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Experiment 4 - Sampling and PCM

Department of Electrical Engineering & Electronics

September 2019, Ver. 3.4

Important: Marking of all coursework is anonymous. Please complete your details in the box shown above, then fold and glue the corner to keep your personal details anonymous during marking. Do not include your name, student ID number, group number, email or any other personal information anywhere else in this workbook. A penalty will be applied to submissions that do not meet this requirement.

Experiment specifications

Module(s)	ELEC224 / ELEC273
Experiment code	4
Semester	1
Level	2
Lab location	Electronics lab, third floor, check the lab timetable
Work	In groups
Timetabled time	7 hrs
Subject(s) of relevance	Sampling, Signals and Systems
Assessment method	This workbook, submitted at the end of the lab (one workbook must be submitted per student)
Submission deadline	Same lab day

Marks:

Part A-Introduction (15 Marks):

Part B-Sampling (25 Marks):

Part C-PCM Encoding (25 Marks):

Part D-PCM Decoding (10 Marks):

Part E-Bandwidth (15 Marks):

Pre-Lab test (10 Marks):

Total mark (100%):

Instructions:

- **Read this script carefully before attempting the experiment.**
- **The Pre-Lab Questions should be answered before the lab day. They are available on VITAL (Online) and worth 10%.**
- **The script questions should be answered while carrying out the experiment.**
- **At each part, you should show your connections and output to one of the demonstrators and get approval.**
- **This script should be completed with the graphs and answers of the questions.**
- **The completed script should be submitted at the end of the lab to the responsible demonstrator for marking.**
- **If you have any feedback on your laboratory experience today, please write it down on the last page of this script.**

1 Objectives

The objectives of this experiment are:

- to study and practically test some important telecommunications concepts like sampling, PCM encoding and decoding and bandwidth limitations.
- to get hands-on experience on using the Emona Telecoms-Trainer 101 kit.

2 Apparatus

- Emona Telecoms-Trainer 101 kit.
- Oscilloscope TDS 210.
- Different wires and BNC cables.

3 Introduction

In modern telecommunications, digital transmission has continually increased since its introduction in 1962. This is due, in large part, to the fact that most of the communication providers require a high degree of accuracy in the information they are transmitting through their networks. With digital transmission (as compared to analogue), systems are better switching interfaces, easier to multiplex and producing clearer signals. A digital signal is depicted as discontinuous discretely variable on/off pulses, as opposed to an analogue signal which is continuously variable.

Each pulse is known as a bit. A bit is the most common digital signal in the telecommunication industry. The number of bits transmitted per second is the bit rate of the signal. To convert analogue signals to digital, a coding system called Pulse Code Modulation (PCM) is used. This process requires other pre- and post-processing steps, among which sampling is needed.

This experiment introduces some of these concepts, offering hands-on experience and practical tests.

4 Part A: Introduction to the Emona Telecoms-Trainer 101 (15 Marks)

The Emona Telecoms-Trainer 101 is used to study the fundamentals of telecommunications principles at the block diagram level. It contains many telecommunications' functional building units that can be patched to implement a wide variety of systems like basic modulators (Figure 1), encoders and other important sub-systems associated with telecommunications theory, using only one piece of lab hardware and without worrying about how the internal circuit works. Examples of common blocks include adder, filter and phase shifter.

The kit offers the required components to practically implement a wide range of the basic analogue communications concepts like AM, FM, DSB, SSB, PM, PAM, TDM, PWM, PLL, QAM and SNR, and digital communications concepts like PCM, PCM-TDM, ASK, BPSK, FSK, GFSK, DPSK, QPSK, spread spectrum, line coding and noise generation.

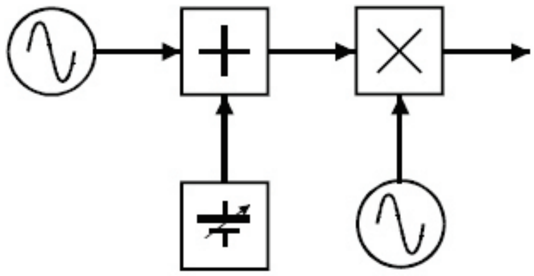


Figure 1: Example of a system built from component blocks.

4.1 The master signals and the buffer modules

- 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]**

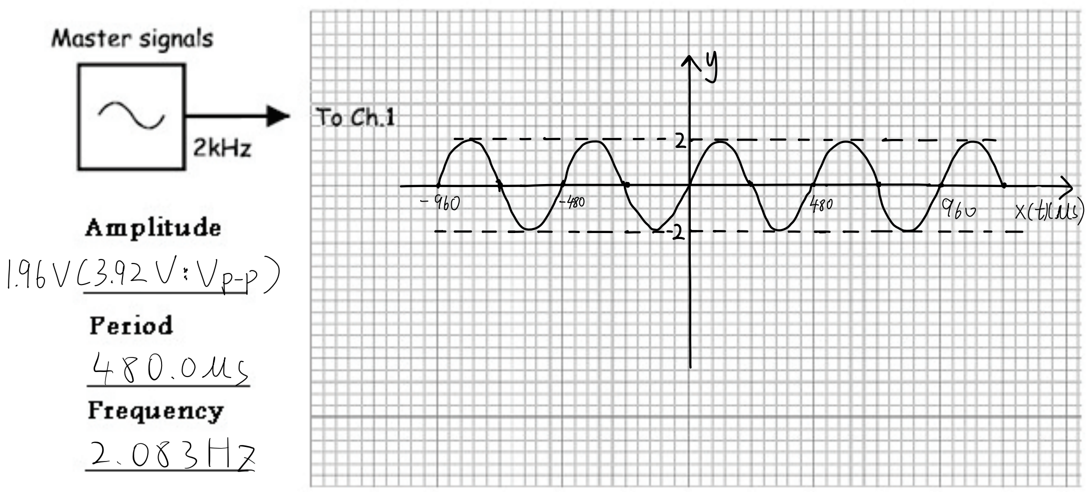


Figure 2: Master signal.

- Locate the Buffer module and connect the setup in Figure 3. Calculate the buffer's gain mentioning the uncertainty in this measurement. **[1 mark]**

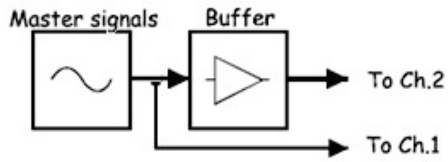


Figure 3: Master signal and buffer block.

Buffer module's gain =

7.24 ± 0.02 (Mean value : 7.24; uncertainty : 0.02)

- Connect the master signal 100 kHz SIN to Ch-1 and 100 kHz COS to Ch-2, as in Figure 4. Sketch both signals in the same grid of the figure, recording amplitude, period, frequency and phase shift, making note of the uncertainty in each of these measurements. **[1 mark]**

	sin	cos
– Amplitude:	1.96V (3.92V : Vpp)	1.96V (3.92V : Vp-p)
– Period:	10.04 μs	10.02 μs
– Frequency:	100.4 HZ	99.8 HZ
– Phase shift:	90° (Theoretical)	100.8° (Experimental)

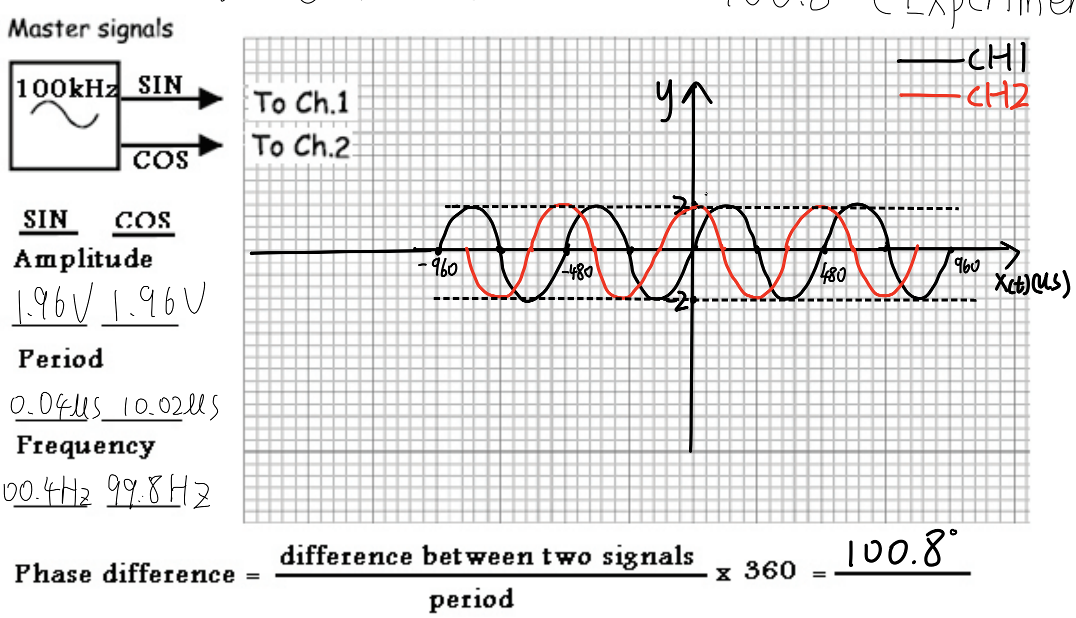


Figure 4: SIN and COS master signals.

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? **[2 marks]**

The specific explanation and answer will be placed at the end of this lab script. (Question part)

4.2 The Speech module

- Locate the speech module on the kit that contains a microphone (Figure 5). Connect its output to Ch-1.

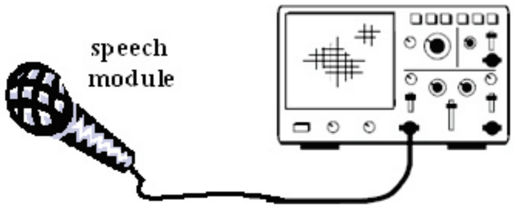


Figure 5: Speech module.

- Talk while observing the signal on the oscilloscope screen.

Question 2 [1 mark]

- Can you sketch the signal? .... Why? ....

The specific explanation and answer will be placed at the end of this lab script. (Question part)

4.3 The Adder module

The adder module has two inputs: A and B. Each input has its own variable gain key, labelled with G for input A and g for input B.

- Locate the adder module and connect the setup shown in Figure 6. Find out the maximum gain for input A, and then connect the setup in Figure 7. Find out the maximum gain for input B, noting the uncertainty in this measurement..

Max Gain for input A = 2.04 ± 0.01 ... **[1 mark]**
(Mean) (Uncertainty)

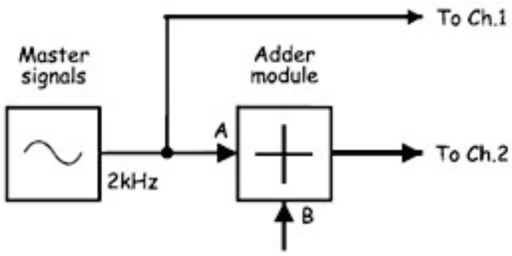


Figure 6: Adder example.

Max Gain for input B = 2.04 ± 0.01 ... **[1 mark]**
(Mean) (Uncertainty)

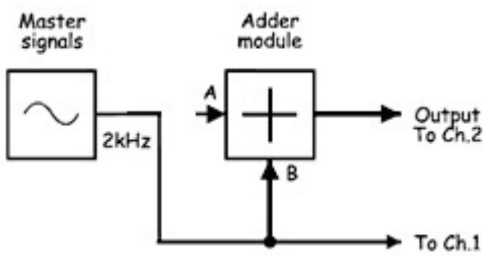


Figure 7: Adder example.

- Connect the setup in Figure 8 and adjust the gains of input A to 1 and input B to 0.5.

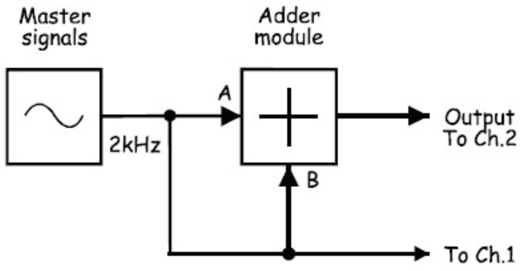


Figure 8: Adder example.

Question 3 [1 mark]

- Write down the equation that represent the output to Ch-2.

The specific explanation and answer will be placed at the end of this lab script. (Question part)

Question 4 [2 marks]

- What is the maximum amplitude of the resulting signal (the signal of the summation)?

The specific explanation and answer will be placed at the end of this lab script. (Question part)

Question 5 [2 marks]

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

The specific explanation and answer will be placed at the end of this lab script. (Question part)

Question 6 [2 marks]

- What will happen if there is phase shift between the two added signals?

The specific explanation and answer will be placed at the end of this lab script. (Question part)

5 Part B: Sampling and reconstruction (25 Marks)

Objectives:

In this part, sampling (by natural sampling and a sample-and-hold schemes) and reconstruction will be studied. The effect of aliasing will be examined as well.

Theory:

In communication systems like AM and FM, the instantaneous value of the message signal is used to change certain parameter of the carrier signal. Pulse modulation systems differ from these systems in a way that they transmit a limited number of discrete states of a signal at a predetermined time. Sampling can be defined as measuring the value of a message signal at predetermined time intervals. The rate of which the signal is sampled is known as the *sampling rate* or *sampling frequency*. It is the major parameter, which decides the quality of the reproduced signal. If the signal is sampled quite frequently (whose limit is specified by Nyquist Criterion) then it can be reproduced exactly at the receiver with no distortion. Sampling is the first step in digitising an analogue signal.

When the message signal is a simple sine wave, the sampled signal in this case consists of the following (why?):

- A sine wave at the same frequency as the message.
- A pair of sine waves that are the sum and difference of the fundamental and message frequencies.
- Many other pairs of sine waves that are the sum and difference of the sampling signals' harmonics and the message.

Nyquist Criterion: The lowest sampling frequency that can be used without the side bands overlapping is twice the highest frequency component present in the message signal. If we reduce this sampling frequency even further, the side bands and the message signal will overlap and the message signal can not be recovered simply by low pass filtering. This phenomenon is known as fold-over distortion or aliasing.

5.1 Sampling a simple message

- Locate the Dual Analogue Switch module and connect the setup shown in Figure 9 using one of the two inputs. It uses an electronically controlled switch to connect the message signal (the 2 kHz SINE from the Master Signals module) to the output. The switch is controlled (opened and closed) by the 8 kHz digital output of the Master Signals module.
- Draw the two signals of Ch-1 and Ch-2 to the same scale in the space provided below using different colours. [**2 marks**]

  Question 7 [**2 marks**]

  – What is the name of this sampling type?

  The specific explanation and answer will be placed at the end of this lab script. (Question part)

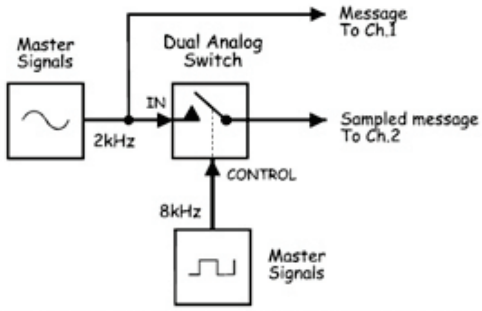
  Question 8 [**2 marks**]

  – Why it is called by that name?

  The specific explanation and answer will be placed at the end of this lab script. (Question part)

- In the space provided below in Figure 10, draw the frequency domain view of the sampled signal. [**1 mark**]
- Modify the setup as shown in Figure 11 below.

  The electronically controlled switch in the original setup has been substituted by a sample-and-hold circuit. The message and sampling signals remain the same (2 kHz sine wave and 8 kHz pulse train, respectively).



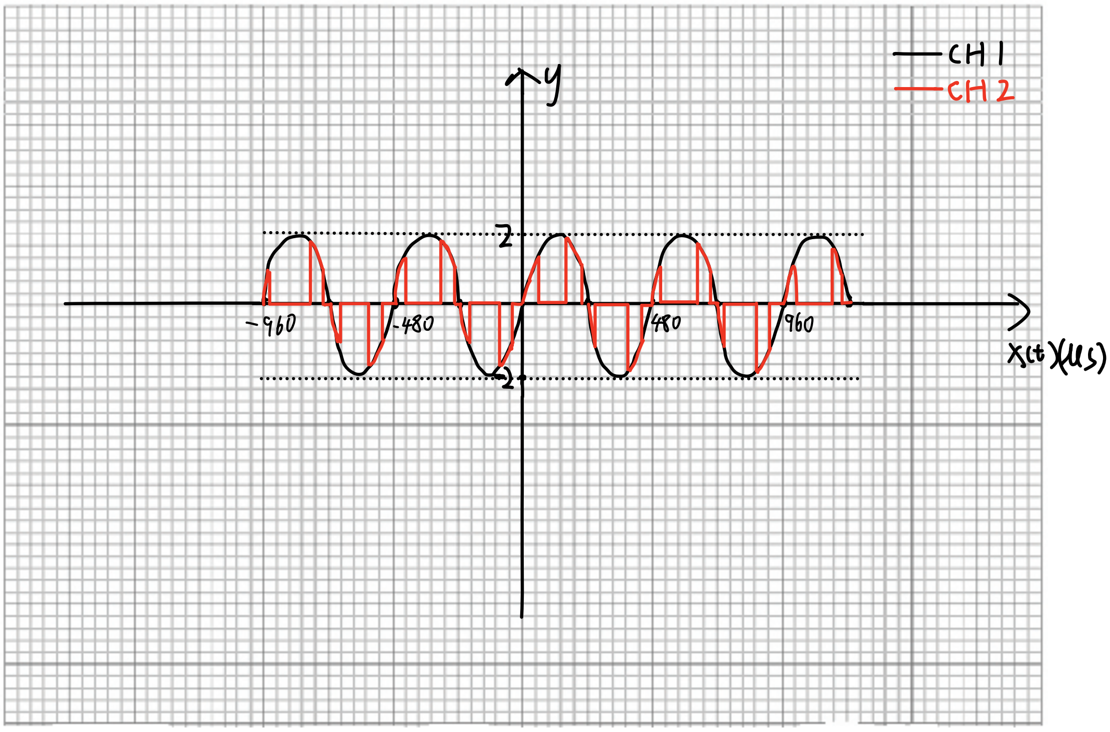


Figure 9: Natural sampling system.

- Draw the two signals of Ch-1 and Ch-2 to the same scale in the space provided below using different colours. [**2 marks**]

Question 9 [**2 marks**]

– What is the name of this sampling type?

The specific explanation and answer will be placed at the end of this lab script. (Question part)

Question 10 [**2 marks**]

– Why it is called by that name?

The specific explanation and answer will be placed at the end of this lab script. (Question part)

5.2 Speech sampling

- Sample the speech signal provided from the speech module by 8 kHz signal. Show your connections and results to one of the demonstrators.

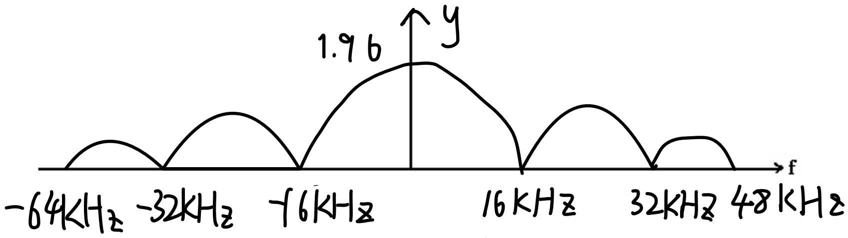


Figure 10: Frequency domain view.

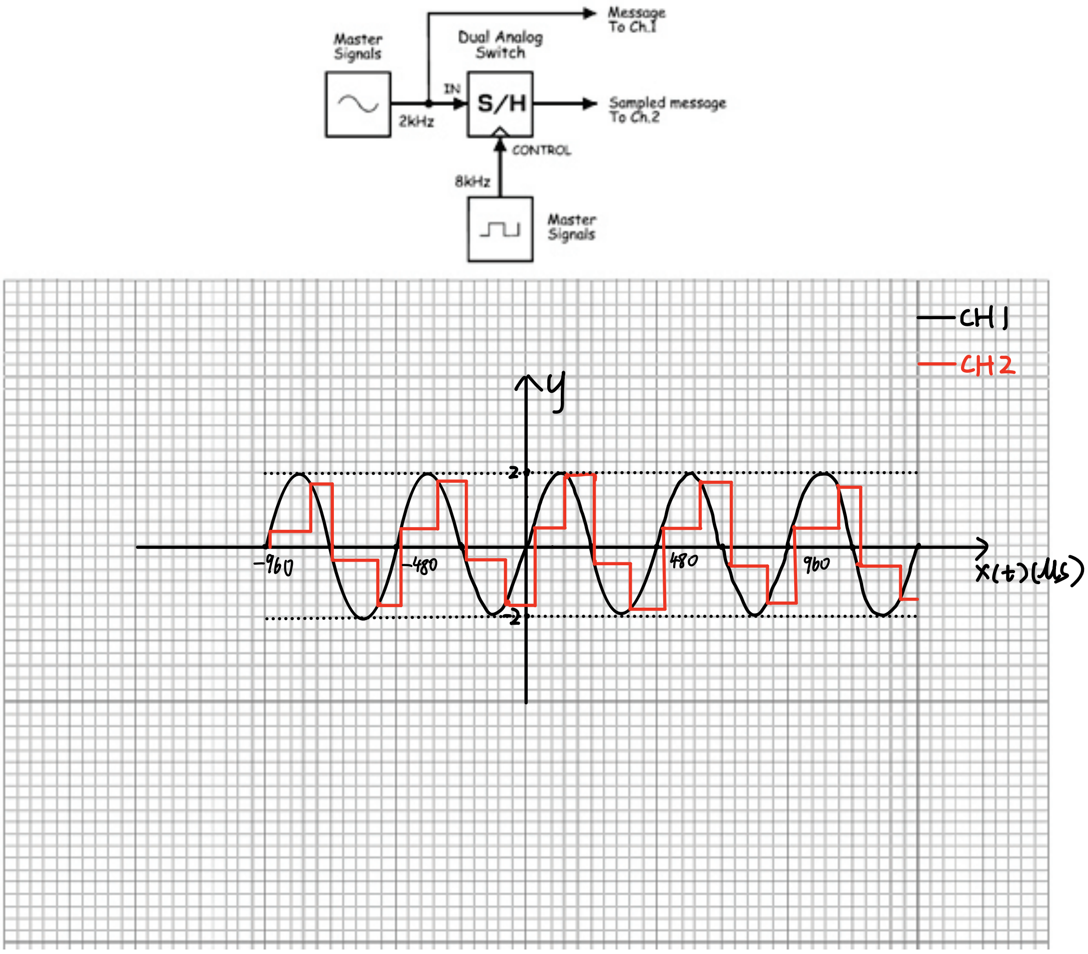


Figure 11: Sample-and-hold sampling system.

Question 11 [2 marks]

- From your general knowledge, what is the theoretical frequency range for the speech signal?

The specific explanation and answer will be placed at the end of this lab script. (Question part)

5.3 Reconstructing a sampled message

Theory:

Recall that the sampled message is made up of many sine waves. Importantly, for every sine wave in the message, there's a sine wave in the sampled message with the same frequency. So, reconstructing the original message involves passing the sampled message signal through a *low-pass filter*. This lets the sine wave(s) with the same frequency as the message pass through while rejecting other sine waves.

- Locate the low-pass filter and modify the setup as shown in Figure 12 below.

  The Tunable Low-pass Filter module is used to recover the message. The filter is tunable because the point at which frequencies are rejected (called the cut-off frequency) is adjustable. At this point and after, there should be nothing out of the Tunable Low-pass Filter module. This is because it has been set to reject almost all frequencies, even the message. However, the cutoff frequency can be increased by turning the module's Cut-off Frequency adjust control clockwise.

- Slowly turn the the Cut-off Frequency control clockwise and stop the moment the message signal has been reconstructed.

- Draw in the space provided in Figure 12 the signal from Ch-1 and Ch-2 overlapped and to the same scale with different colours. [**2 marks**]

Question 12 [2 marks]

- What is the bandwidth of the Low-pass filter? Why?

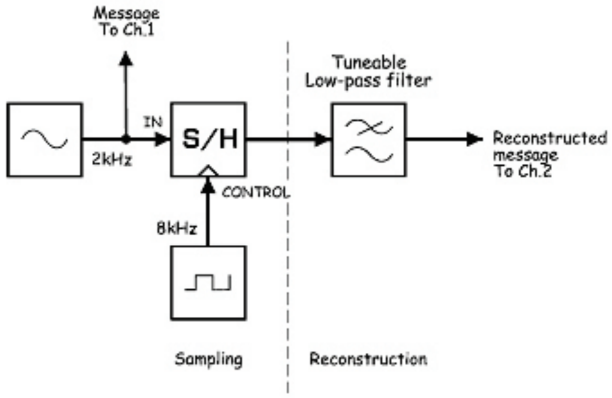
The specific explanation and answer will be placed at the end of this lab script. (Question part)

5.4 Aliasing

The filter is only letting the message signal pass through to the output and rejecting all other frequencies (or sine waves) that make up the sampled message. This is only possible because the frequency of these sine waves is high enough. Their frequency is set by the sampling rate (that is, the sampling signal's frequency).

Now, suppose the frequency of the sampling signal is lowered. You'd still get the message but the other sine waves would have a lower frequency. If the sampling signal's frequency is low enough, one or all of the other sine waves could pass through the filter along with the message. Obviously, this would distort the reconstructed message which is a known problem called **aliasing**.

To avoid aliasing, the sampling signal's theoretical minimum frequency is chosen to be twice the message frequency (or twice the highest frequency in the message if it contains more than one sine wave). This figure is known as the **Nyquist Sample Rate** and helps to ensure that the frequencies of the non-message sine waves in the sampled signal are higher than the message's frequency. Filters aren't perfect; their rejection of frequencies beyond the cut-off is gradual rather than instantaneous. So, in practice, the sampling signal's frequency needs to be a little higher than the Nyquist Sample Rate.



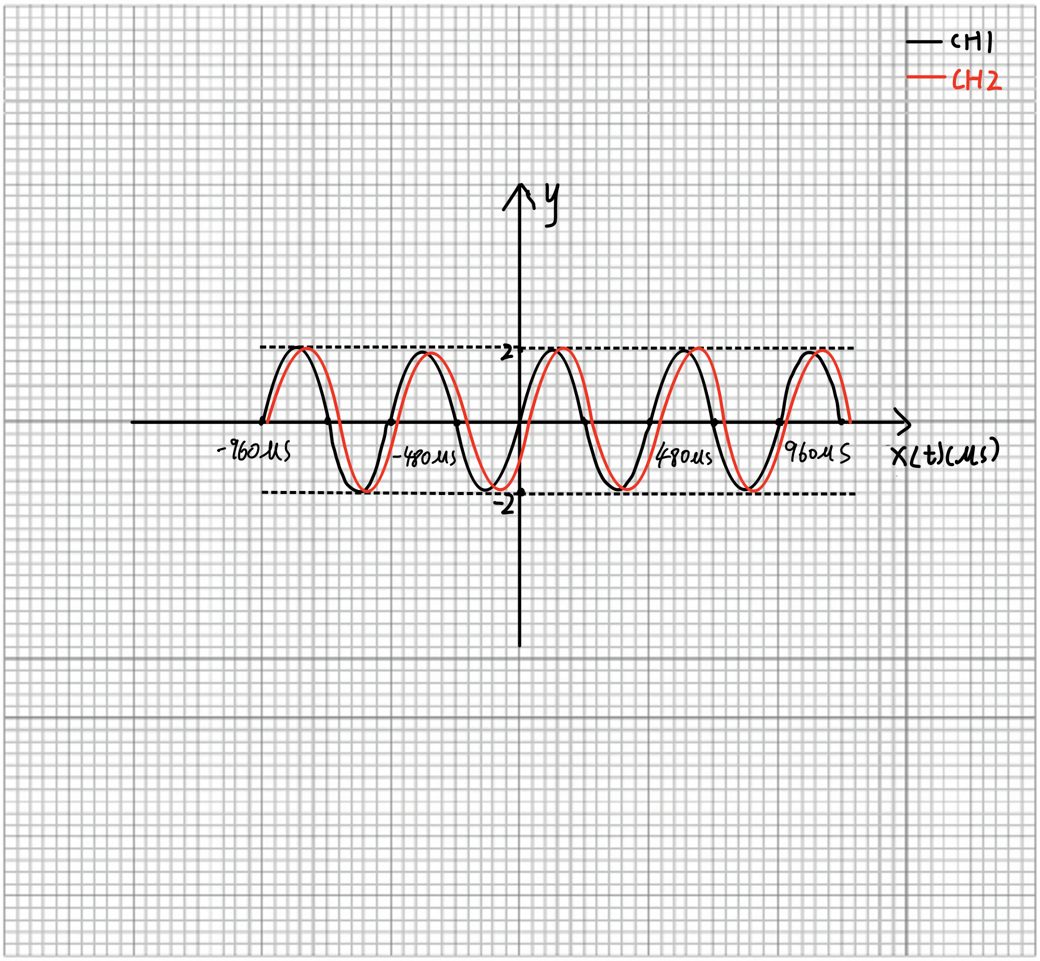


Figure 12: Reconstruction system.

- Locate the VCO module and modify the circuit in Figure 12 to get the circuit in Figure 13, that is to replace the fixed 8 kHz sampling signal by a variable frequency signal from the VCO.

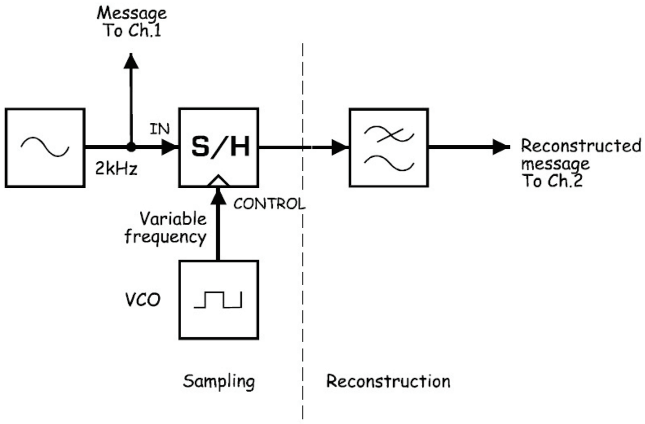


Figure 13: Reconstruction system-Aliasing.

- By starting from the maximum frequency that the VCO can generate (fully clockwise), reduce the frequency slowly while observing the reconstructed signal.

Question 13 [**2 marks**]

- Measure the minimum sampling frequency without getting aliasing?

The specific explanation and answer will be placed at the end of this lab script. (Question part)

Question 14 [**2 marks**]

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

The specific explanation and answer will be placed at the end of this lab script. (Question part)

Question 15 [**2 marks**]

- Why is the measured value larger than the theoretical value?

The specific explanation and answer will be placed at the end of this lab script. (Question part)

6 Part C: PCM encoding (25 Marks)

Digital transmission systems are steadily replacing analogue ones in commercial communications applications. This is especially true in telecommunications. Hence, an understanding of digital transmission systems is crucial for technical people these industries.

PCM is a system for converting analogue message signals to a serial stream of 0s and 1s. The conversion process is called *encoding*. In general, encoding involves:

- Sampling the analogue signal message at regular intervals using a sample-and-hold scheme.

- Comparing each sample to a set of reference voltages called *quantisation levels*.
- Deciding which quantisation level the sampled voltage is closest to.
- Generating the binary number for that quantisation level.
- Transmitting the binary number one bit at a time (that is, in serial form).
- Taking the next sample and repeating the process.

An factor that is crucial to the performance of the PCM system is the encoder's clock frequency. The clock tells the PCM encoder when to sample and, as the previous experiment shows, this must be at least twice the message frequency to avoid aliasing. Another important PCM performance factor is the difference between the sample voltage and the quantisation levels that it is compared to. To explain, most sampled voltages will not be the same as any of the quantisation levels. As mentioned above, the PCM Encoder assigns to the sample the quantisation level that is closest to it. However, the original sample value is lost and the difference is known as *quantisation error*.

The PCM Encoder module built on the kit uses a PCM encoding and decoding chip (called a codec) to convert analogue voltages between -2 V and +2 V to an 8-bit binary number (a word). With eight bits, it's possible to produce 256 different numbers between 00000000 and 11111111 inclusive. This in turn means that there are 256 quantisation levels (one for each number).

Each binary number is transmitted in frames serially. The most significant bit of each word (called bit-7) is sent first, bit-6 is sent next and so on to the least significant bit (bit-0). The PCM Encoder module also generates a separate Frame Synchronisation signal (FS) that goes high at the same time that bit-0 is transmitted. The FS signal has been included to help in PCM decoding, but it can also be used to help trigger another sub-system when looking at the signals that the PCM Encoder module generates.

Figure 14 below shows an example of three frames of a PCM Encoder module's output data (each bit could be either 0 or 1) together with its clock input and its FS output.

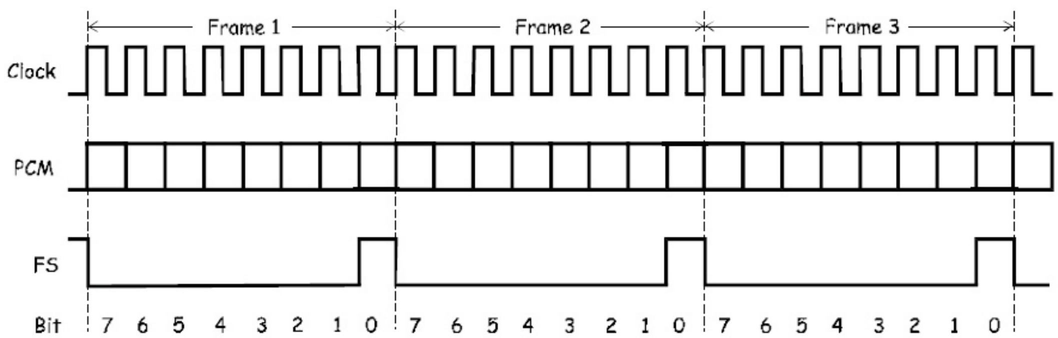


Figure 14: Three frames PCM data.

Objectives:

In this part, PCM Encoder module will be used to convert the following messages to PCM: a 0 V message, a DC voltage message and a continuously changing signal. In the process, PCM encoding will be verified and the corresponding quantisation error will be investigated.

6.1 Encoding a 0 V message

- Locate the PCM Encoder module and set its Mode switch to the PCM position. Connect the setup shown in Figure 15.

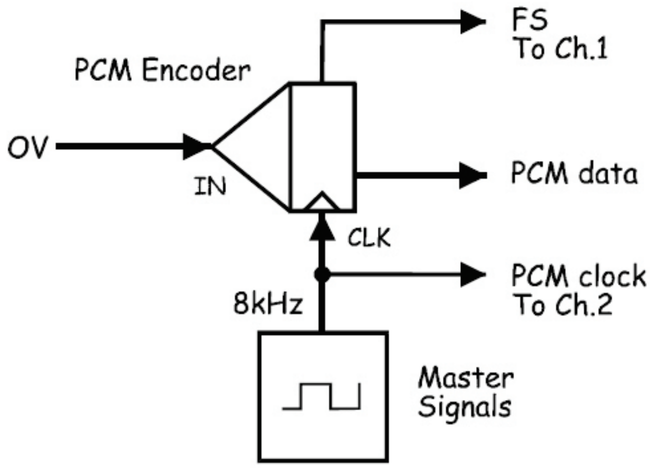


Figure 15: PCM Encoder-0 V message.

- Draw in the space provided of Figure 16 the FS, PCM data and the clock signals, dividing the graphs into three areas, one for each signal. Annotate in your drawings the start and the end of each bit and frame, and indicate bit-0 and bit-7 of each frame. [**3 marks**]

Question 16 [**2 marks**]

- What is the binary number that the PCM Encoder module is generating?

The specific explanation and answer will be placed at the end of this lab script. (Question part)

Question 17 [**2 marks**]

- Why does the code change even though the input voltage is fixed?

The specific explanation and answer will be placed at the end of this lab script. (Question part)

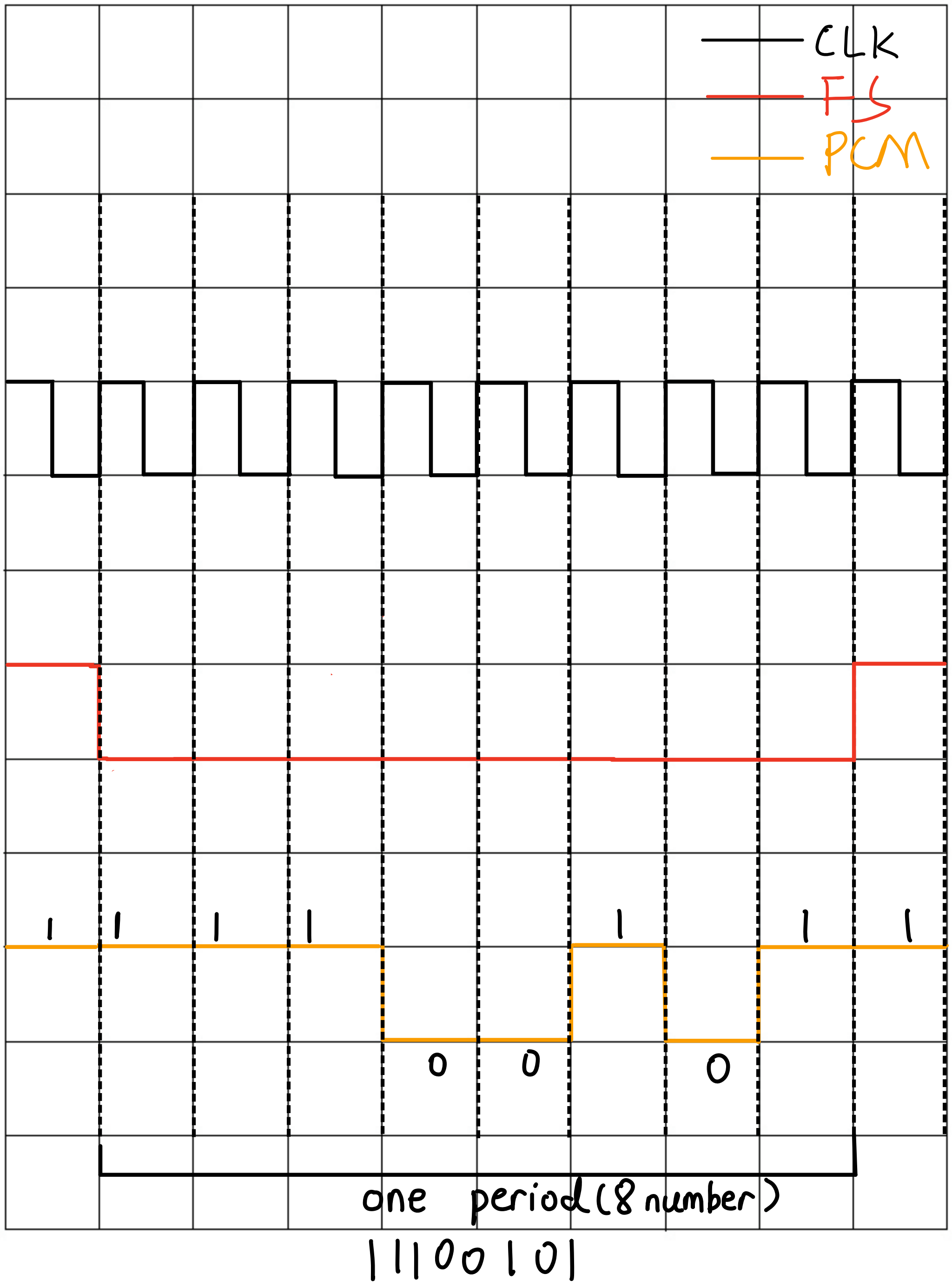


Figure 16: FS, PCM data and the clock signal.

Question 18 [2 marks]

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

The specific explanation and answer will be placed at the end of this lab script. (Question part)

6.2 Encoding a DC voltage

- Modify the setup as shown in Figure 17 using variable DC voltage instead of 0 V (GND) signal and keep changing the DC value from the minimum to the maximum and observe the PCM data output.

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

  2.64 V (Experimental Value), 2.00V (Theoretical Value)

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

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

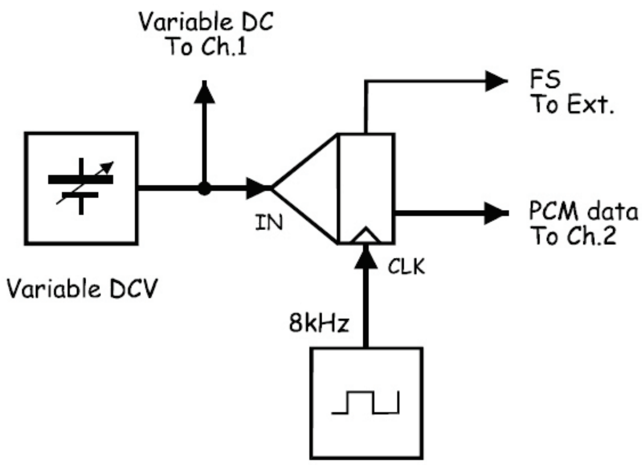


Figure 17: PCM encoder-variable DC voltage message.

Question 19 [2 marks]

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

The specific explanation and answer will be placed at the end of this lab script. (Question part)

Question 20 [2 marks]

- What is the resolution of this PCM encoder? Why?

The specific explanation and answer will be placed at the end of this lab script. (Question part)

Question 21 [2 marks]

- What's the name of the difference between a sampled voltage and its closest quantisation level?

The specific explanation and answer will be placed at the end of this lab script. (Question part)

Question 22 [3 marks]

- How could you reduce this difference?

The specific explanation and answer will be placed at the end of this lab script. (Question part)

6.3 Encoding continuously changing voltages

Let's see what happens when the PCM encoder is used to convert continuously changing signals like speech.

- Modify the setup as shown in Figure 18 below.

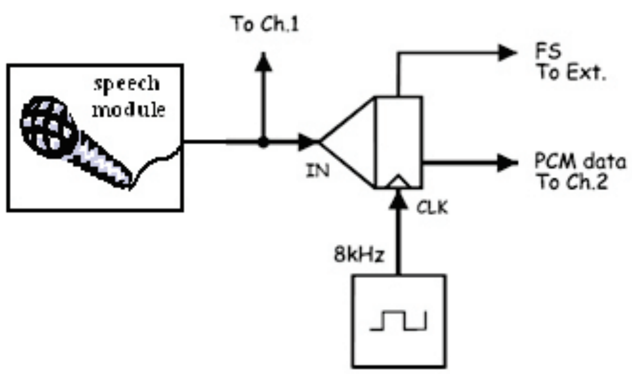


Figure 18: PCM encoder-Speech message.

Question 23 [3 marks]

- Why does the code on PCM Encoder module's output change even when you're not making a sound?

The specific explanation and answer will be placed at the end of this lab script. (Question part)

7 Part D: PCM decoding (10 Marks)

Theory:

The previous part introduced you to the basics of PCM which is a system for converting message signals to a continuous serial stream of binary numbers (encoding). Recovering the message from the serial stream of binary numbers is called *decoding*.
In general, decoding involves:

- Identifying each new frame in the data stream.
- Extracting the binary numbers from each frame.
- Generating a voltage that is proportional to the binary number.
- Holding the voltage on the output until the next frame has been decoded (forming a pulse amplitude modulation (PAM) version of the original message signal).
- Reconstructing the message by passing the PAM signal through a low-pass filter.

Objectives:

In this part, sine wave and speech messages will be converted to a PCM data stream then to a PAM signal using the PCM Decoder module. For this to work correctly, the decoder's clock and frame synchronisation signals are simply taken (stolen) from the PCM Encoder module. The message will be recovered then using the Tunable Low-pass Filter module.

7.1 Decoding the PCM data

- Connect the setup shown in Figure 19 below.

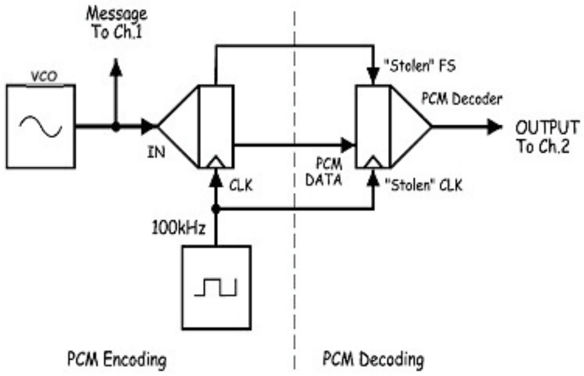


Figure 19: PCM decoder.

Question 24 [2 marks]

– What must be done to the PCM Decoder module's output to reconstruct the message properly? Why?

The specific explanation and answer will be placed at the end of this lab script. (Question part)

- Modify the setup as shown in Figure 20 below.

Slowly turn the Tunable Low-pass Filter module's Cut-off Frequency control clockwise and stop the moment the message signal has been reconstructed (ignoring the phase shift).

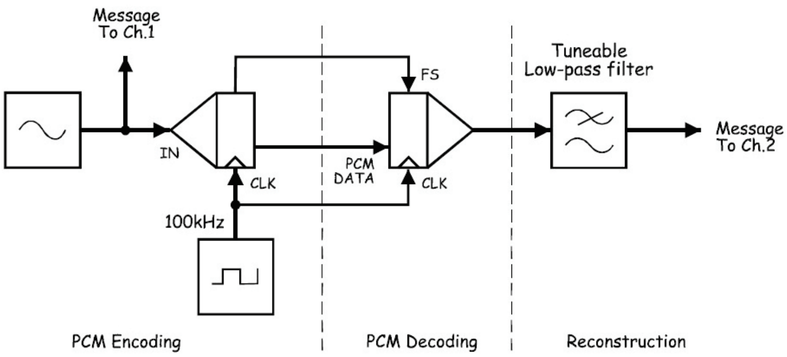


Figure 20: Modified PCM decoder.

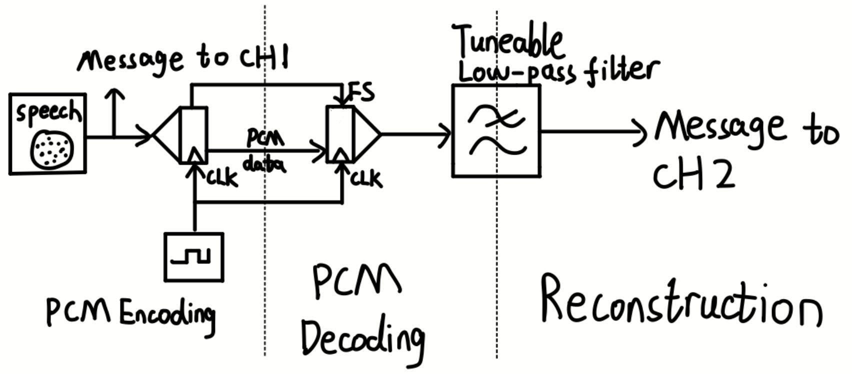
Question 25 [2 marks]

- Why isn't the reconstructed message a perfect replica of the original message?

The specific explanation and answer will be placed at the end of this lab script. (Question part)

7.2 Encoding and decoding of speech [6 marks]

- Draw in the space provided a diagram to encode and decode the speech signal, and connect your suggested setup. Show your system and output to one of the demonstrators.
- Add proper module to make the reconstructed (decoded) signal more like the original message signal.



Explanation: ① Use speech moudle

② Add Low-pass filter

③ Use the same CLK and Connect two FS together

Figure 21: Speech encode-decoder system.

8 Part E: Bandwidth limitation and restoring digital signals (15 Marks)

Theory:

In the classical communications model, a useful message moves from a transmitter to a receiver over a channel. A number of transmission media can be used as channels including: metal conductors (such as twisted-pair or coaxial cable), optical fibre and free-space. Regardless of the medium used, all channels have a certain bandwidth. That is, the medium lets a range of frequencies pass relatively unaffected while frequencies outside the range are made smaller (or attenuated). In this way, the channel acts like a filter, an issue that in fact has important implications. Recall that the modulated signal in analogue and digital modulation schemes consists of many sine waves. If the medium's bandwidth isn't wide enough then some of the sine waves are attenuated and others may be completely lost. In both cases, this causes the demodulated signal (the recovered message) to be a non-faithful reproduction of the original version. Making the matter even worse, the channel is like a filter in that it shifts the phase of the sine waves by different amounts (why?). Imagine the difficulty a digital receiver circuit such as a PCM decoder would have trying to interpret the logic levels of a received signal that has been tackled by the above factors. In this case, codes will be misinterpreted and incorrect voltages are generated. This makes the recovered message noisy which is obviously a problem.

Objectives:

In this part, a PCM communication system will be setup. Then, bandwidth limitation of the channel will be modelled by introducing a low-pass filter. The effect of bandwidth limitation on the PCM data will be observed using an oscilloscope as well as listening to the effect it has on the recovered voice message. Finally, a comparator will be used to recover a digital signal and observe its limitations.

8.1 The effects of bandwidth limitation on PCM decoding

Bandwidth limitation in a channel can distort digital signals and upset the operation of the receiver. This part of the experiment demonstrates this using a PCM transmission system.

- Connect the setup shown by the block diagram in Figure 22 below. The Tunable Low-pass Filter module models the bandwidth limitation of the channel.

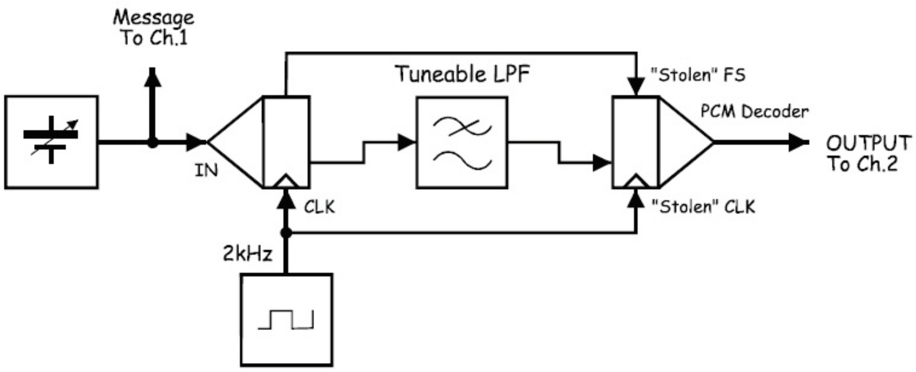


Figure 22: Channel bandwidth limitation model.

- Turn the Variable DCV module's VDC control left and right. At the same time, slowly turn the Tunable Low-pass Filter module's Cut-off Frequency Adjust control anti-clockwise and stop turning it once the PCM Decoder module's output becomes corrupted.

Question 26 [2 marks]

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

The specific explanation and answer will be placed at the end of this lab script. (Question part)

8.2 The effects of bandwidth limitation on a digital signal shape

Use the Sequence Generator module (32-bit) to model a digital data signal.

- Locate the Sequence Generator module and set its dip switches to 00. Connect the setup shown in Figure 23 below. Use SYNC output to trigger the oscilloscope to provide a stable display.

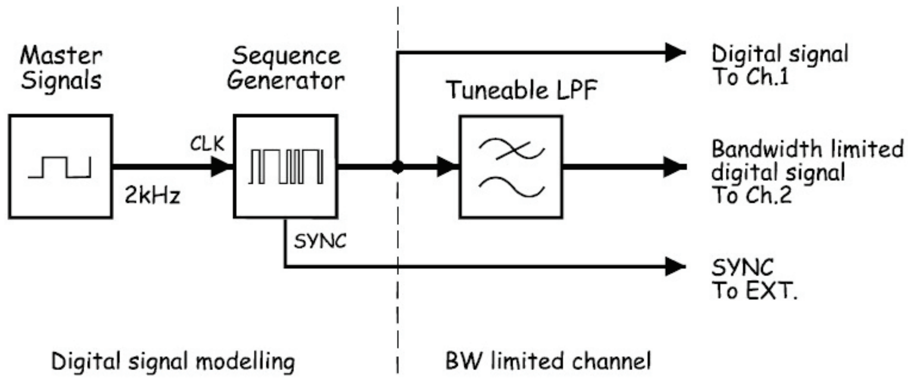


Figure 23: Bandwidth limitation on a digital signal.

- Investigate the effect of the Tunable LPF (the effect of the channel) by making the channel's bandwidth narrower by turning the Tunable Low-pass Filter module's Cut-off Frequency Adjust control anti-clockwise.

Question 27 [2 marks]

– If reducing the channel bandwidth is distorting the signal, how could this be compensated for?

The specific explanation and answer will be placed at the end of this lab script. (Question part)

An obvious solution to the problem of channel bandwidth limitation is to use a transmission medium that has a sufficiently wide bandwidth for the digital data. In principle, this is a good idea yet could be impractical or not possible always. Certain types of cable design have better bandwidths than others. However, as digital technology spreads, there are demands to push more data through the existing channels. To do so without slowing things down requires that the transmission bit rate be increased. This ends up having the same basic effect as reducing the channel's bandwidth. The next part of the experiment illustrates this.

- Modify the setup as shown in Figure 24 below.

  Turn the Tunable Low-pass Filter module's Cut-off Frequency Adjust control to get the maximum possible bandwidth. To model increasing the transmission bit-rate, continue turning the VCO module's Frequency Adjust control clockwise while observing the oscilloscope's display. Show your connection and output to one of the demonstrators.

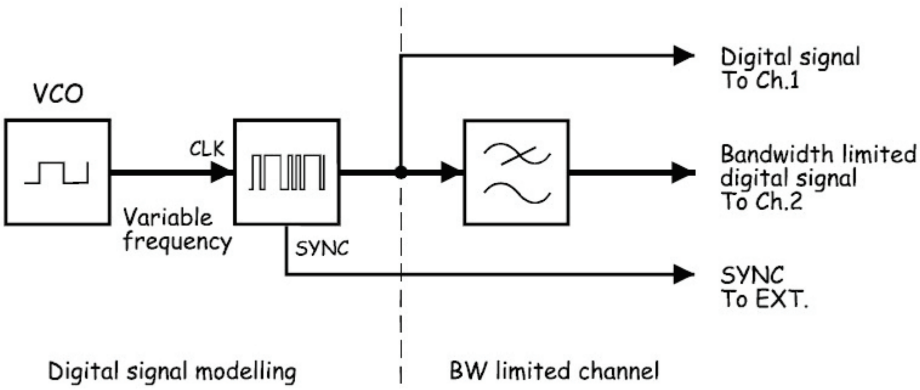


Figure 24: Studying bit-rate effect.

8.3 Restoring digital signals

As it was seen before, bandwidth limitation distorts digital signals. In fact, bandwidth limitation is almost inevitable and its effects get worse as the transmission bit-rate increases.

To manage this problem, the received digital signal must be cleaned-up or restored before it is decoded. A device that is ideal for this purpose is the *comparator*. The comparator amplifies the difference between the voltages on its two inputs by large amount. This always produces a heavily clipped or squared-up version of any AC signal connected to one input if it swings above and below a DC voltage on the other input (reference voltage).

This part of the experiment lets you restore a bandwidth limited digital signal using a comparator.

- Modify the setup as shown in Figure 25.

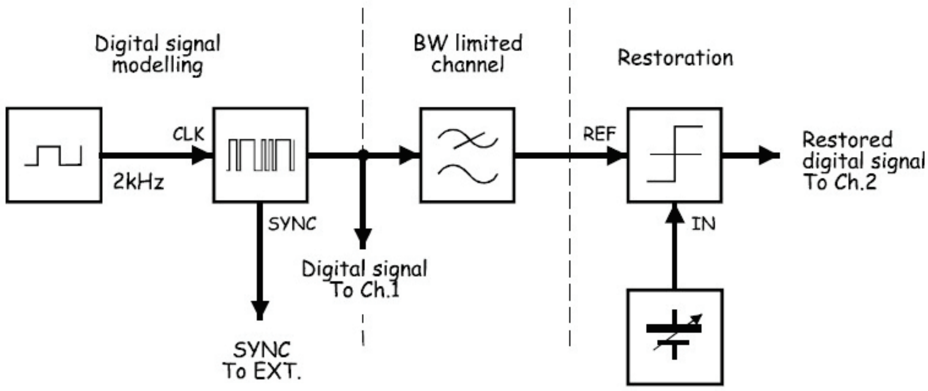


Figure 25: Restoring digital signals using a comparator.

Question 28 [2 marks]

- What is the difference between the original and the restored signal?

The specific explanation and answer will be placed at the end of this lab script.
(Question part)

Slowly turn the Variable DC module's DC Voltage control to fully clockwise and fully anti-clockwise positions and observe the effect.

Question 29 [3 marks]

- Why do some DC voltages cause the comparator to output wrong information?

The specific explanation and answer will be placed at the end of this lab script.
(Question part)

Return the Variable DCV module's Variable DC control to about the middle of its range. Slowly make the channel's bandwidth narrower by turning the Tunable Low-pass Filter module's Cut-off Frequency Adjust control anti-clockwise.

Question 30 [3 marks]

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

The specific explanation and answer will be placed at the end of this lab script. (Question part)

Question 31 [3 marks]

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

The specific explanation and answer will be placed at the end of this lab script. (Question part)

9 Assessment and Marking Scheme

This experiment is assessed by means of a workbook. Don't forget to submit this workbook to the responsible demonstrator before leaving the lab (one workbook must be submitted per student). The marking scheme for this workbook is as follows:

- Results of Part A: **15 Marks**
- Results of Part B: **25 Marks**
- Results of Part C: **25 Marks**
- Results of Part D: **10 Marks**
- Results of Part E: **15 Marks**
- The pre-lab test: **10 Marks**

10 Plagiarism and Collusion

Plagiarism and collusion or fabrication of data is always treated seriously, and action appropriate to the circumstances is always taken. The procedure followed by the University in all cases where plagiarism, collusion or fabrication is suspected is detailed in the University's Policy for Dealing with Plagiarism, Collusion and Fabrication of Data, Code of Practice on Assessment, Category C, available on `https://www.liverpool.ac.uk/media/livacuk/tqsd/code-of-practice-on-assessment/appendix_L_cop_assess.pdf`.

Follow the following guidelines to avoid any problems:

(1) Do your work yourself.

(2) Acknowledge all your sources.

(3) Present your results as they are.

(4) Restrict access to your work.

References

[1] Emona Telecoms-Trainer 101 kit manual, 2008.

Version history

Name	Date	Version
Dr M López-Benítez	September 2019	Ver. 3.4
Dr A Al-Ataby	August 2014	Ver. 3.3
Dr A Al-Ataby	October 2013	Ver. 3.2
Dr A Al-Ataby and Dr W Al-Nuaimy	October 2012	Ver. 3.1
Dr A Al-Ataby and Dr W Al-Nuaimy	October 2011	Ver. 3.0
Dr Nael Al-Zubi	October 2009	Ver. 2.0
Dr Nael Al-Zubi	August 2008	Ver. 1.0

Feedback:

If you have any feedback on your laboratory experience for this experiment (e.g. timing, difficulty, clarity of script, demonstration ...etc) and suggestions to how the experiment may be improved in the future, please write them down in the space below. This feedback is important for future versions of this script and to enhance the laboratory process, and will not be assessed. If you wish to provide this feedback anonymously, you may do so by detaching this page and submitting it to the Student Support Centre (fifth floor office).

Question part

Question 1 [2 marks]
What do you expect the theoretical phase shift between SIN and COS to be and why your measurement value does not match it?

The theoretical phase shift between SIN and Cos will be $\frac{\pi}{2} = 90^{\circ}$

And the measurement value is $\frac{\pi}{2} = 100.8^{\circ}$

The reasons for that will be :

1.Not all sine waveforms will pass the zero axis point accurately, and may be "offset" to a value on the right or left of 0.

2.Not all cosine waveforms are symmetrical about the y axis, and the center may be "offset" to the right or left of y axis.

3.Errors caused by experimental measurement. It may be that the instability of the connecting lines.

4.Systematic errors

Question 2 [1 mark]
Can you sketch the signal? Why?

No.

I can not sketch the signal.

The reason is that the speech module will collect signals from the **environment**. In the experiment, we use an oscilloscope to show the collected signal. However, the signals in the environment are **constantly changing** and there is **not regular** when changing. It is difficult to sketch accurate signals, because signals **cannot be represented by any mathematical expressions.**

Question 3 [1 mark]
Write down the equation that represent the output to Ch-2.

The output to Ch-2 will be:

$$\text{Output} = G*A+g*B$$

If G=1 and g=0.5，the equation will be simplified to:

$$Output = 1*A+0.5*B$$

Question 4 [2 marks]
What is the maximum amplitude of the resulting signal (the signal of the summation)?

If the input A is 1 and input B is 0.5, the maximum amplitude will be :

$$Output = G*A+g*B=2.04*1+2.04*0.5=3.06V$$

If the input A is 1.96(Experimental maximum value) and input B is 1.96(Experimental maximum value), the maximum amplitude will be :

$$Output = G*A+g*B=2.04*1.96+2.04*1.96 \approx 8.00V$$

Question 5 [2 marks]
Is it right to say: Max amp of the summation output signal (A+B) = max amp of A + max amp of B? Why?

Yes

Because the two added signals, which have the same frequencies, **do not have a phase shift**. The original inputs are both generated by master signals and the amplitudes are also both max. In the above questions, the Max Gains for input A and B are approximately equal. If there is no phase difference, the max amplitude of the summation will be equal to the addition of two max amplitude.

Question 6 [2 marks]
What will happen if there is phase shift between the two added signals?

First, the **frequencies** will not change and the **phase and the added results** (Amplitude) will change.

Secondly, the above equation **(Max amp of the summation output signal (A+B) = max amp of A + max amp of B)** will be not available. The reason is that it can not be known whether input signal A and signal B are both positive or negative. If two input signals are a positive number and a negative number, it will cause a calculation error. In the case of **phase shift**, there will always be two signals with one is positive and another one is negative. So this calculation method will no longer be applicable.

Question 7 [2 marks]

What is the name of this sampling type?

Natural sampling and Over-sampling

Question 8 [2 marks]
Why it is called by that name?

1. The signal voltage returns to 0 volts between samples.

2.The sampling voltage will change during the sampling process.

3.There are 8 samples in one period which means that f_N=2*2=4$KHz < f_s$, therefore, the sampling is also the Over-sampling.

Question 9 [2 marks]
What is the name of this sampling type?

Sample-and-hold sampling and Over-sampling

Question 10 [2 marks]
Why it is called by that name?

1.The sampling voltage does not change during the sampling process.

2.There is no space between each sample.

3.There are 4 samples in one period which means that f_N=2*2=4$KHz < f_s$, therefore, the sampling is also the Over-sampling.

Question 11 [2 marks]
From your general knowledge, what is the theoretical frequency range for the speech signal?

Since the laboratory is relatively closed, external interference signals can be ignored. Speech signal is mostly produced by **human voice**, so the theoretical frequency range for speech signal is approximate to the frequency of human voice. When applied to speech, the human voice ranges from approximately 125Hz to 8kHz [1]. Therefore, the frequency range will be from approximately 125Hz to 8kHz. If the noise in the environment is also considered, the frequency range will be changed to 20-20KHz.

Question 12 [2 marks]
What is the bandwidth of Low-pass filter? Why?

2.25KHz (Experimental value)

The sampled message is composed of many sine-wave signals of different frequencies. Therefore, the purpose of low-pass filter is to avoid aliasing and allow signals of the same frequency to pass and reject other signals. Since the original signal to be reconstructed is sine-wave and the frequency is about 2KHz, the bandwidth of the filter will be close to 2KHz.

Question 13 [2 marks]
Measure the minimum sampling frequency without getting aliasing?

4.80KHz

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

4KHz(2*2KHz)

To avoid aliasing, the theoretical minimum frequency of the sampled signal is selected to be **twice the frequency of the message** (if multiple sine waves are included, the highest frequency in the message will be selected) [2]. **Nyquist** Sample Rate helps to ensure that the frequency of the non-message sine-wave in the sampled signal is higher than the message frequency. In this experiment, the frequency of the original signal used is 2 KHz. Therefore, **the minimum theoretical sampling frequency will be 4KHz.**

Question 15 [2 marks]
Why is the measured value larger than the theoretical value?

The filter is not perfect. The cut-off is not instantaneous which means their suppression of frequencies beyond the cut-off frequency is gradual rather than instantaneous. Therefore, the frequency of the sampled signal (4.80KHz) needs to be slightly higher than the Nyquist sampling rate(4KHz).

Question 16 [2 marks]
What is the binary number that the PCM Encoder module is generating?

11100101 (Experimental Value)

Question 17 [2 marks]
Why does the code change even though the input voltage is fixed?

Because PCM Encoder module will convert analogue voltages between -2V and +2V to an 8-bit binary number. Therefore, there are 256 different numbers between 00000000 and 11111111 inclusive, which means **256 quantisation levels** for each voltage. Because 0V is in the middle of the voltages between -2V and +2V, the voltage will be **not stable**. Even if the input voltage is fixed, there will also be very small voltage changes due to the error of the experimental instrument, which will lead code change.

Also, there will be **quantisation error** when comparing between sample levels and quantisation level. Therefore, the sampled voltages will not be the same as any quantisation levels, which means that each quantisation level has a range of sample voltages.

Question 18 [2 marks]
Why does the PCM Encoder module output this code for 0 V DC and not 00000000?

In order to realize the function of encode voltage above and **below 0V**. Since it is the smallest 8-bit number, if making 0V represent 0V DC, then there is no number to represent negative voltage. And encoding voltages below 0V are very common. Both Speech and music need encoder to encode voltages above and below 0V. The theoretical voltage of -2V will represent the 8-bit number: 00000000.

Question 19 [2 marks]
What is the maximum amplitude of the analogue signal that can be transmitted by this PCM system?

2.64V(Experimental Value) 2.00V(Theoretical Value)

Because 2.64V(2.00V) is the analogue voltage for 11111111 binary output, the maximum amplitude will be 2.64V (2.00V), which can be transmitted by this PCM system.

Question 20 [2 marks]
What is the resolution of this PCM encoder? Why?

The Experimental resolution will be: $\frac{2.64-(-2.40)}{256} \approx 0.020$

The Theoretical resolution will be: $\frac{2.00-(-2.00)}{256} \approx 0.015$

With 8-bits, the system will produce 256 different numbers, which means that there will be 256 quantisation levels between -2V and +2V. Therefore, the resolution will be the ratio of voltage differences to 256.

In addition, because the experimental maximum and minimum output are 2.64V and -2.40V, which are different from the theoretical values, the voltage difference will also be different. Therefore, the experimental resolution will be little different from the theoretical resolution.

Question 21 [2 marks]
What's the name of the difference between a sampled voltage and its closest quantisation level?

Quantisation Error

Question 22 [3 marks]
How could you reduce this difference?

First, there are **256** quantisation levels in this experiment. The number can be **increased to 512**, which means there will be 512 different numbers between -2V and +2V and 512 quantisation levels. Through increasing the levels, the error will be reduced because the binary number corresponding to each different voltage will be more accurate.

Secondly, the **Low-pass filter** can be used to reduce the error through reducing the noise from environment.

Thirdly, **improve the accuracy** of PCM encoder. By finding the binary number corresponding to each voltage more accurately, the error can be reduced.

Fourthly, the **low-sensitivity and low-noise filter** can also be used to optimize the amplitude response and reduce the error.

Fifthly, use **over-sampling** to reduce quantisation error, which means more samples can be used by increasing the frequencies of sampling.

Question 23 [3 marks]
Why does the code on PCM Encoder module's output change even when you're not making a sound?

When there is no sound in the environment, the PCM Encoder will generate a zero amplitude signal, which is 0 Volt. Through the above experiment, we can know that if the input is 0 Volt, then the output signal will not be 00000000. Based on the above explanation, when the input voltage is 0V, the output signal will also change continuously.

Additionally, because the encoder samples the input signal all the time. Different input voltages cause the encoder to produce different numbers. Although I did not make a sound, there will be noisy in the environment which are irregular and changing all the time, this kind of signal will also be collected. Therefore the output will change even I am not making a sound.

Question 24 [2 marks]

What must be done to the PCM Decoder module's output to reconstruct the message properly? Why?

1. **Use low-pass filter.** The output should remove the sharp edges (High frequency components) to generate a smooth signal. Therefore, the low-pass filter will avoid aliasing of the message signal.

2.The frame **synchronization** should be achieved using the alternating synchronization bit. After using the same clock signal, the output will reconstruct properly.

3.To make sure that the **switch and frequency is in the low position** and the **gain is around in the middle point**. The frequency and the gain need to be adjusted to proper position.

4.Make sure that **cut-off frequency** is high around **10KHz**, the **input signal is lower than that value**. To reconstruct the message properly, the cut-off frequency need to be larger than the input signal.

Question 25 [2 marks]

Why isn't the reconstructed message a perfect replica of the original message?

If the sampling frequency is less than twice the maximum frequency, **aliasing** will occur. In **aliasing**, the two signals become **indistinguishable** and mix with each other and mix with noise.

The PCM decoder reconstructs an apparent copy of the signal using finite quantization levels. The distortion: **Quantization noise** makes it impossible to reconstruct a signal perfectly.

Question 26 [2 marks]

Why does bandwidth limitation of the channel cause the PCM decoder module to generate incorrect output?

When the bandwidth of the signal is reduced, the **frequency range** of the signal that can be transmitted normally will also be reduced. Part of the original signal will be **lost** due to the bandwidth is reduced less than the Nyquist frequency of the signal when the bandwidth limitation of the channel is set. Loss of frequency will cause the signal to be distorted. Since the signal is distorted, the oscilloscope will display incorrect output. Additionally, as the bandwidth of a signal is decreased, the **quantization error** will increase.

Question 27 [2 marks]
If reducing the channel bandwidth is distorting the signal, how could this be compensated for?

First, **a transmission medium** can be used to reduce the distortion. Transmission medium has enough bandwidth for the original signal. However, without slowing down the transmission speed, the transmission bit rate need to be improved. Therefore, this method could be impractical.

Secondly, **Comparator** can be used to reduce the distortion. Comparator is an amplifier which can amplify the difference between the input voltages. When using a comparator, the voltage is fixed at one input terminal as the reference terminal, and the difference between the reference voltage and the voltage output to the other terminal will be amplified. Through the amplification, the comparator could compensate the distortion.

Thirdly, we can use **Linear modulation techniques**, including all kinds of quadrature-amplitude modulation (QAM) and phase-shift-keying (PSK) and use less bandwidth than nonlinear techniques, including various forms of frequency/minimum-shift-keying (FSK and MSK) [3].

Question 28 [2 marks]
What is the difference between the original and the restored signal?

First, the **amplitude** are different, the input signal is lower than the output signal.

Secondly, there is slight **time delay** between the output signal and input signal.

Thirdly, the restored signal is not displayed on the **negative half-axis** of the y-axis, which will be similar to the original signal shifted to the positive direction of the y-axis

Question 29 [3 marks]
Why do some DC voltages cause the comparator to output wrong information?

Changing the DC voltage will also change voltage level which will be used to compare with the attenuated input signal. Therefore, the DC voltage is necessary to be set to an **appropriate position** between the maximum and minimum voltages. If the DC voltage does not set to an appropriate level, it will lead to the comparator to **swing**, which means the voltage will change from high to low value and output wrong information.

Question 30 [3 marks]
Why does the computer begin to output wrong information when this control is turned far enough?

When the DC voltage is set too high or too low, the DC voltage level will no longer intersects with the the attenuated input signal, which will lead the comparator ouput either constant high or low. Therefore, the control need to be turned to an appropriate position rather than far enough to avoid the error.

Question 31 [3 marks]
How can the comparator restore the bandwidth limited digital signal when it is so distorted?

When the bandwidth limited signal becomes distorted and barely recognizable, some of the remaining information from the original signal will still exist. Additionally, only the sinewave with frequencies which are outside the bandwidth will be lost, the remaining information will be reserved because the different sine-waves have different frequencies. The remaining information is **enough** to reconstruct the accurate original signal using the comparator. When using a comparator, the voltage will be fixed at one input terminal as the reference terminal, and the difference between the reference voltage and the voltage output to the other terminal will be amplified.

References：

[1] Accusonusblog, “The Human Voice and the Frequency Range”, https://blog.accusonus.com/pro-audio-production/human-voice-frequency-range/, 2021

[2] Department of Electrical Engineering & Electronics, “Experiment 4 - Sampling and PCM”, https://liverpool.instructure.com/courses/46000/pages/experiment-4?module_item_id=1258989, 2019, Ver. 3.4

[3] ScienceDirect, “Linear Modulation”, https://www.sciencedirect.com/topics/computer-science/linear-modulation, 2021