Lab-Report
Digital Communications

Pulse Code Modulation

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1. Contents

2. Introduction

Pulse **C**ode **M**odulation (PCM) was invented in the 1920s by P.M. Rainy and rediscovered by A.H Reeves in 1939, but it couldn't be practically used before the 1960s because of the cost and size of the large amount of vacuum tubes needed for a realisation. The first working PCM communication system was introduced in the early 1960s by the Bell Laboratories.
The importance of PCM systems grew up whit the mass production of cheap transistors and digital circuits.
Objectives of the PCM lab were to demonstrate different stages in the conversion of an analogue waveform into a binary signal and the recovery of the original signal at the receiver.

3. Pulse Code Modulation

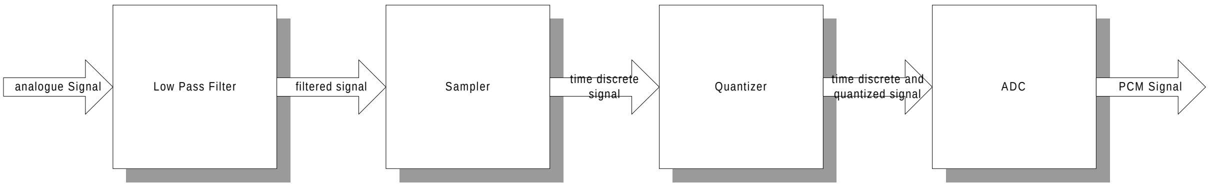


figure 1

Figure 1 shows a block diagram of a typical PCM encoding system. First the incoming analogue signal is lowpass filtered with a cut off frequency f_c lower than half the sampling frequency f_s to prevent aliasing.
The low pass filtered signal is than converted into a time discrete signal by means of a sampler. This sampled signal is called a pulse amplitude modulated signal (PAM).
The quantizer changes the PAM signal into signal, which is discrete in its amplitude too and the encoder converts the quantized signal into a bitstream. This bitstream is called a PCM signal.

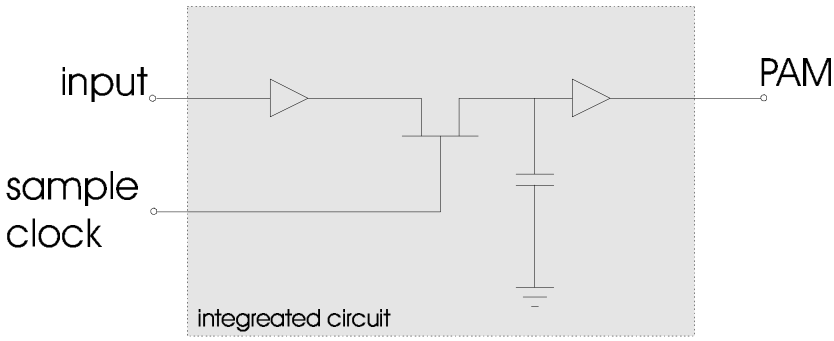


figure 2 – Sampling circuit

The figure on the following page (figure 3) shows the different steps of converting an analogue signal into a PCM bitstream:

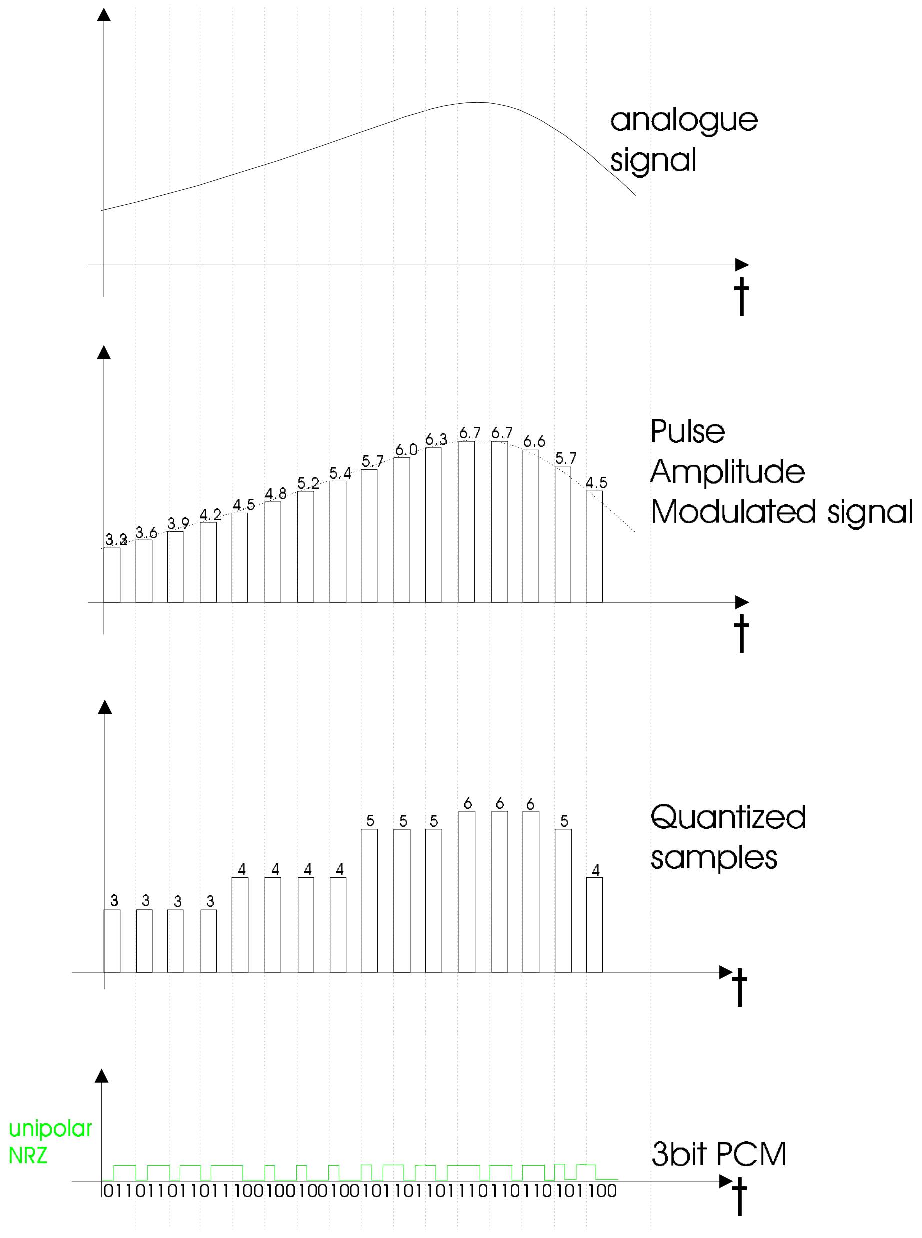
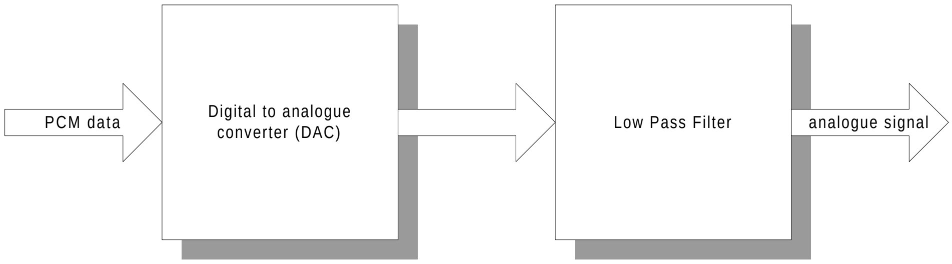


figure 3

The process of converting an analogue signal into a binary signal is also known as “Analogue to Digital Conversion”.

4. Reconstructing at the receiver

The PCM data are after receiving through a usually non ideal channel reconstructed at the receiver to obtain the original fed in signal.



The quality of the reconstructed signal is at all depending on the resolution and so on the number of different bits transmitted. Usually PCM systems work wit 8 or 16 data bits.

5. Lab Objectives

a) Equipment

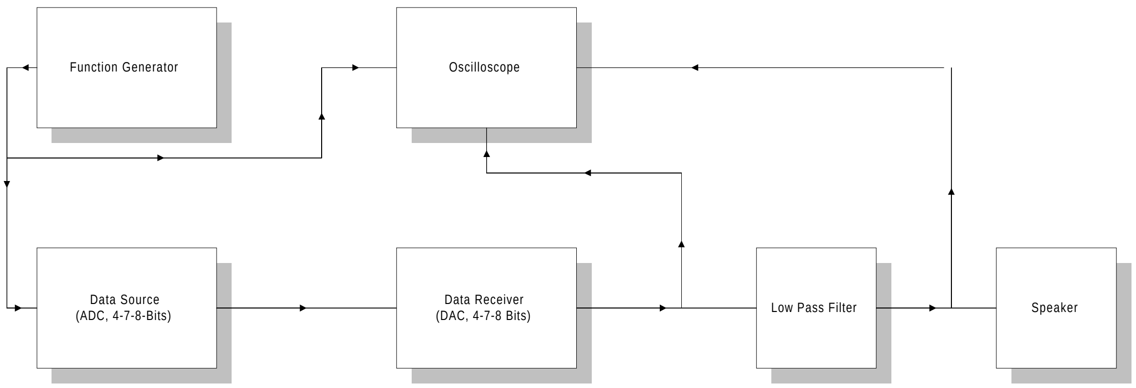


figure 4 - block diagram

The lab equipment consists of a function generator which generates the analogue signals, a data source module which generates a PCM data stream from the analogue signal, the data receiver which recovers the analogue signal from the PCM data stream and a low pass filter to filter the recovered data, an oscilloscope to obtain plots of the different waveform and at last a speaker to convert the analogue signal into sound.
The Data Source works with an sample frequency f_c=10kHz. So the maximum theoretically input frequency is f_m=5kHz.

b) Delay between Input and Output of the system

First after connecting the lab equipment together, setting to 8-Bit sampling and connecting a 100Hz audio signal with an amplitude of $4V_{PP}$ to the inputs of the data source module was to measure the delay due to the two conversions: Analogue to digital at the data source module and digital to analogue at the data receiver module.
The oscillospcope was connected to the output of the frequency generator and to the output of the data receiver module. A plot of both signals can be found at appendix 1.
The delay of 280µs between the input and output signal was automatically determined by the HP digital storage oscilloscope. The stepping of the output signal is caused by the 8-Bit quantization at the encoder and decoder.

c) Steppings

Next step was to raise the signal frequency from 100Hz to 200Hz and then to 500Hz and to determine the number of steps, the cycle and word period.

f_{signal}/Hz	100	200	500
t_{cycle}	10ms	5ms	2ms
$t_{word}=t_{cycle}$/steps	217µs	200µs	181µs
steps	46	25	11

The word period depends only on the sample frequency of 10kHz. Every sample the data source module generates an 8-bit word of about 200µs duration.
The number of steps depends on the frequency of the input signal. The slower the input signal is, the more often it can be sampled and the higher is the number of maximum steps.
The stepping of the 100Hz signal can be found at appendix 2.

d) Signal quality

Next step of the lab was to change the input frequency back to 100Hz and the 2nd channel of the oscilloscope was to connect to the output of the low pass filter. The stepping, which was in the part before clearly visible was now disappeared because of the low pass filter, which filters out the high frequency parts of the recovered signal (the stepping). Turning on the loudspeaker and comparing with the output of the frequency generator there was no difference hearbable.
A plot of the filtered signal and the generator output can be found at appendix 3.

e) Aliasing

Next step was to increase the signal frequency slowly up to 5kHz. At frequency around 5kHz the reconstructed signal became distorted, because the input frequency was greater than the Nyquist frequency f_N=0.5f_s and the signal can not be sampled well. Aliasing occurs.
5kHz is the last frequency, which can be theoretically sampled well at 10kHz sampling rate.
All higher input frequencies cause distortion, if they are not suppressed via a lowpass filter at the data source.
This distortions above 5kHz can be heard as spurious signals through the loudspeaker, if the lowpass filter is not connected. The lowpass filter suppresses all frequencies higher than 4.2kHz.
A plot of the unfiltered output signal with an input frequency of 5kHz can be found at appendix 4. The recovered signal is only a rectangular one. Any increase of the input frequency will result in distortions.

f) Low frequency response

when turning the signal frequency back to 100Hz and then down to 20Hz there was no signal through the loudspeaker hearable. But when connecting an oscilloscope to the lowpass filter output a very clear sine wave could be obtained: The system works well on low frequencies, which cannot be heard by a human ear.

g) Quantization noise

The quantization noise occurs due the finite number of signals by which the original analogue signal is represented. The continuously analogue signal is replaced by a finite number of codes. The difference between the original and the recovered signal is called the Signal to Quantization noise. It can be calculated by $S_{QNR}=S_{(dB)}$ + 4.8dB + 6dBn, where n is the number of bits used for the quantization and $S_{(DB)}$ the signal in dezibel.
The Signal to Quantization noise can be determined by subtracting the recovered from the original signal and putting them in ratio together. The ratio between the noise and the signal is called the Signal to Quantization noise.
A plot of the quantization noise of the lab equipment can be found as appendix 5.

6. Questions

a) Sampling

An Analogue signal having 4kHz bandwidth is sampled at 2.5 times the Nyquist rate and each sample is quantized into one of 512 levels, assumed equally likely. If successive samples are statistically independent,

i) what is the information rate of this source?

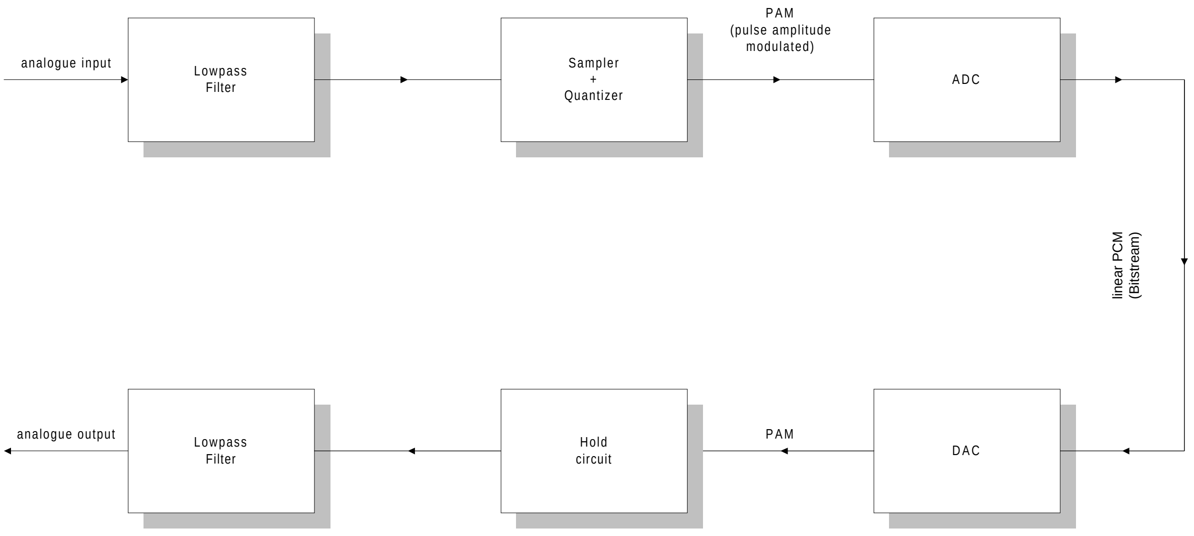
$M=2^n=512 \rightarrow n=9$
$f_m=4kHz$
$f_s=f_m \cdot 2 \cdot 2.5=20kHz$
$r=f_s \cdot n$
$r=20kHz \cdot 9=180kbps \rightarrow$ Information rate

ii) What is the S/N ratio required for error-free transmission over an AWGN channel with a bandwidth of 10kHz?

$(S/N)_{Pe=0}=6.02n=6dB \cdot 9$
$(S/N)_{Pe=0}=54dB$

b) PCM System

Draw a block diagram showing the main components of a complete PCM telecommunication system.



c) PCM System II

An analogue signal is to be converted to a binary PCM signal and transmitted over a channel that is bandlimited to 100kHz. Assume that 32 quantization levels are used and that the overall equivalent transfer function is of the raised cosine type with roll-off factor, r=0.6.

i) Find the max PCM bit rate, that can be used by this system without introducing intersymbol interference.

$$m = 32 = 2^n \quad \rightarrow \quad n = 5$$

$$B_W = \frac{1}{2} B_{Baseband}(1 + \text{rolloff}) \quad \text{where } B_{Baseband} = B_{PCM} = nf_s = r$$

$$B_W = \frac{1}{2} r(1 + \text{rolloff})$$

$$\underline{\underline{r_{max}}} = \frac{2B_W}{1 + \text{rolloff}} = \frac{2 \times 100\text{kHz}}{1.6} = \underline{\underline{125\text{kbps}}}$$

ii) Find the maximum signal bandwidth that can be accommodated for the analogue input signal.

$$r = nf_s$$

$$f_s = \frac{r}{n} = \frac{125\text{kbps}}{5} = 25\text{kHz}$$

$$\underline{\underline{B_{WAnalogue} = \frac{f_s}{2} = 12.5\text{kHz}}}$$

iii) Intersymbol Interference

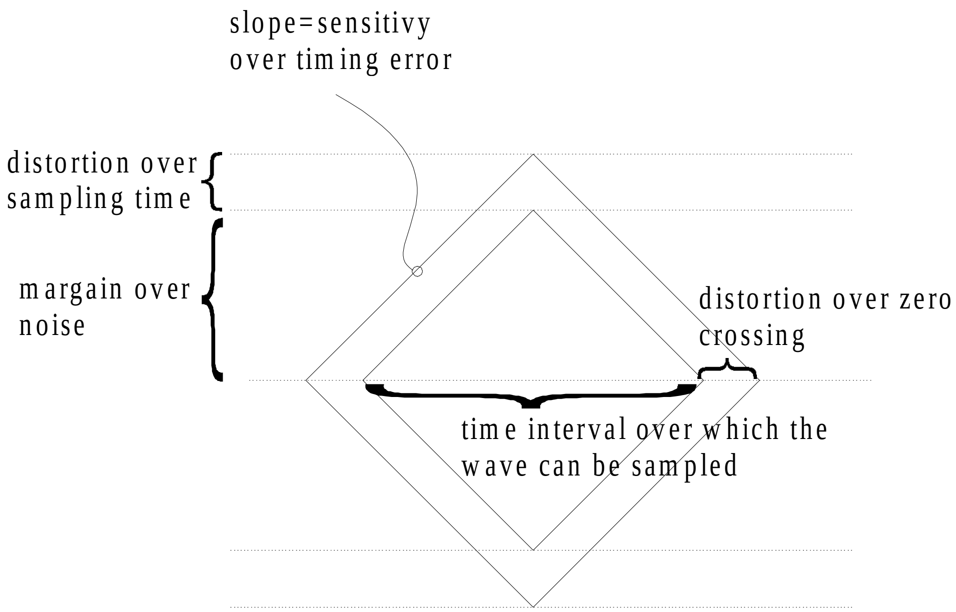
Explain the term intersymbol interference and how it can be reduced.

Due to limited bandwidth the sent rectangular impulses will be distorted and overlap into the other signals. A receiver detects the sum of the individual pulse responses and because of the overlapping between the signals, they can’t be exactly detected. This is called intersymbol interference → ISI.

iv) Eye Pattern

Describe how an eye diagram can be used to assess the level of intersymbol interference as well as other degradation.

An Eye Pattern or Eye Diagram is an easy way to show the level of ISI and other signal degradation.
An eye diagram is the superimposition of successive received symbols.
The more an eye pattern is opened, the better can a symbol be decoded without error.



Appendix 1 – Delay between the Input and Output of the System

Appendix 2 – 100Hz stepping

Appendix 3 – Input and Output after filtering

Appendix 4 – Aliasing – $f_{signal}=0.5f_{sample}$

Appendix 5 – Quantization noise