Experiment 4 – Sampling and PCM

Abstract

This experiment provides a device that can perform numerous functions. By connecting different modules, the process of converting an analogue signal into a digital signal is realised and a PCM coding system is used..

Declaration

Contents

1 Introduction

1.1 Objectives

The objectives of this experiment are: to study and practically test some important telecoms concepts such as sampling, PCM coding and decoding and bandwidth limitations. Gain practical experience using the Emona Telecoms-Trainer 101 suite.

1.2 Theoretical Background

In modern telecommunications, digital transmission has been on the increase since its introduction in 1962. This is due in large part to the fact that most communications providers require a high degree of accuracy in the information they transmit over their networks. With digital transmission (as opposed to analogue transmission), the system is a better switching interface, easier to multiplex and produces a clearer signal. Digital signals are described as discrete and variable on/off pulses, which are discrete variable on/off pulses, rather than continuously variable analogue signals. Each pulse is called a bit. Bits are the most common digital signal in the telecommunications industry. The number of bits transmitted per second is the bit rate of the signal. To convert analogue signals into digital signals, a coding system called Pulse Code Modulation (PCM) is used. This process requires other pre-processing and post-processing steps in which sampling is required. [1].

2 Materials and Methods/Procedure

2.1 Materials

- Emona Telecoms-Trainer 101 kit
- Oscilloscope TDS 210

- Different wires and BNC cables

2.2 Methods/Procedure

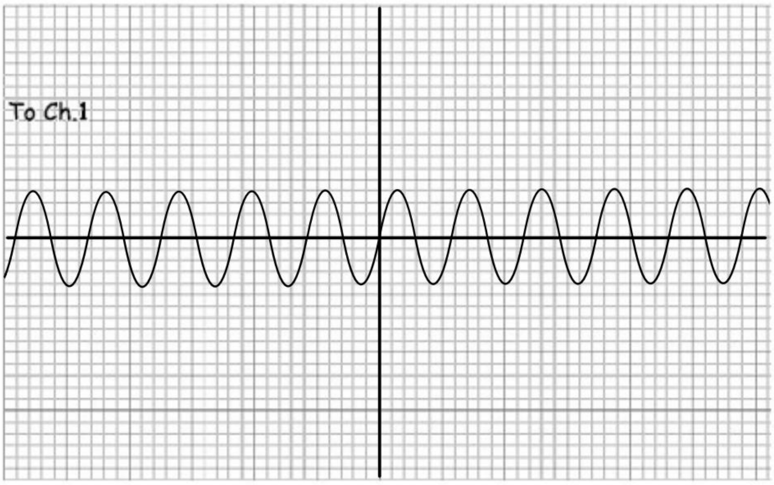
- Connecting the different modules of the Emona Telecoms-Trainer 101 kit to an oscilloscope to observe the signal.

Fig.1. Emona Telecoms-Trainer 101 kit

3 Results and Analysis

3.1 Part A: Introduction to the Emona Telecoms-Trainer 101

- Connect the master signal from the kit to the oscilloscope's Ch-1. Sketch Ch-1 output in the grid below (Figure 2), recording amplitude, period and frequency. [**1 mark**]



Amplitude

19.2 V

Period

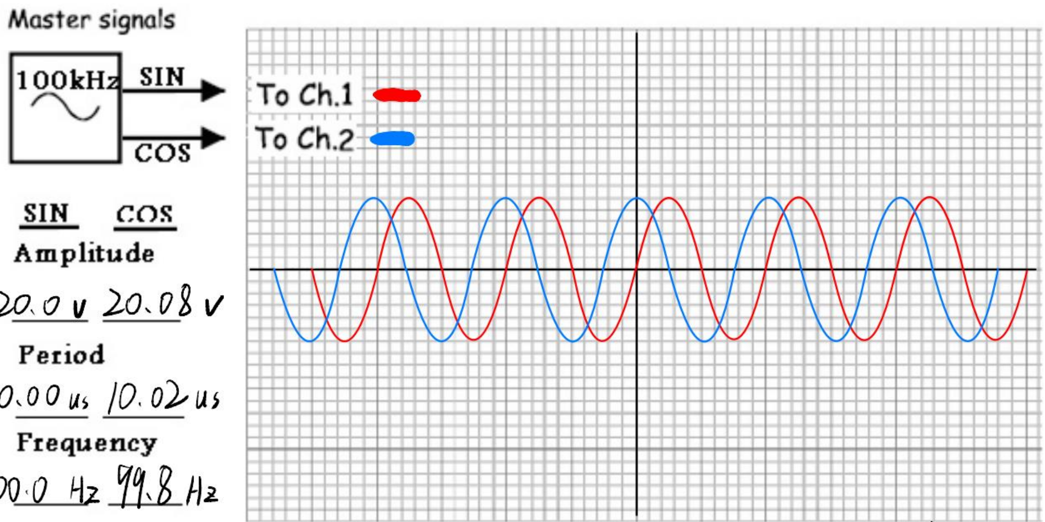
480.0 us

Frequency

2.083 kHz

Buffer module's gain =

7.24 ± 0.02



	SIN	**COS**
Amplitude	20.0 V	20.08 V
Period	10.00 us	10.02 us
Frequency	100.0 Hz	99.8 Hz

Phase difference = $\frac{\textbf{difference between two signals}}{\textbf{period}}$ **x 360** = $\frac{2.6}{10} \times 360 = 93.6$

- **Question 1: What do you expect the theoretical phase shift between SIN and COS to be, and why your measurement value does not match it?**

  Ans: The desired theoretical phase shift value is 90, but there is a slight deviation in phase due to the slightly different frequencies of the two signals.

- **Question 2: Can you sketch the signal? Why?**

Ans: No. The sound signal is too complex and disorganized. At the same time, the sound signal changes from time to time depending on the sound. Therefore, it is not possible to draw it.

Max Gain for input A = 2.08 ± 0.01 **[1 mark]**

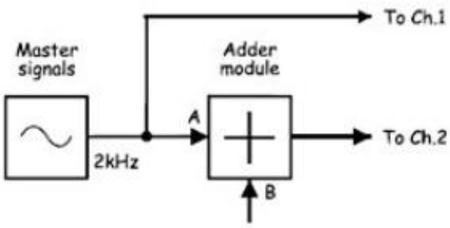
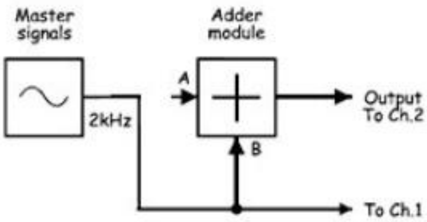


Figure 6: Adder example.

Max Gain for input B = 2.04 ± 0.01 **[1 mark]**



- **Question 3: Write down the equation that represent the output to Ch-2.**

  Ans: A + 0.5 B = output ch-2

- **Question 4: What is the maximum amplitude of the resulting signal (the signal of the summation)?**

  Ans: output = 19.6 + 20.4 = 40. The maximum amplitude of the resulting signal is 40 v.

- **Question 5: Is it right to say: Max amp of the summation output signal (A+B) = max amp of A + max amp of B? Why?**

  Ans: Yes, if there is no small phase difference, the sum of the A signal and the B signal is (A + B) = max map of A + max amp of B. However, there will be a certain phase difference between the different signals. This will result in the amplitude not reaching the desired value.

- **Question 6: What will happen if there is phase shift between the two added signals?**

  Ans: This results in a change in the maximum amplitude of the output signals after summing. Frequency will remain consistent.

3.2 Part B: Sampling and reconstruction

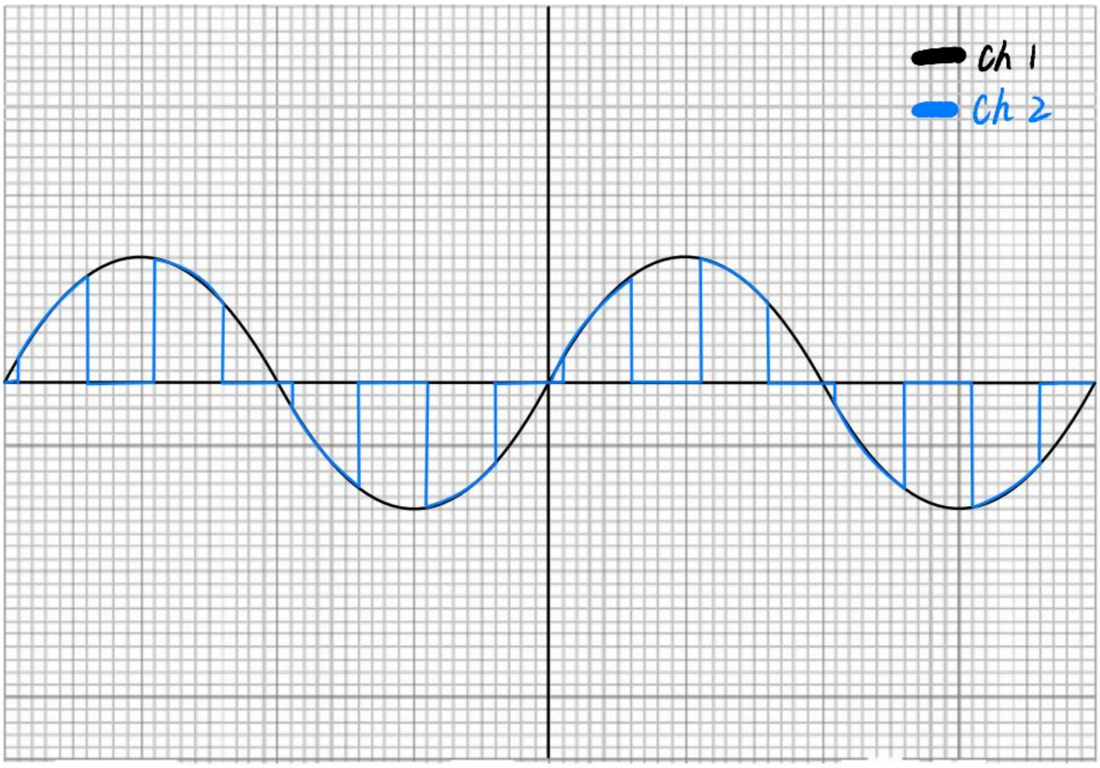


Figure 9: Natural sampling system.

- **Question 7: What is the name of this sampling type?**

  Ans: Natural sampling.

- **Question 8: Why it is called by that name?**

  Ans: The voltage will return to 0v between samples. the sampling voltage will change.

There are 8 samples in one cycle which would also be over-sampling.

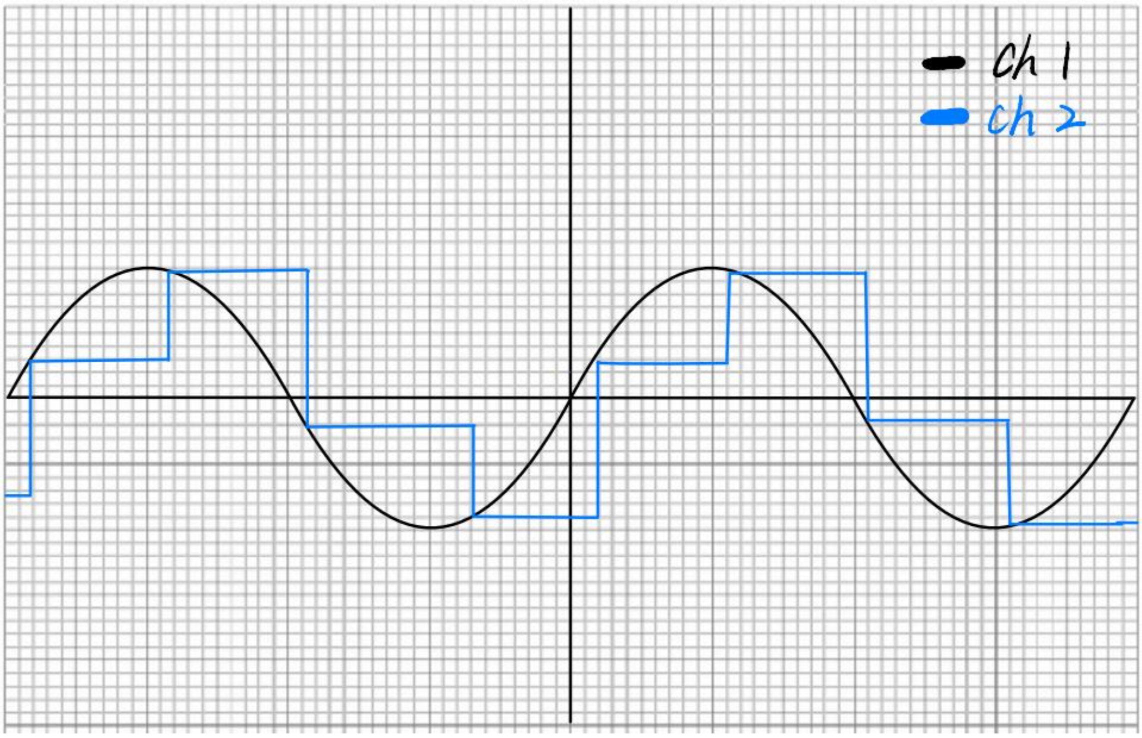


Figure 11: Sample-and-hold sampling system.

- **Question 9: What is the name of this sampling type?**

  Ans: Sample-and-hold sampling.

- **Question 10: Why it is called by that name?**

  Ans: The sampled voltage does not change between samples and there are 4 samples in one cycle, which is still over-sampling.

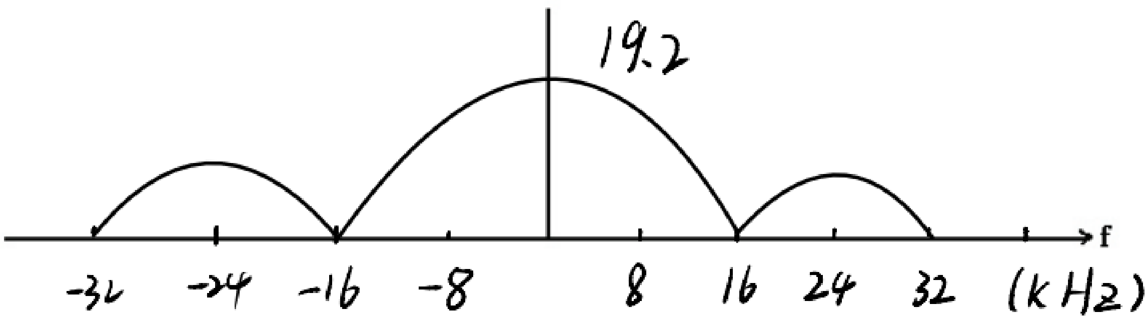
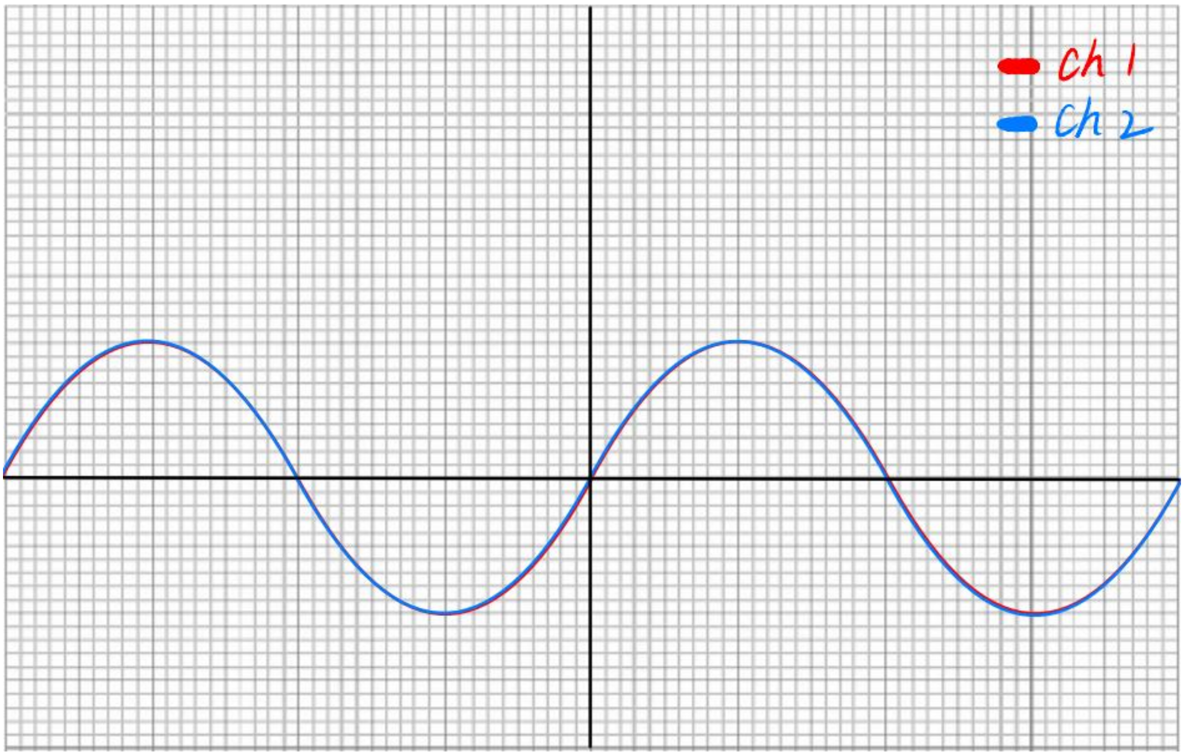


Figure 10: Frequency domain view.

- **Question 11: From your general knowledge, what is the theoretical frequency**

range for the speech signal?

Ans: Because it is a speech signal, this range approximates to the range of human hearing. This range is therefore: 20 Hz – 20 kHz



- **Question 12: What is the bandwidth of the Low-pass filter? Why?**

  Ans: 2.25 kHz. The sampled information consists of many sine wave signals of different frequencies. The purpose of the low-pass filter is therefore to avoid aliasing and to allow signals of the same frequency to pass through and reject other signals. As the original signal to be reconstructed is a sine wave, the frequency is approximately 2KHz.

- **Question 13: Measure the minimum sampling frequency without getting aliasing?**

  Ans: 4.8 kHz

- **Question 14: What is the minimum theoretical sampling frequency at which the message signal (2 kHz) can be reconstructed without distortion? Why?**

Ans: 4 k Hz. To avoid aliasing, the theoretical minimum frequency of the sampled signal is chosen to be twice the message frequency. The Nyquist sampling rate ensures that the frequency of the non-message sine wave of the sampled signal is higher than the message frequency.

- **Question 15: Why is the measured value larger than the theoretical value?**

  Ans: Filters have errors. Cut-offs are not instantaneous and their rejection of frequencies above the cut-off frequency is gradual rather than instantaneous. As a result, the sampled signal will be at a frequency slightly above the Nyquist sampling rate.

3.3 Part C: PCM encoding

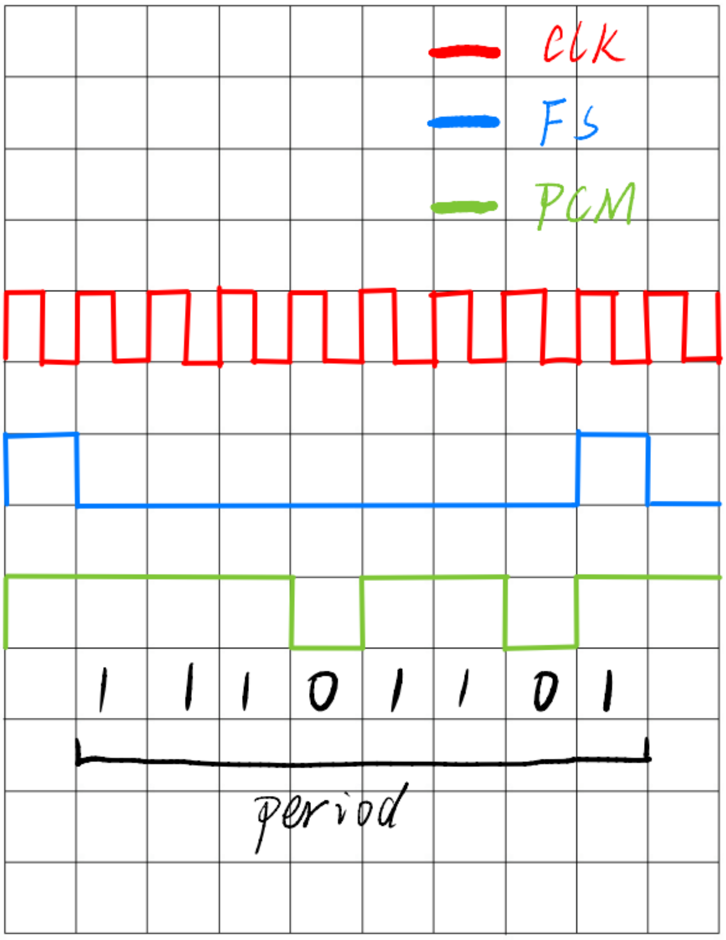


Figure 16: FS, PCM data and the clock signal.

- **Question 16: What is the binary number that the PCM Encoder module is generating?**

  Ans: 11101101

- **Question 17: Why does the code change even though the input voltage is fixed?**

  Ans: The PCM encoder module converts the analogue voltages between -2V and +2V into 8-bit binary numbers. There are therefore 256 different digits between 0000000000 and 11111111, which means that there are 256 quantization levels for each voltage. As 0V is in the middle of the voltage between -2V and +2V, the voltage will be unstable. Even if the input voltage is fixed, there will be very small voltage variations due to errors in the experimental instrumentation, which can lead to code variations.

- **Question 18: Why does the PCM Encoder module output this code for 0 V DC and not 00000000?**

  Ans: For coded voltage functions above 0V and below 0V. As it is the smallest 8-bit number, if 0V is made to indicate 0V DC, there is no number to indicate a negative voltage. Speech requires the encoder to encode voltages below 0V.

  What is the analogue voltage for 11111111 binary output? [**2 marks**]

  2.64 V (Experimental) 2.00 V (Theoretical)

  What is the analogue voltage for 00000000 binary output? [**2 marks**]

  -2.40 V (Experimental) -2.00 V (Theoretical)

- **Question 19: What is the maximum amplitude of the analogue signal that can be transmitted by this PCM system?**

Ans: 2.64V (experimental value) 2.00V (theoretical value) Because 2.64V (2.00V) is the analogue voltage of the 11111111 binary output, the maximum amplitude will be 2.64V.

- **Question 20: What is the resolution of this PCM encoder? Why?**

  Ans: The resolution is $\frac{2.64-(-2.4)}{256} \approx 0.02\ (experimental), \frac{2-(-2)}{256} \approx 0.015\ (theoretical)$. For 8 bits, the system will produce 256 different digits, which means that there will be 256 quantization levels between -2V and +2V. The resolution will therefore be the ratio of the voltage difference to 256.

- **Question 21: What is the name of the difference between a sampled voltage and its closest quantization level?**

  Ans: Quantisation error.

- **Question 22: How could you reduce this difference?**

  Ans: 1. By increasing the level, the error can be reduced because the binary number corresponding to each different voltage will be more accurate. 2. A low-pass filter can reduce the error by reducing ambient noise. 3. Using oversampling to reduce quantization error can be done by increasing the sampling frequency to use more samples.

- **Question 23: Why does the code on PCM Encoder module's output change even when you are not making a sound?**

  Ans: When the environment is quiet, the PCM encoder will generate a zero-amplitude signal of 0 volts. If the input is 0 volts, the output signal will also vary continuously. Furthermore, because the encoder is always sampling the input signal. Different input voltages cause the encoder to produce different numbers. Although I am not making a sound, there will be noise in the environment and this signal will be collected. Therefore, even though I am not making a sound, the output will change.

3.4 Part D: PCM decoding

- **Question 24: What must be done to the PCM Decoder module's output to reconstruct the message properly? Why?**

  Ans: A low-pass filter is used to remove the high-frequency component of the output to produce a smooth signal and avoid message signal aliasing. Also, alternate synchronisation bits are used to achieve frame synchronisation. The output will be correctly reconstructed after using the same clock signal.

- **Question 25: Why isn't the reconstructed message a perfect replica of the original message?**

  Ans: If the sampling frequency is less than twice the maximum frequency, then aliasing occurs. In aliasing, the two signals are mixed with noise. the PCM decoder uses a limited level of quantization to reconstruct the signal with some distortion, and it is not possible to perfectly reconstruct the signal with quantized noise.

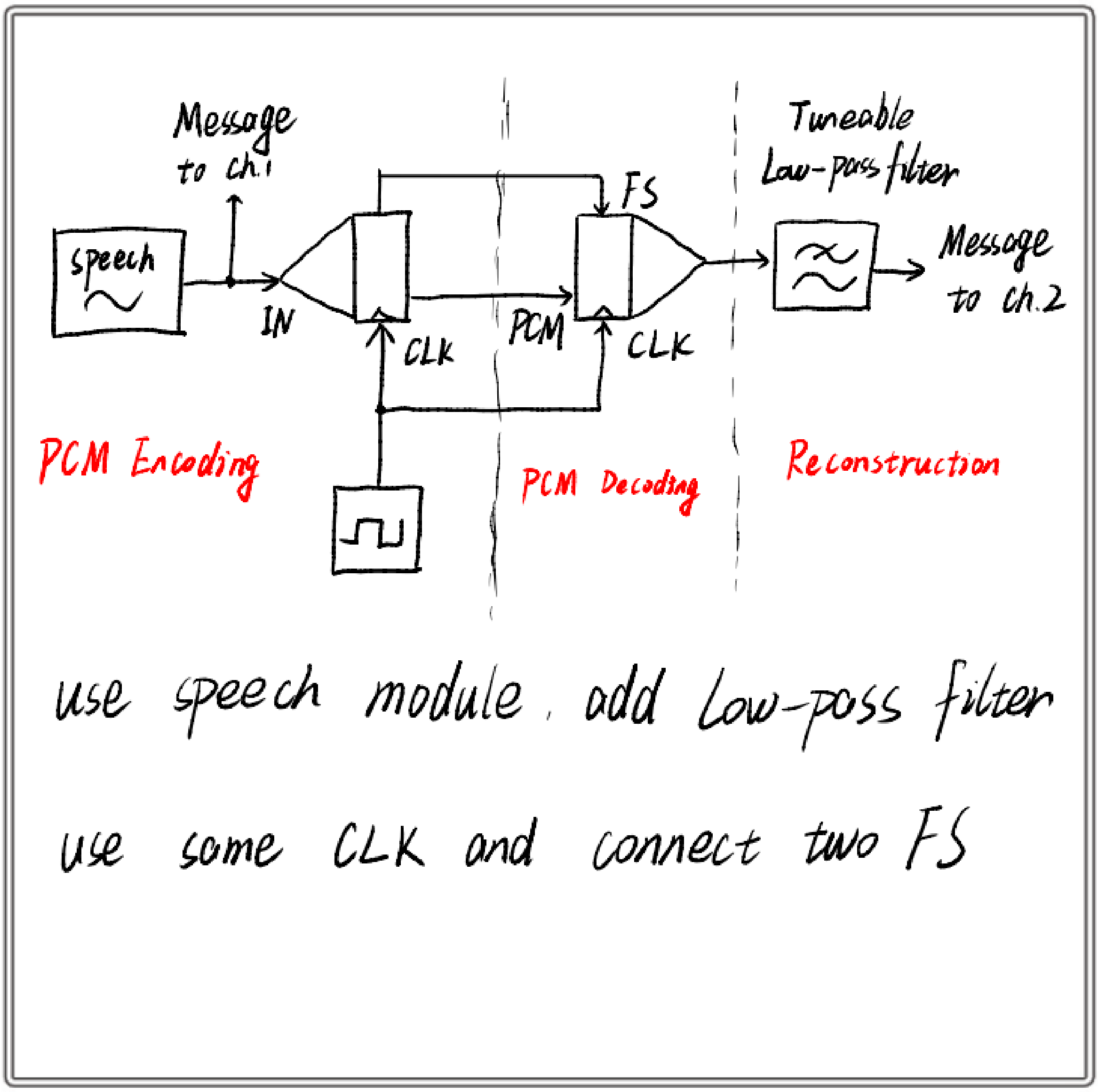


Figure 21: Speech encode-decoder system.

3.5 Part E: Bandwidth limitation and restoring digital signals

- **Question 26: Why does bandwidth limitation of the channel cause the PCM Decoder module to generate incorrect output?**

  Ans: If the bandwidth of the signal is reduced, the frequency range of the normally transmitted signal will also be reduced. If the bandwidth is reduced to less than the Nyquist frequency of the signal, part of the original signal will be lost. Frequency loss will cause signal distortion, and the oscilloscope will display incorrect output.

- **Question 27: If reducing the channel bandwidth is distorting the signal, how could this be compensated for?**

  Ans: In order to reduce distortion in signal transmission, the channel should have sufficient bandwidth. The use of low-pass filters reduces the effective harmonic distortion of the signal source. The use of a comparator to reduce distortion amplifies the difference between the input voltages. The voltage is fixed at one input as a reference and the difference between the reference voltage and the voltage at the output to the other end is amplified. By amplifying, the comparator compensates for distortion.

- **Question 28: What is the difference between the original and the restored signal?**

  Ans: The amplitudes are different and the input signal is lower than the output signal. And, there is a small delay between the output signal and the input signal.

- **Question 29: Why do some DC voltages cause the comparator to output wrong information?**

  Ans: The voltage level is used for comparison with the attenuated input signal. Therefore, the DC voltage needs to be set at the appropriate position between the maximum and minimum voltage. If the DC voltage is not set to the appropriate level it will cause the comparator to swing and output an incorrect message.

- **Question 30: Why does the comparator begin to output wrong information when this control is turned far enough?**

  Ans: When the DC voltage is set too high, the DC voltage level no longer intersects with the attenuated input signal, which causes the comparator output to be constantly high.

- **Question 31: How can the comparator restore the bandwidth limited digital signal when it is so distorted?**

  Ans: When a bandwidth-limited digital signal is distorted, some signal remains in the original signal and only signals with frequencies outside the bandwidth are lost, as different sine waves have different frequencies. If the bit rate is increased, there is enough information left over to reconstruct an accurate original signal using a comparator.

6 Conclusions

This experiment implements some basic signal sampling, as well as a PCM-encoded signal circuit. Various signals were observed and recorded through an oscilloscope. This gave me a good understanding of signal sampling and PCM coding. I also learnt to use the Emona Telecoms-Trainer 101 kit and became more proficient in the operation of the oscilloscope.

7 References

[1] García-Fernández, A., 2022. *Experiment 4 – Sampling and PCM.* [online] Canvas.liverpool.ac.uk. Available at: https://liverpool.instructure.com/courses/59021/pages/experiment-4?module_item_id=1543465