Telecommunications
Communications Technologies

Digital Modulation (PCM / DPCM / Delta)

Courseware Sample

39863-F0

Order no.: 39863-10
First Edition
Revision level: 01/2015

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Printed in Canada

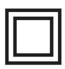
ISBN 978-2-89640-287-8 (Printed version)
Legal Deposit – Bibliothèque et Archives nationales du Québec, 2008
Legal Deposit – Library and Archives Canada, 2008

Safety and Common Symbols

The following safety and common symbols may be used in this manual and on the equipment:

Symbol	Description
⚠ DANGER	**DANGER** indicates a hazard with a high level of risk which, if not avoided, will result in death or serious injury.
⚠ WARNING	**WARNING** indicates a hazard with a medium level of risk which, if not avoided, could result in death or serious injury.
⚠ CAUTION	**CAUTION** indicates a hazard with a low level of risk which, if not avoided, could result in minor or moderate injury.
CAUTION	**CAUTION** used without the *Caution, risk of danger* sign ⚠, indicates a hazard with a potentially hazardous situation which, if not avoided, may result in property damage.
	Caution, risk of electric shock
	Caution, hot surface
	Caution, risk of danger
	Caution, lifting hazard
	Caution, hand entanglement hazard
	Notice, non-ionizing radiation
⎓	Direct current
∼	Alternating current
≂	Both direct and alternating current
3∼	Three-phase alternating current
⏚	Earth (ground) terminal

Safety and Common Symbols

Symbol	Description
	Protective conductor terminal
	Frame or chassis terminal
	Equipotentiality
	On (supply)
	Off (supply)
	Equipment protected throughout by double insulation or reinforced insulation
	In position of a bi-stable push control
	Out position of a bi-stable push control

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Preface

Digital communication offers so many advantages over analog communication that the majority of today's communications systems are digital.

Unlike analog communication systems, digital systems do not require accurate recovery of the transmitted waveform at the receiver end. Instead, the receiver periodically detects which waveform is being transmitted, among a limited number of possible waveforms, and maps the detected waveform back to the data it represents. This allows extremely low error rates, even when the signal has been corrupted by noise.

The digital circuits are often implemented using application specific integrated circuits (ASIC) and field-programmable gate arrays (FPGA). Although this "system-on-a-chip" approach is very effective for commercial and military applications, the resulting systems do not allow access to internal signals and data and are therefore poorly suited for educational use. It is for this reason that we designed the Communications Technologies Training System.

The Communications Technologies Training System, Model 8087, is a state-of-the-art communications training system. Specially designed for hands-on training, it facilitates the study of many different types of digital modulation/demodulation technologies such as PAM, PWM, PPM, PCM, Delta Modulation, ASK, FSK, and BPSK as well as spectrally efficient technologies such as QPSK, QAM, and ADSL. The system also enables the study of direct-sequence and frequency-hopping spread spectrum (DSSS and FHSS), two key technologies used in modern wireless communication systems (CDMA cellular-telephony networks, Global Positioning System, Bluetooth interface for wireless connectivity, etc.) to implement code-division multiple access (CDMA), improve interference rejection, minimize interference with other systems, etc. The system is designed to reflect the standards commonly used in modern communications systems.

Unlike conventional, hardware-based training systems that use a variety of physical modules to implement different technologies and instruments, the Communications Technologies Training System is based on a Reconfigurable Training Module (RTM) and the Communications Technologies (LVCT) software, providing tremendous flexibility at a reduced cost.

Each of the communications technologies to be studied is provided as an application that can be selected from a menu. Once loaded into the LVCT software, the selected application configures the RTM to implement the communications technology, and provides a specially designed user interface for the student.

The LVCT software provides settings for full user control over the operating parameters of each communications technology application. Functional block diagrams for the circuits involved are shown on screen. The digital or analog signals at various points in the circuits can be viewed and analyzed using the virtual instruments included in the software. In addition, some of these signals are made available at physical connectors on the RTM and can be displayed and measured using conventional instruments.

The courseware for the Communications Technologies Training System consists of a series of student manuals covering the different technologies as well as instructor guides that provide the answers to procedure step questions and to review questions. The Communications Technologies Training System and the accompanying courseware provide a complete study program for these key information-age technologies.

We invite readers of this manual to send us their tips, feedback, and suggestions for improving the book.

Please send these to did@de.festo.com.

The authors and Festo Didactic look forward to your comments.

About This Manual

Manual Objective

When you have completed this manual, you will be familiar with data transmission using three basic types of digital modulation: PCM, DPCM and Delta/CVSD. You will be familiar with the encoding and decoding techniques used and with the advantages and disadvantages of each technology. You will also have learned useful techniques for troubleshooting communications systems using these technologies.

Description

This Student Manual is divided into several units each of which covers one topic. Each unit begins with an Introduction presenting important background information. Following this are a number of exercises designed to present the subject matter in convenient instructional segments. In each exercise, principles and concepts are presented first followed by a step-by-step, hands-on procedure to complete the learning process.

Each exercise contains:

- A clearly defined Exercise Objective
- A Discussion Outline listing the main points presented in the Discussion
- A Discussion of the theory involved
- A Procedure Outline listing the main sections in the Procedure
- A step-by-step Procedure in which the student observes and measures the important phenomena, including questions to help in understanding the important principles.
- A Conclusion
- Review Questions

In this manual, all New Terms are defined in the Glossary of New Terms. In addition, an index of New Terms is provided.

Safety considerations

Safety symbols that may be used in this manual and on the equipment are listed in the Safety Symbols table at the beginning of the manual.

Safety procedures related to the tasks that you will be asked to perform are indicated in each exercise.

Make sure that you are wearing appropriate protective equipment when performing the tasks. You should never perform a task if you have any reason to think that a manipulation could be dangerous for you or your teammates.

About This Manual

Systems of units

Units are expressed using the International System of Units (SI) followed by the units expressed in the U.S. customary system of units (between parentheses).

To the Instructor

You will find in this Instructor Guide all the elements included in the Student Manual together with the answers to all questions, results of measurements, graphs, explanations, suggestions, and, in some cases, instructions to help you guide the students through their learning process. All the information that applies to you is placed between markers and appears in red.

Accuracy of measurements

The numerical results of the hands-on exercises may differ from one student to another. For this reason, the results and answers given in this manual should be considered as a guide. Students who correctly performed the exercises should expect to demonstrate the principles involved and make observations and measurements similar to those given as answers.

Exercise 2-2

Characteristics of Quantization Noise

EXERCISE OBJECTIVE

When you have completed this exercise, you will be familiar with quantization noise in both the time and frequency domains. You will be able to verify how the number of quantization intervals affects quantization noise. You will also be able to demonstrate the effect of low-pass filtering on the quantization error signal.

DISCUSSION OUTLINE

The Discussion of this exercise covers the following points:

- Quantizing
- Distortion due to quantizing

DISCUSSION

Quantizing

After a message signal has been sampled, the next step is to quantize the samples. Quantization consists of approximating the sampled signal by a signal made up of discrete amplitudes. Because a voice signal has a continuous range of amplitudes, the pulses in the sampled signal will have an infinite number of amplitude levels. It is not possible to assign different code words to an infinite number of amplitudes. Fortunately, the human ear cannot detect a minute change in sound intensity. Because of this fact an approximation to the original signal can be constructed of discrete amplitude levels.

A typical input-output relationship of a quantizer is shown in Figure 2-25. The difference between two consecutive discrete values is called a **quantization interval**. When all the quantization intervals are the same size the quantizer is called a linear quantizer. Any input signal falling between the end points of a tread (e.g. between x_i and x_{i+1} in Figure 2-25), the quantizer output will be the discrete level represented by that tread (e.g. y_i in Figure 2-25). To determine the size of the quantization interval of a linear quantizer it is necessary to know the expected maximum and minimum input values (e.g. x_0 and x_n in Figure 2-25) and the number of levels to be used. Then, the quantization interval q will be:

$$\begin{aligned} q &= \frac{2A}{2^n} \\ &= \frac{2A}{N} \end{aligned} \tag{2-4}$$

where q is the quantization interval
A is the maximum value (for a bipolar input signal)
n is the number of code word bits
N is the number of levels

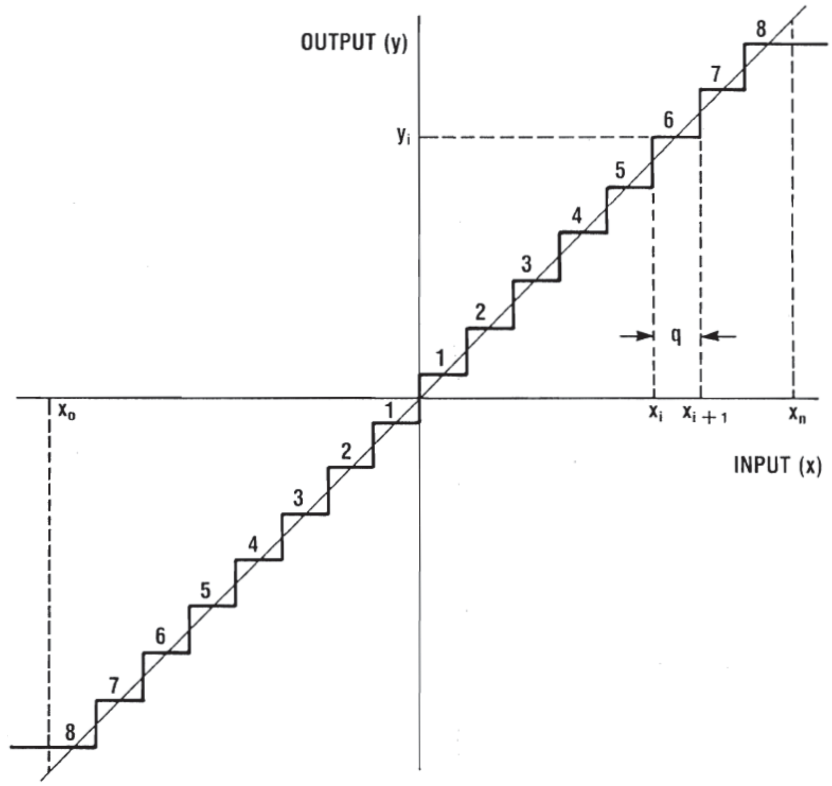


Figure 2-25. Characteristic of a linear quantizer.

For example, if the bipolar input signal has maximum value of 1.0 V (that is, a range of -1 V to +1 V) and the code words are eight bits long, then:

$$
\begin{aligned}
q &= \frac{2 \times 1\,\text{V}}{2^8} \\
&= \frac{2\,\text{V}}{256} \\
&= 7.81\,\text{mV}
\end{aligned}
$$

Distortion due to quantizing

There are two types of distortion associated with a quantizer: overload or clipping distortion and quantization distortion. Overload distortion occurs when the input signal exceeds the quantizer's input range. Once the quantizer range has been exceeded the output will remain at its maximum (or minimum) value until the input falls within the quantizer's input range. As was seen in Unit 1, overload distortion results in a clipped output signal. To avoid clipping, a quantizer is matched to the input signal.

Figure 2-26 shows the error signal introduced by the quantizer. From this figure, it can be seen that quantization error occurs when the input signal is within the input range of the quantizer. It arises because of the difference between the input amplitude and the quantized sampled amplitude and because of the limited sampling rate. The quantization error signal produces **quantization noise** or distortion in the reconstructed message signal. Its frequency spectrum covers a large bandwidth. Low-pass filtering which is used to smooth the waveform will remove most of the quantization error above its cutoff frequency. However, some of the quantization error is in the signal band, and that cannot be removed by the low-pass filter. This will produce a gritty sound at the output of a PCM system called quantization noise.

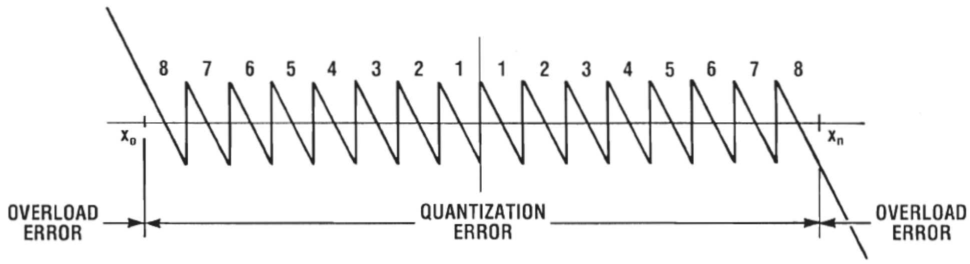


Figure 2-26. Characteristic of quantization and overload errors.

Quantization noise is the result of the quantization process. Since the quantization process adjusts the height of each sample, the original waveform cannot be exactly reconstructed using a low-pass filter as is the case with PAM signals and the classical sampling theorem. The sampling rate will also affect the quantization noise since the quantization error will become larger as the sampling rate decreases.

Figure 2-27 shows an analog input signal and its quantized waveform. Shown below this is the resulting quantization error signal. The maximum amplitude of this error signal is half a quantization interval. The overall amplitude variation is from half a quantization interval to minus half a quantization interval. During a period of small intervals, the error signal appears to be a sawtooth wave.

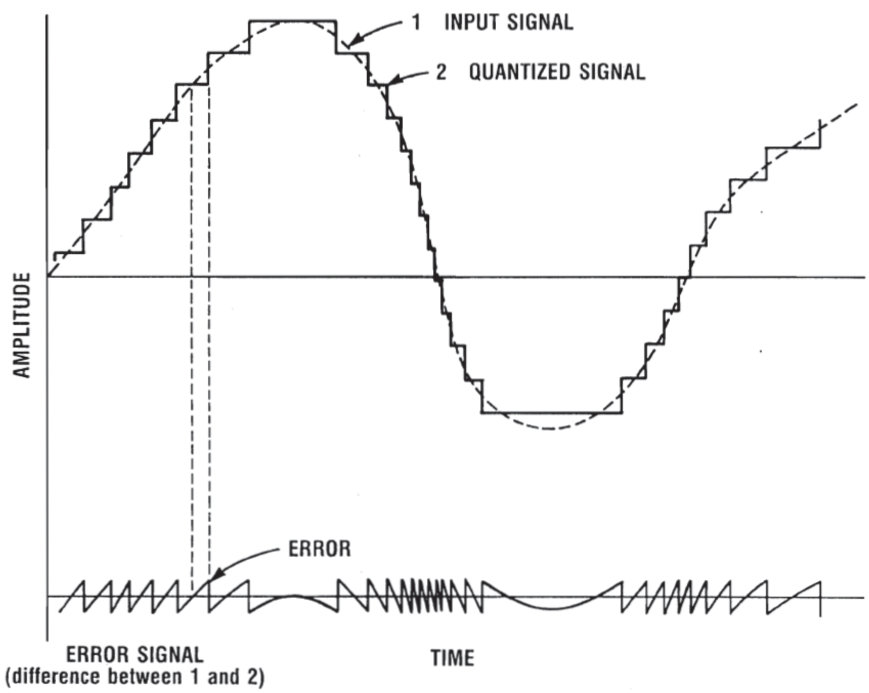


Figure 2-27. Analog input signal, quantized waveform, and quantization error waveform.

PROCEDURE OUTLINE

The Procedure is divided into the following sections:

- Set-up and connections
- Effect of sampling frequency on quantization noise
- Effect of resolution on quantization noise
- Effect of low-pass filtering the PCM Decoder output signal

PROCEDURE

Set-up and connections

1. Turn on the RTM Power Supply and the RTM and make sure the RTM power LED is lit.

File ► Restore Default Settings *returns all settings to their default values, but does not deactivate activated faults.*

2. Start the LVCT software. In the Application Selection box, choose *PCM* and click OK. This begins a new session with all settings set to their default values and with all faults deactivated.

   If the software is already running, choose *Exit* in the *File* menu and restart LVCT to begin a new session with all faults deactivated.

3. Make the *Default* external connections shown on the System Diagram tab of the software. For details of connections to the Reconfigurable Training Module, refer to the RTM Connections tab of the software.

   Click the *Default* button to show the required external connections.

4. As an option, use a conventional oscilloscope during this exercise to observe any of the outputs on the RTM (refer to the RTM Connections tab of the software for the available outputs). Use BNC T-connectors where necessary.

Effect of sampling frequency on quantization noise

5. Make the following settings:

   Generator Settings

   ⊡ Function Generator A
   Function .... Sine
   Output level.. 1.5 V
   Frequency.. 1000 Hz

   PCM Settings

   ⊡ General
   Sampling Frequency .... 142 045 Hz

   ⊡ Pre-Filtering
   Pre-Filtering .... Off

   ⊡ PCM Encoder
   Data source .... A/D
   Compression Law .... Dir
   Bit Interruptor 1111 1111

- PCM Decoder
  Input Code .... Offset
  Output Gain 1

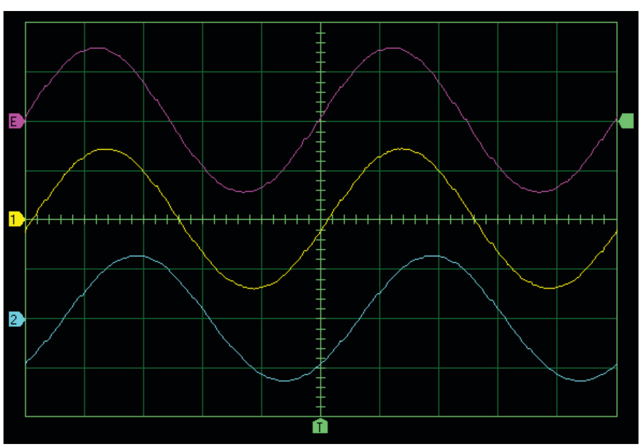
- Post-Filtering
  Notch 1 .... Off
  Notch 1 Center Frequency .... 1000 Hz
  Low-Pass Cutoff Frequency .... 3400 Hz
  Low-Pass Order.. 4th
  Low-Pass Filter Coupling........................ AC
  Notch 2 .... Off

6. Connect the probes as follows.

Probe	Test point		Signal
Oscilloscope E	PCM Decoder	TP1	AUDIO OUTPUT
Oscilloscope 1	Post-Filtering	TP2	Low-pass filter input
Spectrometer	Post-Filtering	TP2	Low-pass filter input
Oscilloscope 2	Post-Filtering	TP3	Low-pass filter output

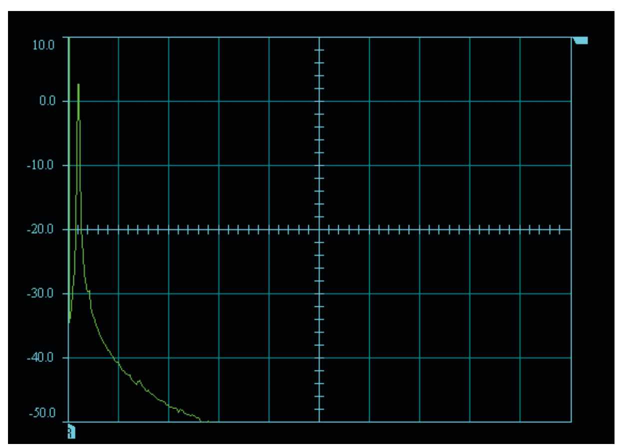
Show the Oscilloscope and the Spectrum Analyzer. Figure 2-28 shows an example of what you should observe. With the Notch 1 filter Off, the PCM Decoder AUDIO OUTPUT and the low-pass filter output are virtually identical, except for a slight delay. The spectrum of the PCM Decoder AUDIO OUTPUT shows only one peak at the message signal frequency.

Oscilloscope Settings:
Channel 1 ....1 V/div
Channel 2 ....1 V/div
Channel E ....1 V/div
Time Base .... 0.2 ms/div
Trigger Slope ....Rising
Trigger Level................................ 0 V
Trigger Source .... Ext



E PCM Decoder AUDIO OUTPUT
1 Low-pass filter input (same as E)
2 Low-pass filter output (reconstructed signal)

Spectrum Analyzer Settings:
Maximum Input .... 10 dBV
Scale Type...................... Logarithmic
Scale .... 10 dBV/div
Averaging..Off
Time Window .... Square
Frequency Span .... 5 kHz/div
Reference Frequency .... 0 kHz



PCM Decoder AUDIO OUTPUT

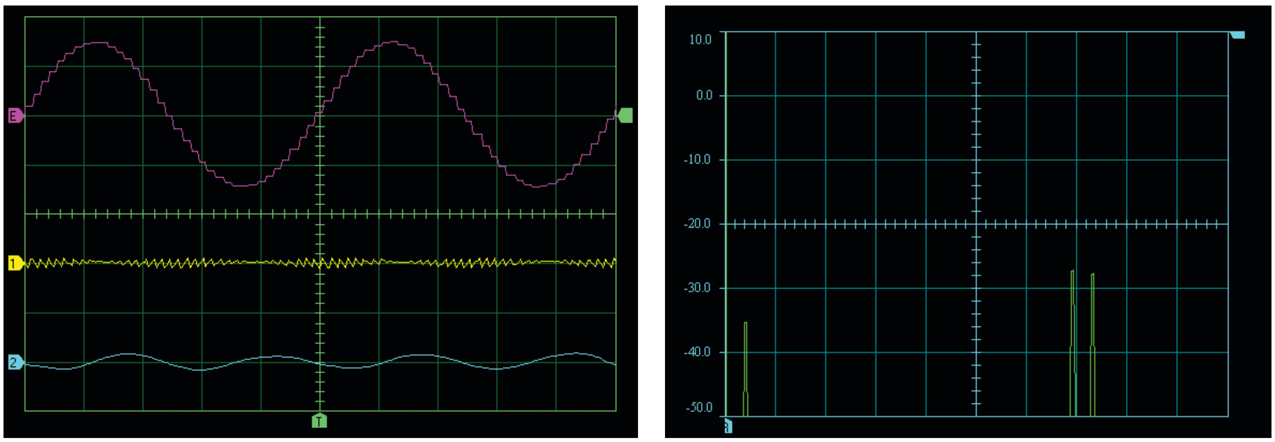
Figure 2-28. Waveforms and spectrum (Notch 1 filter off, f_s = 142 045 Hz, f_m = 1000 Hz).

7. Turn the Notch 1 filter On. Adjust the Channel 1 and 2 Scale settings on the Oscilloscope as desired to view the filtered waveforms.

   Reduce the Sampling Frequency step-by-step while observing the instruments. Figure 2-29 and Figure 2-30 show examples of what you should observe at different Sampling Frequencies.

Oscilloscope Settings:
Channel 1 ....1 V/div
Channel 2 .... 100 mV/div
Channel E..............................1 V/div

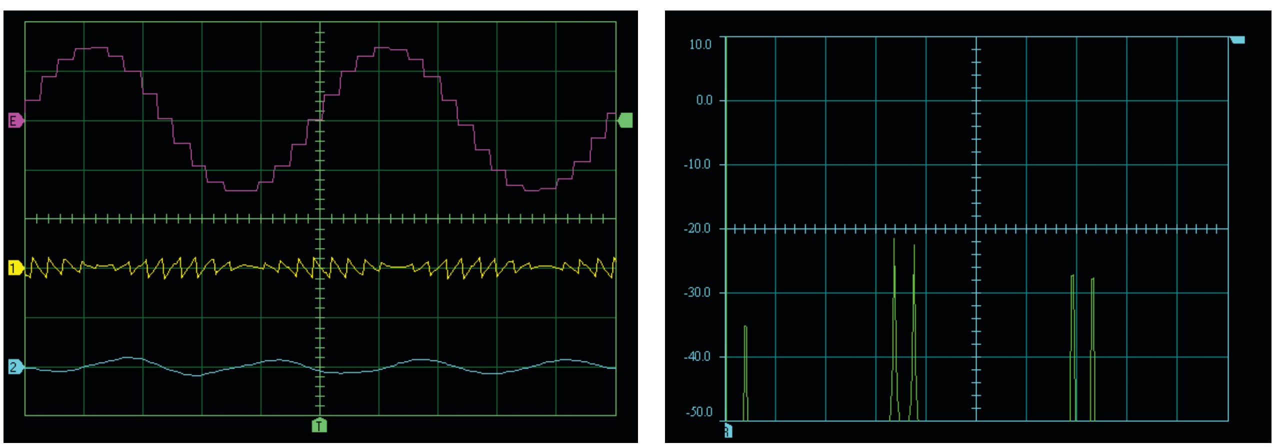
Spectrum Analyzer Settings:
Maximum Input ....10 dBV
Frequency Span ....5 kHz/div



E PCM Decoder output
1 Notch 1 filter output (quantization error)
2 Low-pass filter output (audible quantization noise)

Notch 1 filter output (quantization error)

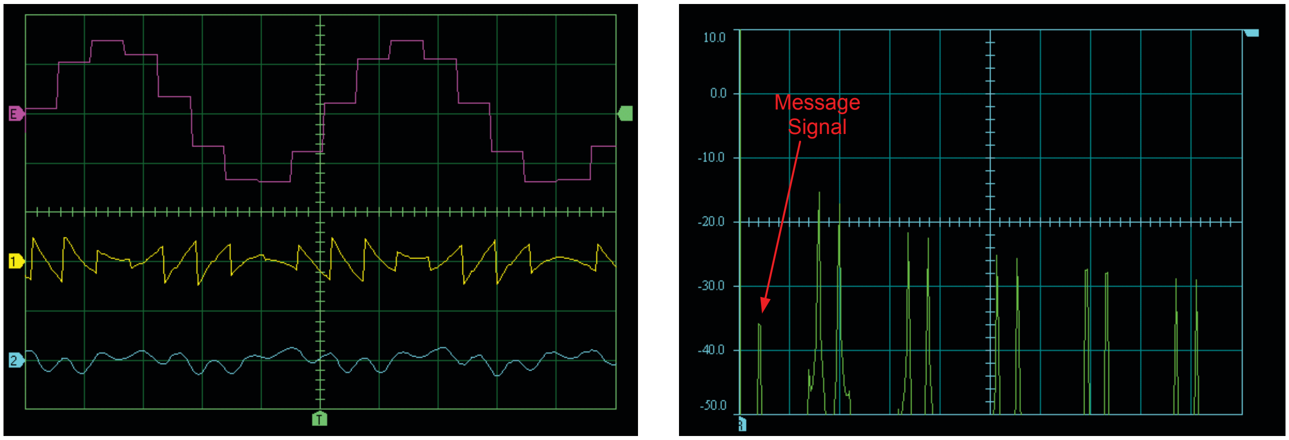
Figure 2-29. Waveforms and spectrum (Fs = 35 511 Hz, f_m = 1000 Hz).



E PCM Decoder output
1 Notch 1 filter output (quantization error)
2 Low-pass filter output (audible quantization noise)

Notch 1 filter output (quantization error)

Figure 2-30. Waveforms and spectrum (f_s = 17 756 Hz, f_m = 1000 Hz).



E PCM Decoder output
1 Notch 1 filter output (quantization error)
2 Low-pass filter output (audible quantization noise)

Notch 1 filter output (quantization error)

Figure 2-31. Waveforms and spectrum (f_s = 8878 Hz, f_m = 1000 Hz).

What is the effect of reducing the Sampling Frequency?

As the Sampling Frequency decreases, the quantization error increases in amplitude and in the number of frequency components present. The spacing of the frequency components in the spectrum decreases.

Does the low-pass filter remove all of the quantization error?

No, some quantization noise remains after passing through the low-pass filter. This is visible and audible in the waveform of the low-pass filter output.

8. Make the following settings:

Generator Settings

⊡ Function Generator A
Function.. Sine
Output level .... 1.5 V
Frequency .... 1000 Hz

PCM Settings

⊡ General
Sampling Frequency.................................... 142 045 Hz

⊡ Post-Filtering
Notch 1 .... Off
Low-Pass Cutoff Frequency .... 10 000 Hz
Low-Pass Order.. 4th
Low-Pass Filter Coupling.............................. AC
Notch 2 .... Off

Using the supplied adapter(s), connect the headphones to the Post-Filtering OUTPUT (refer to the RTM Connections tab of the software).

Gradually reduce the Sampling Frequency as you observe the spectrum and listen to the sound of the reconstructed signal.

What is the audible effect of reducing the Sampling Frequency on the reconstructed signal?

The effect of reduce the Sampling Frequency is to introduce undesired frequency components into the reconstructed signal.

With a low Sampling Frequency setting, vary the Low-Pass Cutoff Frequency as you listen to the sound of the reconstructed signal. Is reducing the cutoff frequency of the low-pass filter an effective way to compensate for a low Sampling Frequency?

As the cutoff frequency is reduced, some of the undesired frequency components are attenuated. Unfortunately, this also attenuates the higher frequencies of the message signal.

Effect of resolution on quantization noise

9. The probes should be connected as follows:

Probe	Test point		Signal
Oscilloscope E	PCM Decoder	TP1	AUDIO OUTPUT
Oscilloscope 1	Post-Filtering	TP2	Notch 1 filter output
Spectrometer	Post-Filtering	TP2	Notch 1 filter output
Oscilloscope 2	Post-Filtering	TP3	Low-pass filter output

Make the following settings:

Generator Settings

- Function Generator A
  Function .... Sine
  Output level.. 1.5 V
  Frequency .... 1000 Hz

PCM Settings

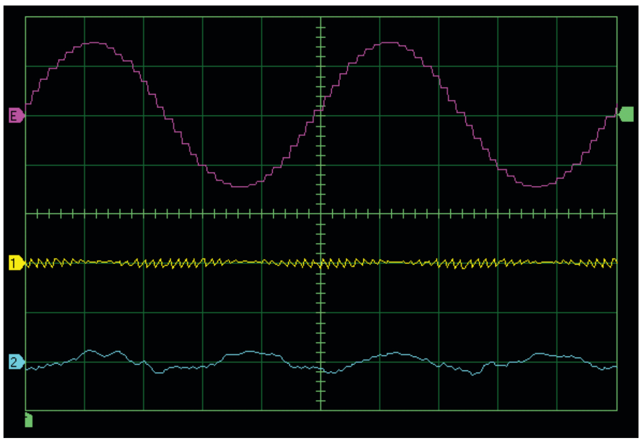
- General
  Sampling Frequency .... 35 511 Hz
- Pre-Filtering
  Pre-Filtering .... Off
- PCM Encoder
  Data source .... A/D
  Compression Law .... Dir
  Bit Interruptor 1111 1111
- PCM Decoder
  Input Code .... Offset
  Output Gain 1

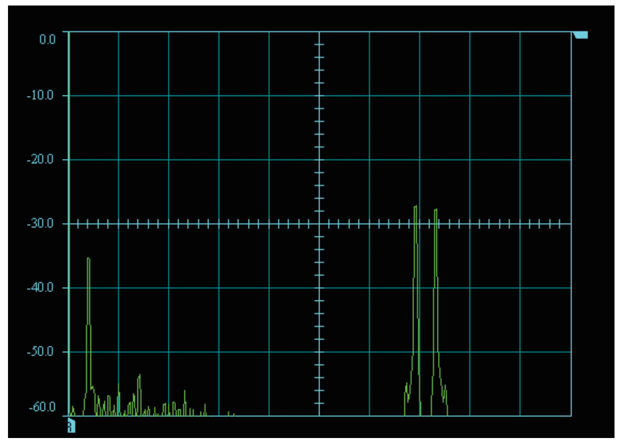
☐ Post-Filtering
Notch 1 .... On
Notch 1 Center Frequency .... 1000 Hz
Low-Pass Cutoff Frequency .... 10 000 Hz
Low-Pass Order... 4th
Low-Pass Filter Coupling.............................. AC
Notch 2 .... Off

In the PCM Encoder, use the Bit Interruptor switches to remove one bit at a time, each time removing the least-significant remaining bit. Figure 2-32, Figure 2-33, and Figure 2-34 show examples of what you should observe.

Oscilloscope Settings:
Channel 1 ....1 V/div
Channel 2 .... 100 mV/div
Channel E..............................1 V/div

Spectrum Analyzer Settings:
Maximum Input .... 0 dBV
Frequency Span................. 5 kHz/div

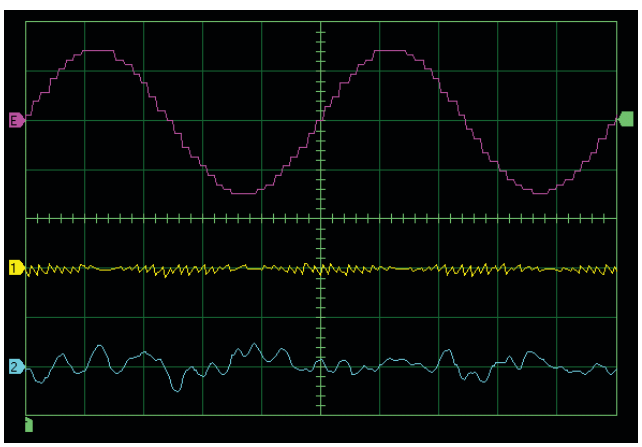


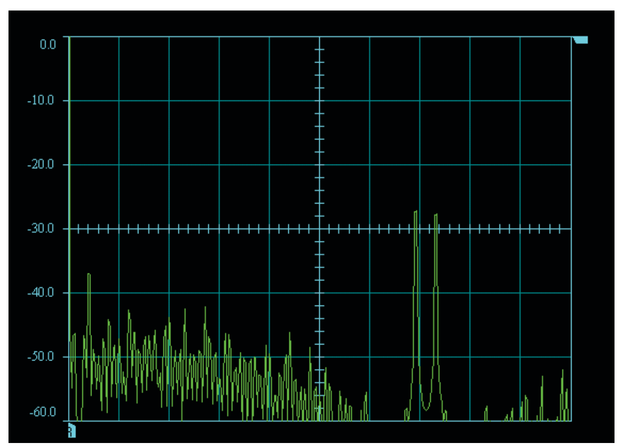


E PCM Decoder output
1 Notch 1 filter output (quantization error)
2 Low-pass filter output (audible quantization noise)

Notch 1 filter output (quantization error)

Figure 2-32. Waveforms and spectrum (f_s = 35551 Hz, f_m = 1000 Hz, 7-bit resolution).

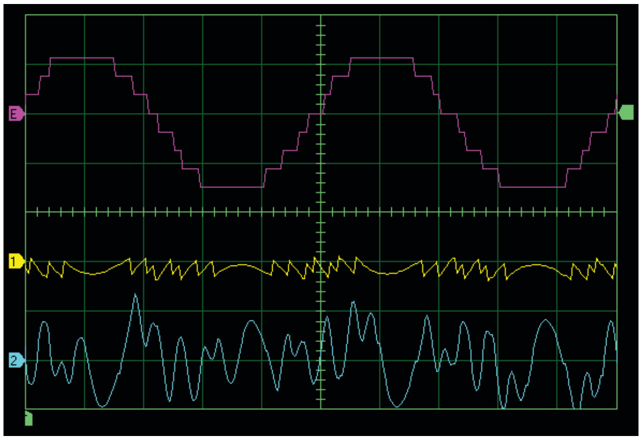




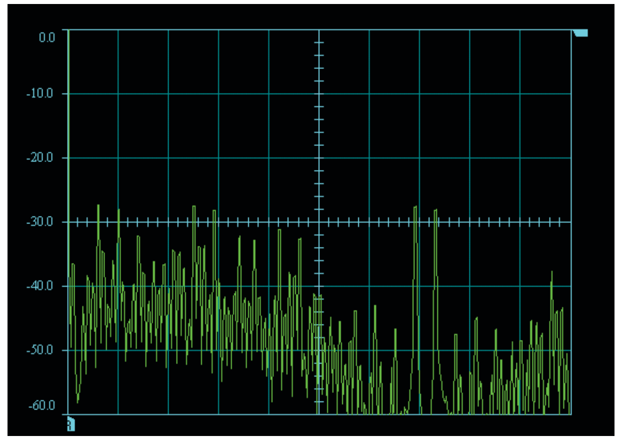
E PCM Decoder output
1 Notch 1 filter output (quantization error)
2 Low-pass filter output (audible quantization noise)

Notch 1 filter output (quantization error)

Figure 2-33. Waveforms and spectrum (f_s = 35551 Hz, f_m = 1000 Hz, 5-bit resolution).



E PCM Decoder output
1 Notch 1 filter output (quantization error)
2 Low-pass filter output (audible quantization noise)



Notch 1 filter output (quantization error)

Figure 2-34. Waveforms and spectrum (f_s = 35551 Hz, f_m = 1000 Hz, 3-bit resolution).

Describe the effect of reducing the bit resolution.

Reducing the bit resolution increases the quantization error and the quantization noise in the filtered signal.

Effect of low-pass filtering the PCM Decoder output signal

10. Make the following settings:

PCM Settings

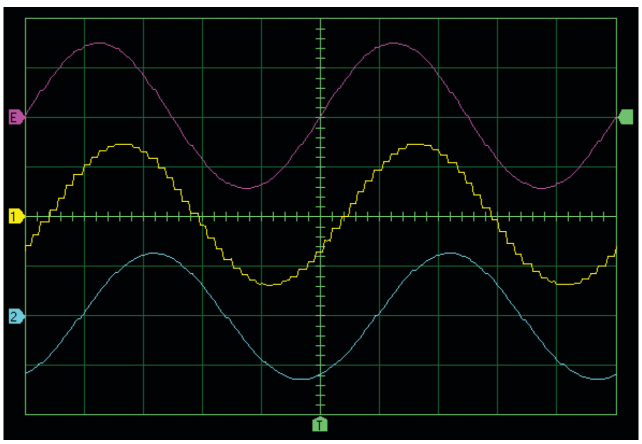
- General
  Sampling Frequency .... 35 511 Hz
- PCM Encoder
  Bit Interruptor 1111 1111
- Post-Filtering
  Notch 1 .... Off
  Low-Pass Cutoff Frequency .... 3400 Hz
  Low-Pass Order 4th
  Low-Pass Filter Coupling .... AC
  Notch 2 .... Off

Connect the probes as follows.

Probe	Test point		Signal
Oscilloscope E	PCM Encoder	TP1	AUDIO INPUT
Oscilloscope 1	Post-Filtering	TP2	Low-pass filter input (PCM Decoder AUDIO OUTPUT)
Oscilloscope 2	Post-Filtering	TP3	Low-pass filter output
Spectrometer	Post-Filtering	TP3	Low-pass filter output

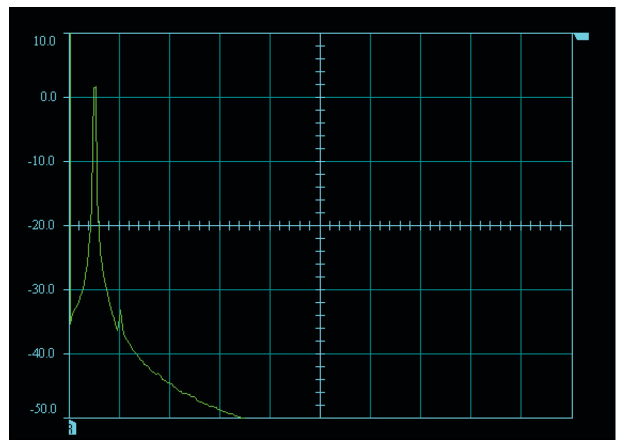
Figure 2-35 shows an example of what you should observe.

Oscilloscope Settings:
Channel 1 .... 1 V/div
Channel 2 .... 1 V/div
Channel E .... 1 V/div
Time Base .... 0.2 ms/div
Trigger Slope .... Rising
Trigger Level............................... 0 V
Trigger Source .... Ext



E PCM Encoder AUDIO INPUT (message signal
1 PCM Decoder AUDIO OUTPUT
2 Low-pass filter output (reconstructed signal)

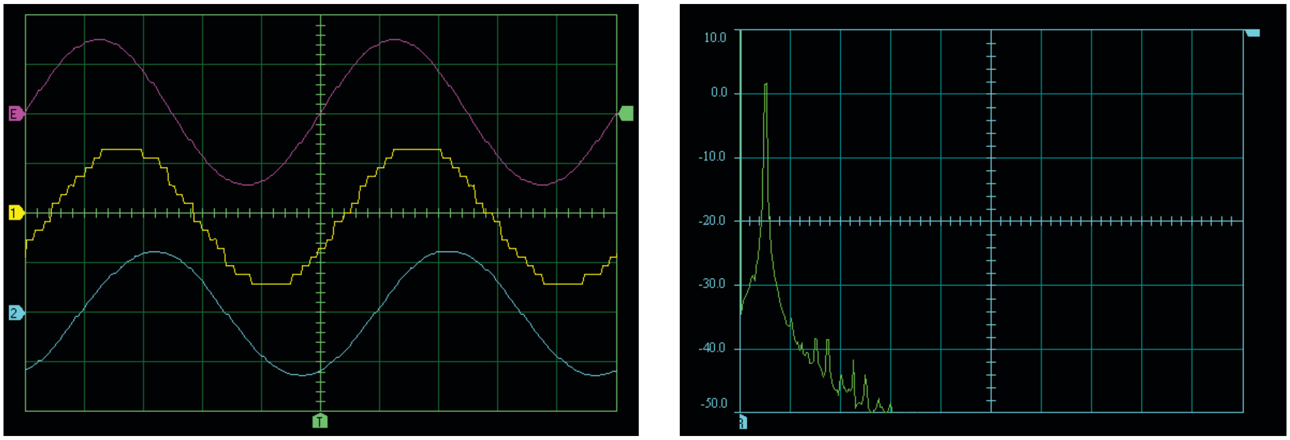
Spectrum Analyzer Settings:
Maximum Input .... 10 dBV
Scale Type .... Logarithmic
Scale .... 10 dBV/div
Averaging .... Off
Time Window .... Square
Frequency Span................. 2 kHz/div
Reference Frequency .... 0 kHz



Low-pass filter output (reconstructed signal)

Figure 2-35. Waveforms and spectrum (f_s = 35551 Hz, f_m = 1000 Hz, 8-bit resolution).

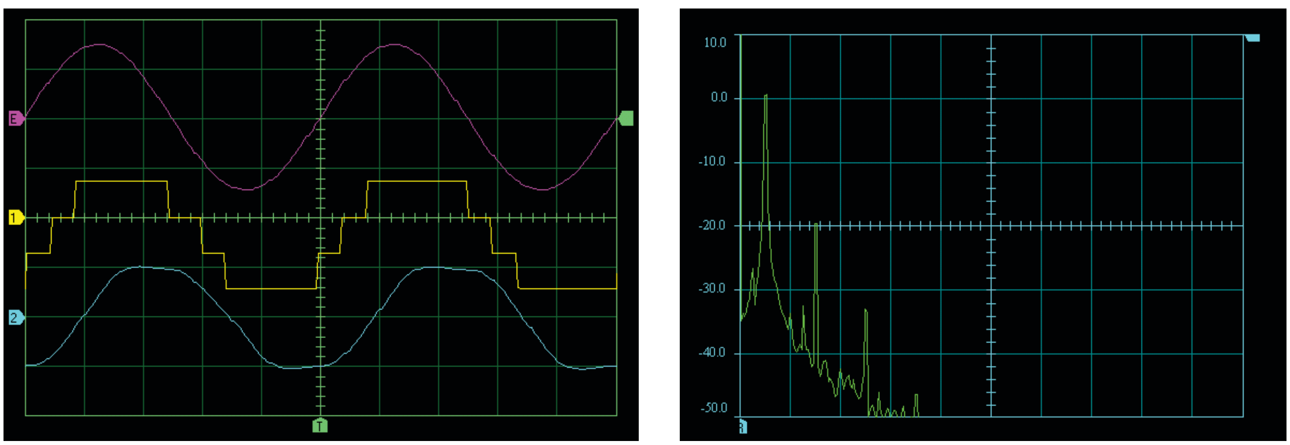
Gradually reduce the bit resolution while observing the instruments and listening to the sound of the reconstructed signal (see Figure 3-35 and Figure 3-36).



E PCM Encoder AUDIO INPUT
1 PCM Decoder AUDIO OUTPUT
2 Low-pass filter output (reconstructed signal)

Low-pass filter output (reconstructed signal)

Figure 2-36. Waveforms and spectrum (f_s = 35551 Hz, f_m = 1000 Hz, 4-bit resolution).



E PCM Encoder AUDIO INPUT
1 PCM Decoder AUDIO OUTPUT
2 Low-pass filter output (reconstructed signal)

Low-pass filter output (reconstructed signal)

Figure 2-37. Waveforms and spectrum (f_s = 35551 Hz, f_m = 1000 Hz, 2-bit resolution).

Does the low-pass filter remove all of the quantization noise? Explain.

No, the low-pass filter does not remove all of the quantization noise. Some of the quantization noise falls in the passband of the filter so it is not removed.

What is the audible effect of the quantization noise?

At first, the quantization noise level is small and it sounds much like background noise. As the resolution is reduced, the noise level increases and sounds increasingly like distortion.

11. If you wish, you can repeat Steps 9 and 10 using a compatible audio signal source connected to the PRE-FILTERING INPUT of the RTM. The Model 9415 FM / PM Receiver is recommended. Tune to a station where a person is speaking and observe the effect of reducing the bit resolution on the the quality of the audio signal.

Alternatively, you could connect the line output of the computer sound card, using the appropriate cables and adapters, to the PRE-FILTERING INPUT of the RTM, and then connect to an Internet radio station using a Web browser as an audio source.

It is your responisiblity to ensure that any signal from an external source is compatible with the inputs of the Data Acquisition Interface, Model 9466. The analog inputs of this module are designed for a voltage range of ±1.5 V and have an impedance of 10 kΩ.

12. When you have finished using the system, exit the LVCT software and turn off the equipment.

Conclusion

In this exercise, you observed quantization error in the time domain and in the frequency domain. In the time domain, you showed the effect of the sampling frequency on the quantization error waveform. You learned that quantization noise decreases when the number of quantization intervals is increased. You also observed the broad bandwidth of quantization noise and how it can be reduced by increasing the number of quantization intervals. You further observed that quantization noise cannot be totally eliminated by low-pass filtering the output of the PCM Decoder.

Review Questions

1. Explain how quantization noise arises.

Quantization noise arises because there are a finite number of code words available to represent an infinite number of amplitudes. The difference between the original message signal and the quantized message signal is the quantization error. This contains frequency components that were not present in the message signal and which produce the quantization noise.

2. For a given sampling rate, how can quantization noise be reduced?

Quantization noise can be reduced by increasing the number of quantization intervals used to quantize the message signal.

3. Why does the quantization noise get worse when you use a lower sampling rate to quantize a sine wave?

The effect is similar to reducing the number of quantization intervals. With a lower sampling rate the sampled value is held for longer between samples. Therefore, there will be a greater difference between the message signal and the quantized signal.

4. What effect does a low-pass filter have on quantization noise?

The low-pass filter will remove some of the quantization noise but not all. The quantization noise frequency components below the filter's cutoff frequency will pass through the filter.

5. What does quantization noise sound like with a 6-, 7-, or 8-bit PCM system?

Quantization noise sounds like a gritty background noise.

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