**Question 1:**

**What is the binary number that the PCM Encoder module is outputting?**

Input Signal: The current value of the analog signal being encoded.

Quantization Levels: The number of levels into which the input signal range is divided. This determines the bit depth of the PCM. For example, if there are 256 levels (8 bits), then each sample is represented by an 8-bit binary number.

Sampling Rate: How frequently the analog signal is sampled.

Range and Scale of the Input: The range (minimum and maximum values) of the input signal and how it's scaled into the quantization levels.

In a PCM system, the analog input signal is sampled at regular intervals (defined by the sampling rate). Each sampled value is then quantized to the nearest level and converted into a binary number.

For instance, if you have a PCM system with a bit depth of 8 bits (256 levels), and the current sample of the input signal is quantized to the 100th level, then the binary output for this sample would be the binary equivalent of the decimal number 100, which is 01100100.

Without specific details about the input signal and the PCM system configuration, it's not possible to determine the exact binary number being output.

**Question 2: What’s the name for the difference between a sampled voltage and its closest quantisation level?**

The difference between a sampled voltage and its closest quantization level in the context of PCM (Pulse Code Modulation) is known as "quantization error" or "quantization noise". This error occurs because the process of quantization involves approximating the actual amplitude of the signal to the nearest available quantization level. Consequently, there's almost always a small difference between the original signal's amplitude and the quantized value, which is referred to as the quantization error. This error is an inherent part of the quantization process in digital signal processing.

**Question 3: It’s possible to work out how far apart a PCM encoder’s quantisation levels are, by using the information you’ve gathered so far. Given that the PCM Encoder convert analogue voltages between -2V and +2V to an 8-bit binary number. Calculate the difference between the quantisation levels.**

The difference between the quantization levels of a PCM encoder that converts analog voltages between -2V and +2V into an 8-bit binary number is 0.015625 volts or 15.625 millivolts. This value represents the smallest change in voltage that can be distinguished by this 8-bit PCM system.

**Question 4: What is the maximum quantisation noise in this case?**

The maximum quantization noise in this case is 0.0078125 volts or 7.8125 millivolts. This value represents the largest possible error between the actual analog signal value and its nearest quantized value in this 8-bit PCM system.

**Question 5: To reduce quantisation error is it better to have fewer or more quantisation levels between ±2V? How do you change the number of quantisation levels?**

Increase Bit Depth: The number of quantization levels is determined by the bit depth of the PCM system. It's calculated as 2 bit depth. For instance, an 8-bit system has

2^8=256 levels. To increase the number of levels, you need to increase the bit depth. For example, changing from an 8-bit to a 10-bit system increases the levels from 256 to 1024.

Modify the ADC: In practical terms, this is usually done by using an Analog-to-Digital Converter (ADC) with a higher bit depth. For example, replacing an 8-bit ADC with a 16-bit ADC in your PCM encoder would significantly increase the number of quantization levels.

**Question 6: What does the PCM Decoder’s ‘stepped’ output tell you about the type of signal that it is?**

The 'stepped' output of the PCM Decoder indicates that the signal it produces is a digital representation of the original analog signal. This digital signal closely resembles the original message, but it is not exactly the same due to the quantization process inherent in PCM encoding. In PCM, the continuous analog signal is sampled and each sample is quantized to the nearest level within a finite set of quantization levels. The output of the PCM Decoder, therefore, consists of discrete steps that approximate the continuously varying amplitude of the original signal​​.

**Question 7: What must be done to the PCM Decoder module’s output in order to reconstruct the message appropriately?**

To appropriately reconstruct the message from the PCM Decoder module's output, a low-pass filter is used. This process involves the following steps:

Locate the Tuneable Low-pass Filter module and set its Gain control to about the middle of its travel.

Turn the Tuneable Low-pass Filter module’s Cut-off Frequency Adjust control fully anti-clockwise.

Disconnect the plugs to the Speech module’s output.

Modify the setup as shown in the provided figure (Figure 12 in the document).

Slowly turn the Tuneable Low-pass Filter module’s Cut-off Frequency control clockwise and stop the moment the message signal has been reconstructed, ignoring any phase shift.

This procedure utilizes the low-pass filter to smooth out the stepped output of the PCM Decoder module, effectively reconstructing the original message signal from the PCM Decoder module's Pulse Amplitude Modulation (PAM) output​​.

**Question 8 Even though the two signals look and sound the same, why isn’t the reconstructed message a perfect copy of the original message?**

Quantization Error: During the quantization process in PCM encoding, the continuous range of the analog signal is approximated to a finite number of levels. This approximation introduces a quantization error as the exact amplitude of the original signal cannot be perfectly matched.

Sampling Limitations: The original analog signal is sampled at discrete intervals. The Nyquist Theorem dictates that the sampling frequency should be at least twice the highest frequency of the input signal to accurately reconstruct it. However, any frequency components of the signal above half the sampling rate (the Nyquist frequency) cannot be accurately captured (this leads to aliasing).

Filtering Effects: The use of a low-pass filter in the reconstruction process can introduce phase shifts and may not perfectly pass all frequency components of the signal, especially near the cutoff frequency. This can alter the signal slightly.

Bandwidth Limitations: The bandwidth of the system may not fully encompass the entire spectrum of the original signal, leading to the loss of certain frequency components.

Noise and Distortion: System noise and distortion, albeit minimal in a well-designed system, can also contribute to the difference between the original and reconstructed signals.