustain, Release defines the **encoding process**.

Additional information:

1. Audible frequency range – 20 Hz to 20,000 Hz
2. Sampling frequency used in:
    a. Studio environment – 44100 Hz
    b. CD Quality – 44100 Hz
    c. Video-graphics – 48000Hz
3. Nyquist Theorem, which rules out to make Sampling frequency as double that of maximum audible frequency (i.e., 20000Hz). Hence we are having 44,100Hz.
4. Trade off: Changing the sampling frequency changes these as follows

Sampling frequency	Time period of the audio	Maximum frequency
44,100 Hz	60 Seconds	20,000 Hz
88,200 Hz	30 Seconds	40,000 Hz
22,050 Hz	120 Seconds	10,000 Hz

5. Encoding of:
    a. MIDI takes – 7 bits (128 levels)
    b. Digital audio/ CD takes – 16 bits (65 thousand levels)
    c. Studio recording takes – 24 bits (167 lac levels)