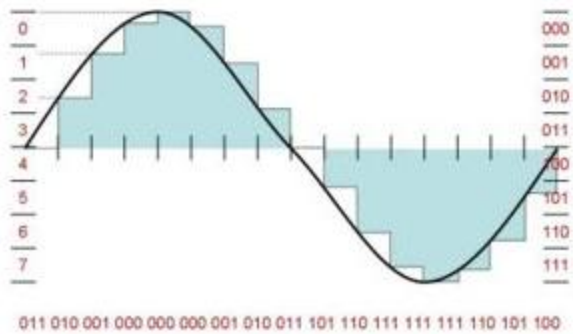
Experiment
Pulse Code Modulation Decoder (PCM)

Objectives

1- Introduction to PCM Decoding and Digital-to-Analog Conversion.
2- To understand the operation theory of pulse coded modulation (PCM) Decoder.
3- To understand the theory of reconstruction the massage by passing the signal through a low- pass-filter .

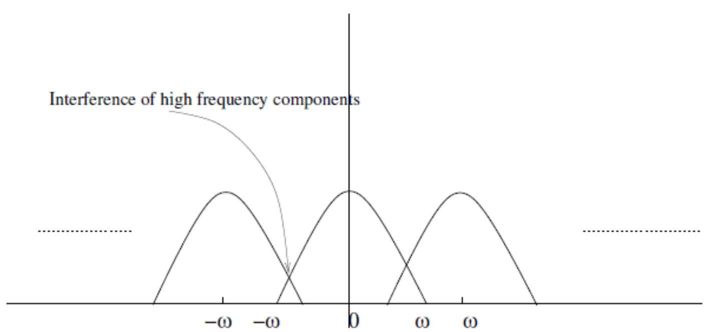
Basic Information

A PCM decoder for converting to an analog voice signal an 8-bit PCM signal the first bit of which is a polarity specifying bit, the PCM decoder comprising a capacitor array having binary-weighted capacitors and a resistor string circuit having plural resistors for dividing a reference voltage to obtain different tap voltages, wherein the tap voltages corresponding to the four lower bits of the PCM signal are derived from the resistor string circuit and the combination of the reference voltage and each of the tap voltages, made according to the contents of the second, third and fourth bits of the PCM signal is applied to the corresponding one of the capacitors in the capacitor array circuit whereby the capacitor array circuit delivers an analog voltage signal corresponding to the received signal, the resistor string circuit having two groups of intermediate taps so that the conversion characteristic for obtaining voltages in the signaling frame may be different from that in the non-signaling frame.



Aliasing

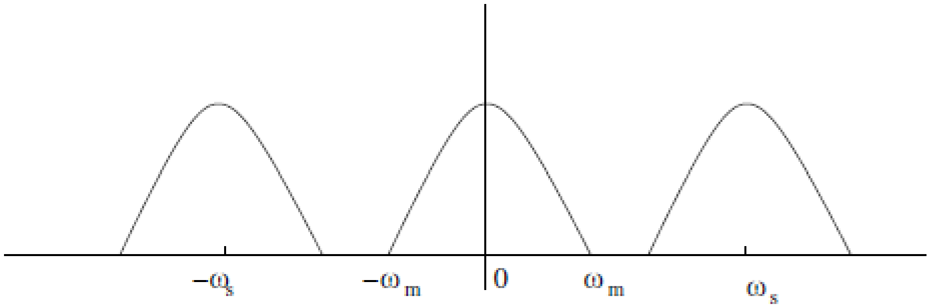
Aliasing is a phenomenon where the high frequency components of the sampled signal interfere with each other. because of inadequate sampling $\omega s < 2\omega m$.



Aliasing leads to distortion in recovered signal. This is the reason why sampling frequency should be at least twice the bandwidth of the signal.

Oversampling

In practice signal are oversampled, where Fs is significantly higher than Nyquist rate to avoid aliasing.



Message Reconstruction

You can see, qualitatively, that the output is related to the input. The message could probably be recovered from this waveform. But it would be difficult to predict with what accuracy.

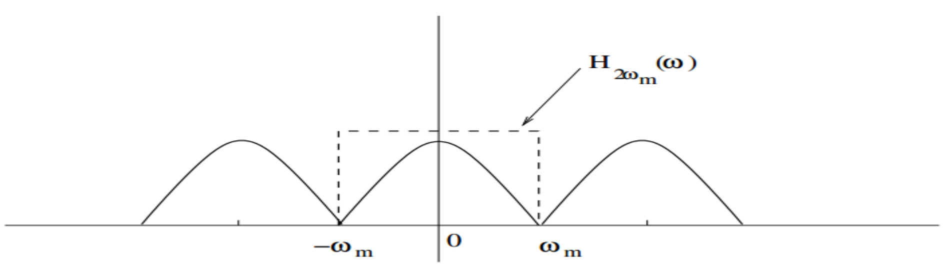
Low-pass-filtering of the waveform at the output of the decoder will reconstruct the message, although theory shows that it will not be perfect. It will improve with the number of quantizing levels. If any distortion components are present they would most likely include harmonics of the message. If these are to be measurable (visible on the oscilloscope, in the present case), then they must not be removed by the filter and so give a false indication of performance. So we could look for harmonics in the output of the filter. But we do not have conveniently available a spectrum analyzer. An alternative is to use a two-tone test message. Changes to its shape (especially its envelope) are an indication of distortion, and are more easily observed (with an oscilloscope) than when a pure sine-wave is used. It will be difficult to make one of these. But there is provided in the PCM ENCODER a message with a shape slightly more complex than a sine-wave. It can be selected with the switch SW2 on the encoder circuit board. Set the left hand toggle UP, and the right hand toggle down. A message reconstruction LPF is installed in the PCM DECODER module.

To recover the original signal G(ω):

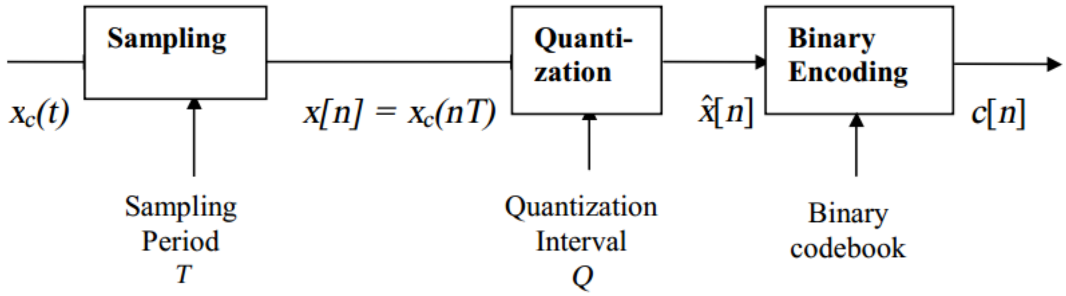
1- Filter with a Gate function, H2ωm(ω) of width 2ωm

2- Scale it by T

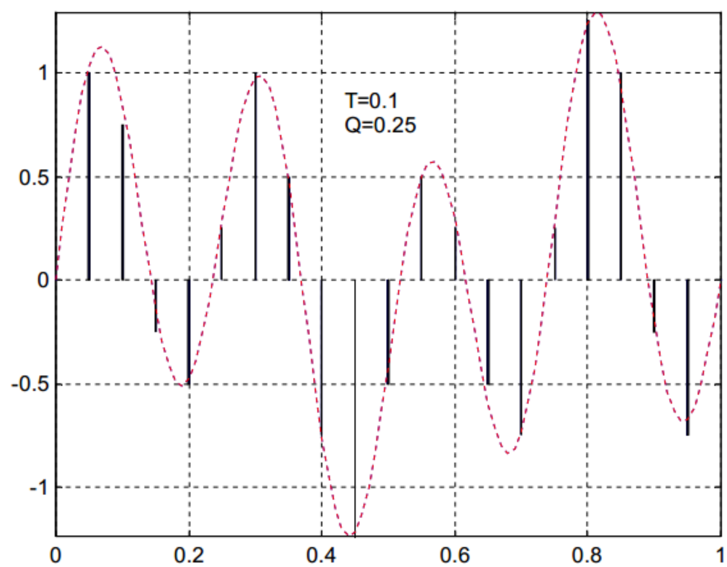
$$G(\omega) = TG_s(\omega)H_{2\omega_m}(\omega).$$



2. Quantization
Uniform

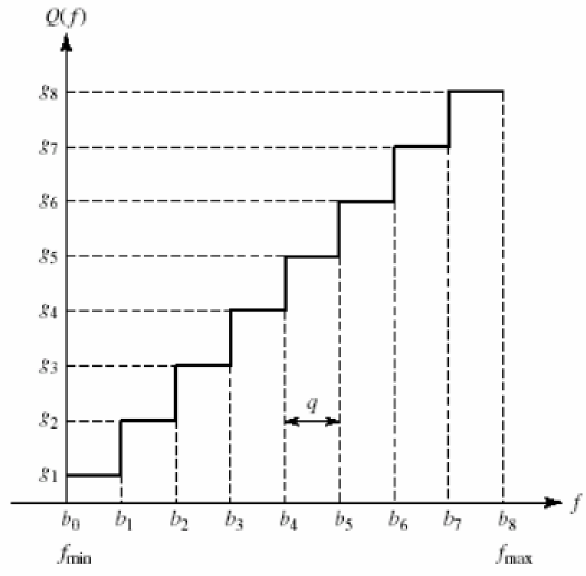


• Sampling: take samples at time nT
T: sampling period;
Fs = 1/T: sampling frequency.
• Quantization: map amplitude values into a set of discrete values kQ; where k is integer
Q: quantization interval
• Binary Encoding – Convert each quantized value into a binary codeword



How to Determine sampling period and quantization interval?
• T (or Fs) depends on the signal frequency range.
– A fast varying signal should be sampled more frequently!
– Theoretically governed by the Nyquist sampling theorem
• Fs > 2Fm (Fm is the maximum signal frequency)

• Q depends on the dynamic range of the signal amplitude and perceptual sensitivity Q and the signal range D determine bits/sample R
• 2R =D/Q
• One can trade off T (or Fs) and Q (or R), lower R ➔ higher Fs; higher R ➔ lower Fs.



• Applicable when the signal is in a finite range (Fmin,Fmax)
• The entire data range is divided into L equal intervals of length Q (known as quantization interval or quantization level)

$$Q=(f_{max}-f_{min})/L$$

• interval I is mapped to the middle value of this interval .
• We store/send only the index of quantized value.

•Index of quantized value

$$=Q_i(f)=\left\lfloor \frac{f-f_{\min}}{Q} \right\rfloor$$

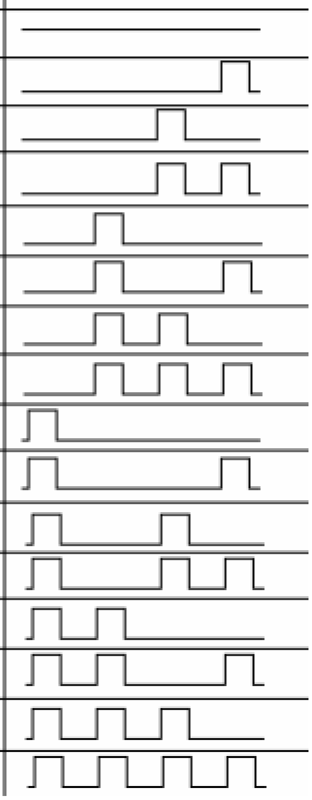
•Quantized value

$$=Q(f)=Q_i(f)Q+Q/2+f_{\min}$$

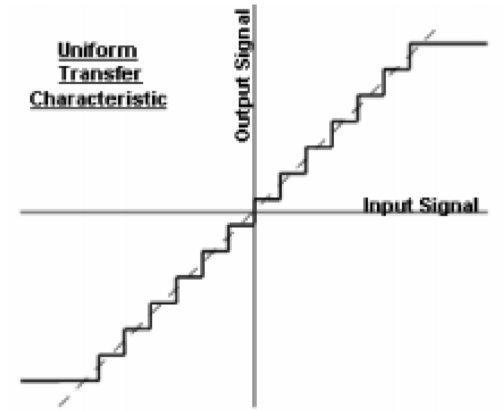
As stated before, in PCM, the information signal x(t) is first sampled with the appropriate sampling frequency (sampling frequency Fs ≥ 2×highest frequency of the information signal (Fx)), then the sampled levels are quantized to appropriate quantization levels. In the last step, each quanta level is demonstrated by a two-code word, that is by a finite number of {0,1} sequence. After this step, the signal is called as PCM wave.
If the max and min amplitude values of information signal x(t) are A_{max} and A_{min}, respectively, and if n-digit code words will be used, then the quantizing interval/pace "a" becomes:

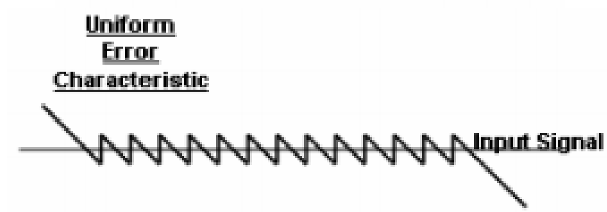
$$a=\frac{A_{\max}-A_{\min}}{2^n}$$

Digit	Binary Equivalent	PCM Waveform
0	0000	
1	0001	
2	0010	
3	0011	
4	0100	
5	0101	
6	0110	
7	0111	
8	1000	
9	1001	
10	1010	
11	1011	
12	1100	
13	1101	
14	1110	
15	1111	



The signal is divided into 16 amplitude levels (0-1.5) between its max and min values. Therefore, n=4 and the quantizing pace a =0.1. If the quantizing levels are selected equally, then this is called as “linear quantizing”.





A **little information about the PCM Encoder module on the Emona FOTEx**

The PCM encoder module uses a PCM encoding and decoding chip (called a codec) to convert analog voltages between -2.5V and +2.5V to a 7-bit binary number. with seven bits, its possible to produce 128 different number between 0000000 and 1111111 inclusive, this in turn means that there are 128 quantization levels (one for each number).

Each binary number is available on the PCM encoder modules output in serial from in 8-bit frames. The binary number's most significant bit is sent first and so is found on bit-7 of the frame. The numbers next most significant bit is sent next and so on to the least significant bit (which is found on bit-1 of the frame). Bit-0 of the frame is a frame synchronization but used by the PCM decoder module to find the beginning of each frame. It simply alternates between 0-1 on successive frames.

The PCM encoder module also outputs a separate Frame Synchronization signal FS that goes high at the same time as the frame's synchronization but is outputted. The FS output is not needed by the PCM decoder module and has been provided on the FOTEx purely for the Scope triggering.