Lab-Volt® Communications Systems

Digital Communications 1

FACET®

Student Workbook

91581-00
Edition 4

3091581000503

FOURTH EDITION

Second Printing, March 2005

ISBN 0-86657-283-X

Lab-Volt License Agreement

By using the software in this package, you are agreeing to become bound by the terms of this License Agreement, Limited Warranty, and Disclaimer.

This License Agreement constitutes the complete agreement between you and Lab-Volt. If you do not agree to the terms of this agreement, do not use the software. Promptly return the F.A.C.E.T. Resources on Multimedia (CD-ROM) compact discs and all other materials that are part of Lab-Volt's F.A.C.E.T. product within ten days to Lab-Volt for a full refund or credit.

1. License Grant. In consideration of payment of the license fee, which is part of the price you paid for this Lab-Volt product, Lab-Volt, as Licensor, grants to you, the Licensee, a nonexclusive, nontransferable license to use this copy of the CD-ROM software with the corresponding F.A.C.E.T. Lab-Volt reserves all rights not expressly granted to the Licensee.

2. Ownership. As the Licensee, you own the physical media on which the CD-ROM is originally or subsequently recorded or fixed, but Lab-Volt retains title to and ownership of the software programs recorded on the original compact disc and any subsequent copies of the CD-ROM, regardless of the form or media in or on which the original and other copies may exist. This license is not a sale of the original software program of Lab-Volt's CD-ROM or any portion or copy of it.

3. Copy Restrictions. The CD-ROM software and the accompanying materials are copyrighted and contain proprietary information and trade secrets of Lab-Volt. Unauthorized copying of the CD-ROM even if modified, merged, or included with other software or with written materials is expressly forbidden. You may be held legally responsible for any infringement of Lab-Volt's intellectual property rights that is caused or encouraged by your failure to abide by the terms of this agreement. You may make copies of the CD-ROM solely for backup purposes provided the copyright notice is reproduced in its entirety on the backup copy.

4. Permitted Uses. This CD-ROM, Instructor's Guide, and all accompanying documentation is licensed to you, the Licensee, and may not be transferred to any third party for any length of time without the prior written consent of Lab-Volt. You may not modify, adapt, translate, reverse engineer, decompile, disassemble, or create derivative works based on the Lab-Volt product without the prior written permission of Lab-Volt. Written materials provided to you may not be modified, adapted, translated, or used to create derivative works without the prior written consent of Lab-Volt.

5. Termination. This agreement is effective until terminated. It will terminate automatically without notice from Lab-Volt if you fail to comply with any provisions contained herein. Upon termination you shall destroy the written materials, Lab-Volt's CD-ROM software, and all copies of them, in part or in whole, including modified copies, if any.

6. Registration. Lab-Volt may from time to time update the CD-ROM. Updates can be made available to you only if a properly signed registration card is filed with Lab-Volt or an authorized registration card recipient.

7. Miscellaneous. This agreement is governed by the laws of the State of New Jersey.

Limited Warranty and Disclaimer

This CD-ROM software has been designed to assure correct operation when used in the manner and within the limits described in this Instructor's Guide. As a highly advanced software product, it is quite complex; thus, it is possible that if it is used in hardware configurations with characteristics other than those specified in this Instructor's Guide or in environments with nonspecified, unusual, or extensive other software products, problems may be encountered by a user. In such cases, Lab-Volt will make reasonable efforts to assist the user to properly operate the CD-ROM but without guaranteeing its proper performance in any hardware or software environment other than as described in this Instructor's Guide.

This CD-ROM software is warranted to conform to the descriptions of its functions and performance as outlined in this Instructor's Guide. Upon proper notification and within a period of one year from the date of installation and/or customer acceptance, Lab-Volt, at its sole and exclusive option, will remedy any nonconformity or replace any defective compact disc free of charge. Any substantial revisions of this product, made for purposes of correcting software deficiencies within the warranty period, will be made available, also on a licensed basis, to registered owners free of charge. Warranty support for this product is limited, in all cases, to software errors. Errors caused by hardware malfunctions or the use of nonspecified hardware or other software are not covered.

LICENSOR MAKES NO OTHER WARRANTIES OF ANY KIND CONCERNING THIS PRODUCT, INCLUDING WARRANTIES OR MERCHANTABILITY OR OF FITNESS FOR A PARTICULAR PURPOSE. LICENSOR DISCLAIMS ALL OBLIGATIONS AND LIABILITIES ON THE PART OF LICENSOR FOR DAMAGES, INCLUDING BUT NOT LIMITED TO SPECIAL OR CONSEQUENTIAL DAMAGES ARISING OUT OF OR IN CONNECTION WITH THE USE OF THE SOFTWARE PRODUCT LICENSED UNDER THIS AGREEMENT.

Questions concerning this agreement and warranty and all requests for product repairs should be directed to the Lab-Volt field representative in your area.

LAB-VOLT SYSTEMS, INC.
P.O. Box 686
Farmingdale, NJ 07727
Attention: Program Development
Phone: (732) 938-2000 or (800) LAB-VOLT
Fax: (732) 774-8573
Technical Support: (800) 522-4436
Technical Support E-Mail: techsupport@labvolt.com

Table of Contents

Introduction

This Student Workbook provides a unit-by-unit outline of the Fault Assisted Circuits for Electronics Training (F.A.C.E.T.) curriculum.

The following information is included together with space to take notes as you move through the curriculum.

- The unit objective
- Unit fundamentals
- A list of new terms and words for the unit
- Equipment required for the unit
- The exercise objectives
- Exercise discussion
- Exercise notes

The **Appendix** includes safety information.

UNIT 1 – INTRODUCTION TO DIGITAL COMMUNICATIONS 1

UNIT OBJECTIVE

At the completion of this unit, you will be able to describe the basic principles of pulse modulation, digital communications, and the components on the DIGITAL COMMUNICATIONS 1 circuit board.

UNIT FUNDAMENTALS

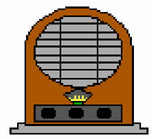
You are in a communication and information explosion! The rapid development of digital communication technology is sustaining this explosion. Almost everyday the printed and electronic press talks about

- compact disks
- faxes
- E-mail
- interactive computer-based training
- the digital information superhighway
- fiber optic telephone networks
- integrated services digital network (ISDN)
- interactive multimedia TV
- virtual reality.

You are taking this interactive DIGITAL COMMUNICATIONS 1 course by digitally communicating with the Lab-Volt F.A.C.E.T. Computer-Based Laboratory training system.

One of the earliest forms of digital communications was sending simple messages by smoke signals. Words were encoded into data represented by puffs of smoke.

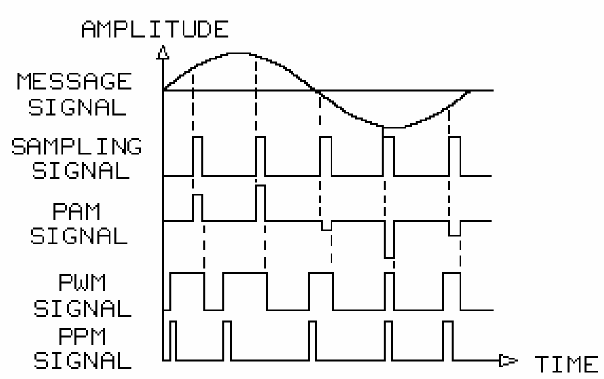
The first electrical communications system was the telegraph, which appeared in 1844. In 1876, the first telephone was patented; today, long-distance calls are digitally transmitted. The analog radio was demonstrated in 1895, and it dominated wireless communication for 50 years. Television with analog circuits appeared in the late 1920s but did not become popular for communication and entertainment until the 1950s. The invention of the transistor in 1948 and the rapid growth of integrated circuits since the 1960s have paved the way for today's digital computers and communication systems.



Analog communication signals, such as the signal your AM radio receives, are continuous and vary in amplitude, frequency, or phase.

Digital signals are discrete, discontinuous pulses that have one of two voltage levels.
In this course, you will learn about

- pulse modulation, which includes pulse-amplitude modulation (PAM) and pulse-time modulation (PTM)
- digital modulation, which includes pulse-code modulation (PCM) and delta modulation (DM)
- time-division multiplexing (TDM) of PAM and PCM signals.
- the effect of noise on pulse and digital modulation signals.
- troubleshooting pulse and digital communication signals.

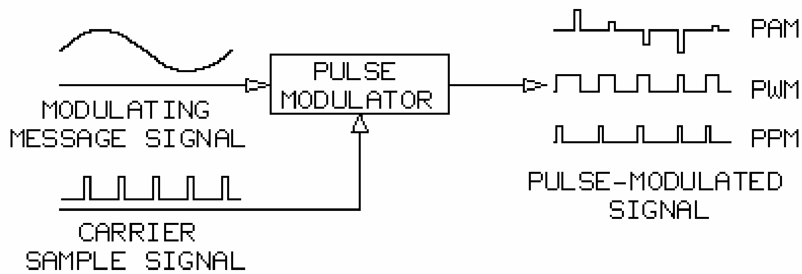


Pulse modulation produces discontinuous pulses that represent **amplitude samples** of the **analog message signal**

Pulse modulation includes

- pulse-amplitude modulation (PAM)
- **pulse-width modulation (PWM)**
- **pulse-position modulation (PPM)**

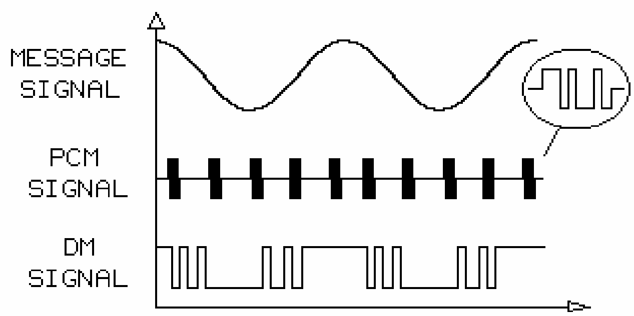
PWM and PPM are types of pulse time modulation (PTM).



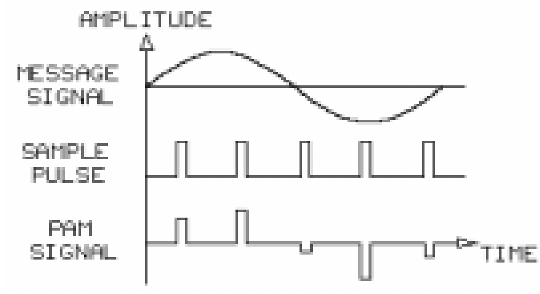
In pulse modulation, the message signal is the **modulating signal**; and the sample signal is the **carrier signal**. The message signal modulates the sample signal to produce the pulse modulated signal (PAM, PWM, or PPM).

Although pulse-modulated signals are discontinuous, these pulses are not true digital signals. PAM, PWM, and PPM are, respectively, the pulse equivalents of AM, FM, and PM of analog carrier signals. To understand digital communications, you must understand pulse sampling and modulation.

For pulse-modulated signals to contain all of the **intelligence** in the analog message signal, the sample frequency (f_S) must be greater than two times the maximum message signal frequency

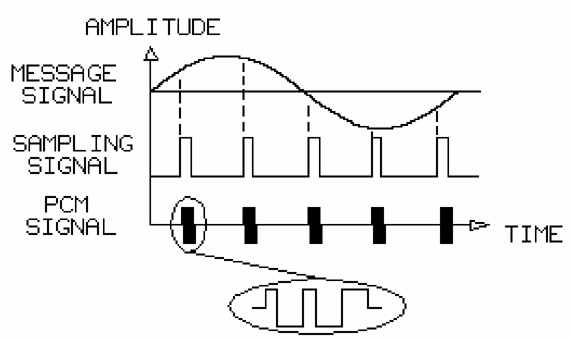


Pulse-code modulation (PCM) is a process that produces a binary code, usually 8 bits, for each amplitude sample. PCM signals are binary encoded PAM signals. Delta modulation (DM) is a process that produces a 1-bit code, which indicates an increase or decrease in the message signal's amplitude.

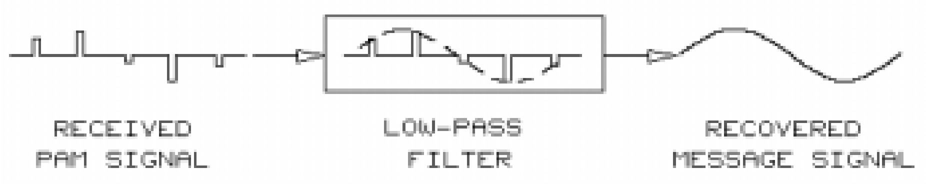


In PAM, the blank time space between the transmitted PAM amplitude pulses is uniform because the sampling frequency is constant. PAM signals from other message signals can be inserted into this blank time space for the transmission of **multiplexed signals**.

Two time-division multiplexed (TDM) message signals appear as shown.



- Pulse-modulated signals, such as PAM, PWM, and PPM, are not transmitted directly but in most cases are used to frequency-modulate an analog carrier.
- However, PCM or DM signals, which are true digital codes, can be transmitted by digital computer methods.



In a receiver, a low-pass filter reconstructs (demodulates) PAM, PWM, and PPM signals into a recovered message signal. PCM and DM signals are first decoded and partially reconstructed before a low-pass filter recovers the message signal.

- Locate and examine the seven circuit blocks on your DIGITAL COMMUNICATIONS 1 circuit board. board.
  PAM
  PTM
  PCM
  PAM-TDM
  DELTA
  CHANNEL SIMULATOR
  SPEAKER AMP

An explanation of each circuit block is given in Exercise 2, Circuit Board Familiarization.

The F.A.C.E.T. DIGITAL COMMUNICATIONS 1 Computer-Based Laboratory course will prepare you to take advantage of this era of digitally transmitted information.

NEW TERMS AND WORDS

pulse-amplitude modulation (PAM) - a modulation method in which the amplitude of each pulse sample is proportional to the amplitude of the message signal at the time of sampling.
pulse-time modulation (PTM) - a type of pulse modulation where the timing of the pulse varies with the message signal's amplitude.
pulse-code modulation (PCM) - a modulation process that produces binary serial codes for amplitude samples of the analog message signal.
delta modulation (DM) - a modulation process that produces 1-bit codes that indicate an increase or decrease in the message signal's amplitude.
time-division multiplexing (TDM) - a process that transmits two or more message signals over the same line by using a different time interval (slot) for each signal.
noise - an unwanted signal that interferes with a communication signal.
pulse-width modulation (PWM) - a type of PTM where the pulse width varies with the message signal's amplitude.
pulse-position modulation (PPM) - a type of PTM where the pulse position varies with the message signal's amplitude.
modulating signal - a signal that varies some characteristic
(amplitude, frequency, or phase)
of a carrier signal; in pulse modulation, the modulating signal is the message signal.
carrier signal - a high-frequency transmission signal that is modulated by a message signal; in pulse modulation, the carrier signal is the sample signal.
amplitude samples - periodic pulses whose amplitudes are directly proportional to a message signal.
analog message signal - an analog signal that contains information (intelligence).
multiplexed signals - signals from different sources combined on a single transmission channel in a maner that permits independent recovery of each signal in a receiver.
intelligence - the information contained in a message signal.
frame - a period of time equal to the sampling period that is divided into smaller equal periods called time slots.
time slots - periods of time within a frame that are equal to the period of the samples.
full-duplex transmission - transmission that can occur simultaneously in both directions between communicators.

EQUIPMENT REQUIRED

In order to complete the following exercises, you will need:
F.A.C.E.T. base unit
Oscilloscope, dual trace
DIGITAL COMMUNICATIONS 1 circuit board

NOTES

Exercise 1 – Digital Communication Concepts

EXERCISE OBJECTIVE

When you have completed this exercise, you will be able to describe the basic principles of pulse modulation and digital communications recognize digitally and pulse modulated signals

DISCUSSION

- For effective electronic communication, the complete continuous analog voice, music, or data signals do not have to be transmitted.
- Effective communication can occur only if you transmit periodic amplitude samples of the analog signal, provided the samples are taken at a high enough frequency.
- For the amplitude samples of the message signal to contain all the intelligence in the original signal, the samples must be taken at a frequency greater than twice the maximum analog message frequency.
- Usually, sampling frequencies are in the range of 8 kHz to 32 kHz.
- The samples of the analog signal are pulses; each pulse represents the message signal's amplitude.
- In PAM, PWM, and PPM, the electronic signals representing the pulse height, width, and position are transmitted.
- In PCM and DM, binary digital codes representing the message signal's amplitude are transmitted.
- PCM signals are usually 8-bit codes.
- DM signals are 1-bit codes.
- Time-division multiplexing (TDM) combines several channels of PAM or PCM signals for transmission.
- In TDM, the time (period) between individual message signals samples is a **frame**.
- **Time slots** are smaller periods of time within the frame where each signal is positioned.
- Each PAM or PCM signal representing a distinct message signal sample is assigned its own time slot within a frame.
- A frame of time slots repeats at the sampling frequency.
- **Full-duplex transmission** is transmission in both directions at the same time.
- TDM permits full-duplex transmission of PCM signals.
- Telephone communication is an example of full-duplex transmission that uses TDM for long-distance transmission.
- The FILTER in the receiver section of the PAM circuit block is also used to recover the message signal from PWM or PPM signals generated in the PTM circuit block.

NOTES

Exercise 2 – Circuit Board Familiarization

EXERCISE OBJECTIVE

When you have completed this exercise, you will be able to describe digital communication circuits and locate circuits on the DIGITAL COMMUNICATIONS 1 circuit board. You will connect circuits and use an oscilloscope to observe signals.

DISCUSSION

- The seven circuit blocks that make up the DIGITAL COMMUNICATIONS 1 circuit board are
  PAM
  PTM
  CHANNEL SIMULATOR
  SPEAKER AMP
  PCM
  PAM-TDM
  DELTA

- The PAM, PTM, PCM, PAM-PTM, and DELTA circuit blocks contain transmitters and receivers.
  - The transmitter is the circuit section that produces the pulse-modulated or digital signal.
  - The receiver is the circuit section that recovers or reconstructs the message signal from the transmitted pulse-modulated or digital signal.
  - The FILTER in the PAM circuit block is also used for recovering the message signal from PTM signals.
  - The following circuits are found in digital communication transmitters and receivers:

sampler	limiter	sample/hold
filter	comparator	CODEC
differentiator	PLL	adder
channel simulator	ramp generator	compressor
digital sampler	expander	counter
speaker amplifier		

  - The resources contain a circuit schematics selection.
  - On the circuit board, there are two internally generated message signals: M1 and M2.
  - M1 is a sine wave about 5 $V_{pk\text{-}pk}$ and 1 kHz.
  - M2 is a sine wave about 5 $V_{pk\text{-}pk}$ and 2 kHz.
  - In each circuit block, the M1 and M2 message signals are synchronized with the sample signals so that you can easily observe the relationships of the message, sample, and pulse signals with an oscilloscope.

- In the PAM-TDM circuit block, M1 and S1 are synchronized, and M2 and S2 are synchronized.
- You will also use a signal generator to provide a sine wave message signal that is not synchronized to the sample signal.
- You use the signal generator to provide the message signal to determine the effect of the ratio of sampling and message frequencies on the recovered message signal.
- The frequencies of the sampling signals (SP, SH, S1, S2, H1, H2, SX, SR, and CLK) on the circuit board can be changed by circuit modification (CM) switches.
- For each type of modulation process, you will be able to observe the effect of sampling frequency on the recovered message.
- CM switches also enable you to see the effect of changing circuit component values on pulse modulation and demodulation.
- The circuit board contains 12 fault switches.
- When turned on, a fault switch introduces a fault in one of the circuits.
- The fault may be an open circuit, a short circuit, or a change in the value of a circuit component.
- In Exercise 2 of the unit on Troubleshooting Digital Communications 1 circuits, you will troubleshoot each pulse modulation circuit to locate the problem caused by the activation of a fault switch.

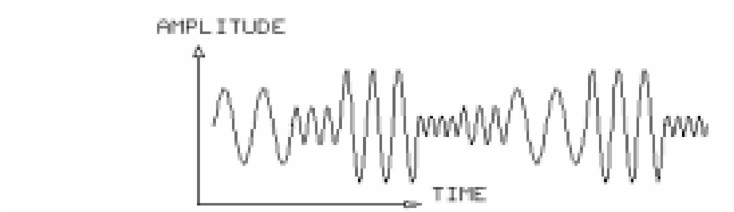
NOTES

UNIT 2 – PULSE-AMPLITUDE MODULATION

UNIT OBJECTIVE

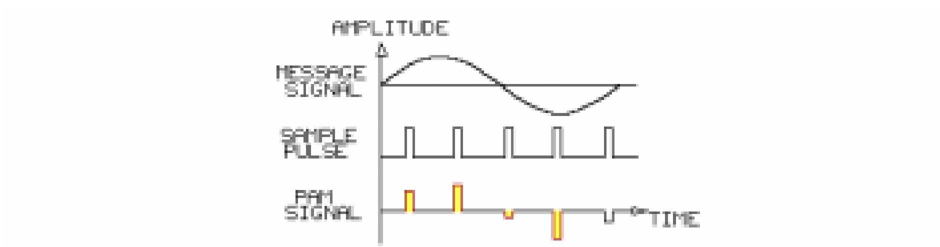
At the completion of this unit, you will be able to generate a pulse-amplitude modulation (PAM) signal and demodulate the PAM signal by using the PAM circuit block on the DIGITAL COMMUNICATIONS 1 circuit board.

UNIT FUNDAMENTALS



An analog message signal, representing voice for example, has continuous amplitude and frequency values that vary with time.

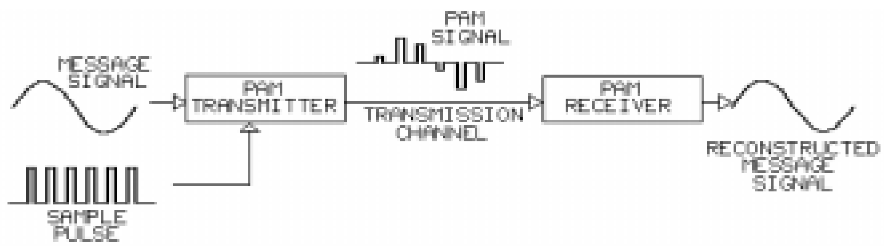
Analog communications systems transmit the complete analog waveform.



Instead of transmitting the analog waveform, it is possible to transmit pulses that represent the message signal's waveform.

Samples of the analog message signal can be taken at regular intervals.

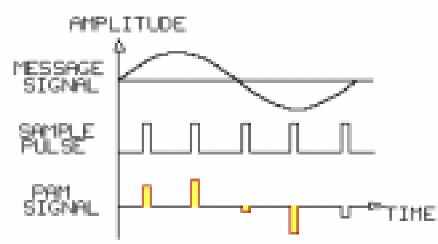
In **pulse-amplitude modulation (PAM)**, the amplitude of each pulse sample is proportional to the amplitude of the message signal at the time of sampling.



The PAM signal, rather than the analog message signal, is transmitted to a receiver. The PAM receiver demodulates (reconstructs or recovers) the PAM signal into the original analog message signal.

Two advantages of transmitting PAM signals rather than complete analog signals are:

- If the duration of the PAM pulse is small, the energy required to transmit the pulses is much less than the energy required to transmit the analog signal.
- The time interval between the PAM pulses may be filled with samples of other messages, which allows several messages to be transmitted simultaneously on one channel; this technique is called **time-division multiplexing (TDM)**.

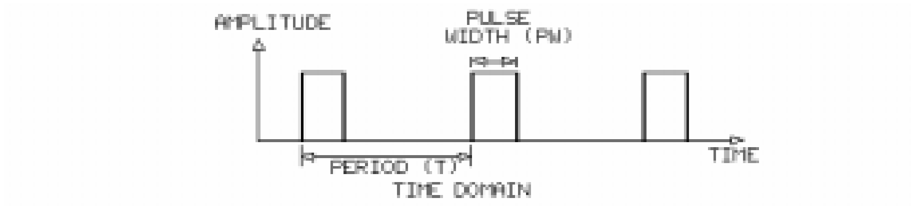


To demodulate the PAM signal into the original message signal, the PAM signal must be formed by a sample pulse frequency (f_s) greater than two times the maximum message signal frequency (f_m).

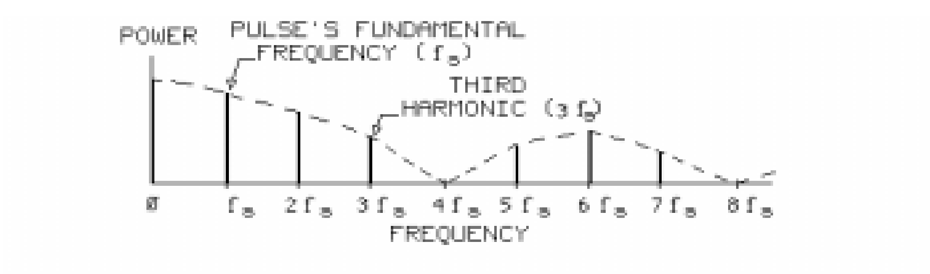
Two times f_m is the **Nyquist rate**. Nyquist rate = 2 x f_m

The Nyquist rate is the minimum theoretical sampling rate to form a pulse-modulated signal (PAM, PWM, PPM, PCM, etc.) that represents an analog signal.

However, to reconstruct a message signal that represents the original signal from a PAM signal, f_s must be greater than the Nyquist rate. $f_s > 2 \times f_m$



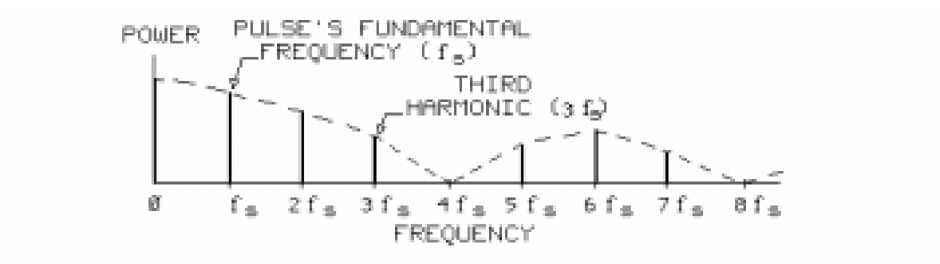
To understand why the pulse frequency (f_S) must be greater than two times the maximum message frequency (f_m), you must be familiar with the following pulse characteristics: **time domain.** You can observe these time domain characteristics of a pulse with an oscilloscope:



The frequency domain of a pulse signal is shown.

The frequency domain shows the power levels of the frequency components in a signal; a **spectrum analyzer** can be used to observe a signal's frequency domain. In addition to the pulse's **fundamental frequency** (f_S) being present, **harmonic frequencies** are also present.

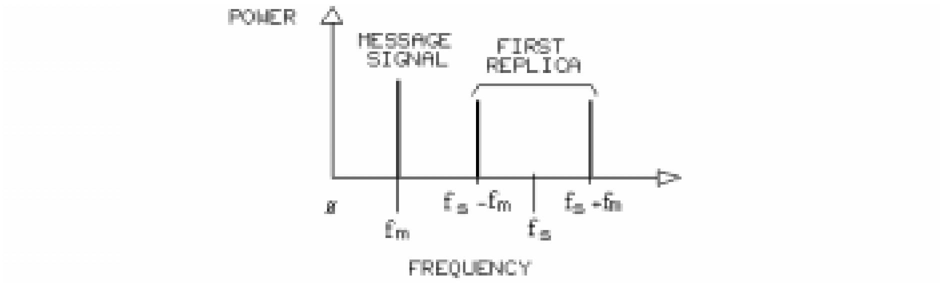
The harmonic power levels reach a minimum level at regular intervals that depend on the pulse's width and period.



The frequency spectrum of a PAM signal is shown. The message signal frequency (f_m) is present in addition to a number of **replicas**; only three are shown.

The replica frequencies are spaced around the sample pulse's fundamental (f_S) and harmonic frequencies by a plus and minus range equal to f_m.

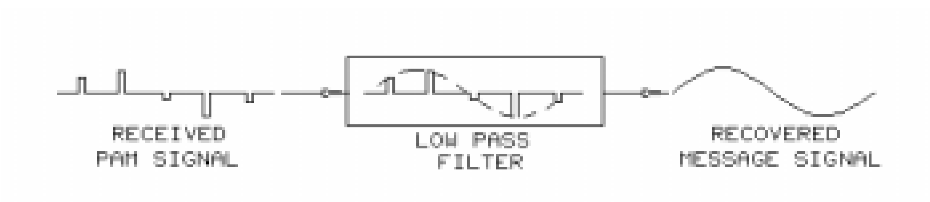
In a PAM signal, f_S and the harmonic frequencies (2 f_S and 3 f_S) may have zero power levels.



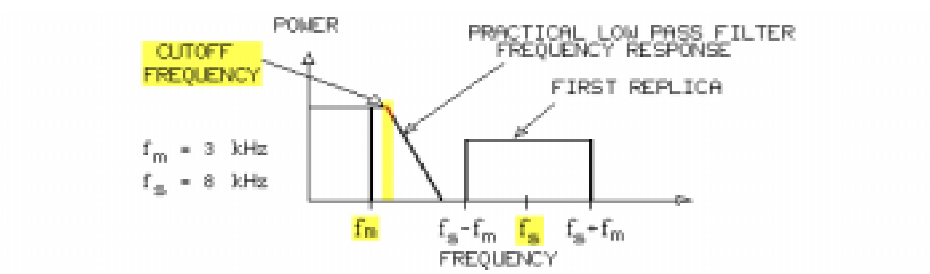
If the first replica is removed during demodulation, all the other replicas are also removed.

The frequency spectrums in the following screens will only show f_m and the first replica frequencies, f_s - f_m and f_s + f_m.

The separation, or guard space, between f_s - f_m in the first replica and f_m is the main concern in demodulating the PAM signal.



A low pass filter reconstructs the message signal by removing all the frequencies in the PAM signal except the message signal frequency.



The low pass filter must have a **frequency response curve** sharp enough to pass the maximum message signal frequency (f_m) and remove the lower sideband frequency (f_s - f_m) of the first replica.

Suppose a PAM signal is formed from a message signal with a maximum frequency (f_m) of 3 kHz and an 8 kHz sample signal (f_s).

NEW TERMS AND WORDS

pulse-amplitude modulation (PAM) - a modulation method in which the amplitude of each pulse sample is proportional to the amplitude of the message signal at the time of sampling.
time-division multiplexing (TDM) - a method of transmitting many digital message signals over the same line by assigning time slots that are synchronized on the transmitting and receiving ends.
Nyquist rate - the minimum theoretical sampling rate to form a pulse-modulated signal (PAM, PWM, PPM, PCM, etc). that represents an analog signal . The Nyquist rate equals two times the maximum message signal frequency.
time domain - a signal's amplitude as a function of time.
frequency domain - the amplitude of a signal's frequency components as a function of frequency.
duty cycle fraction (PW/T) - the ratio of the pulse duration (PW) to the period (T).
spectrum analyzer - an instrument that shows the relative power distribution of a signal as a function of frequency.
fundamental frequency - the principal frequency component of a waveform or pulse signal; the frequency with the greatest amplitude.
harmonic frequencies - a frequency component of a periodic waveform or pulse signal; the frequency of each harmonic is a multiple of the fundamental frequency.
replicas - frequencies that are spaced around the sample pulse's fundamental (fs) and harmonic frequencies by a plus and minus range equal to the maximum message signal frequency (fm).
frequency response curve - a graphical representation of a circuit's response to different frequencies within its operating range.
cutoff frequency - the frequency at which the output amplitude of a circuit (filter) is attenuated by -3dB (0.707 of the input).
sampler circuit - a circuit in which the output is a series of discrete values (pulses) representing the value of the input at sampling points in time.
natural PAM signals - a type of sampling in which the amplitude of each pulse follows the amplitude of the message signal at the sampling time.
flat-top PAM signals - a type of sampling in which the amplitude of each pulse remains constant during the sampling time.
sample/hold circuit - a circuit that holds the amplitude of each output pulse at a constant level between the input sampling times
staircase PAM signals - signals that are output from a sample/hold circuit; the amplitude of each output pulse remains at a constant level between the input sampling times.
aliasing - a condition in a PAM signal where the lower sideband frequency (fs-fm) is less than the maximum message signal frequency (fm); aliasing is also called fold-over distortion.

EQUIPMENT REQUIRED

In order to complete the following exercises, you will need:
F.A.C.E.T. base unit
DIGITAL COMMUNICATIONS 1 circuit board
Oscilloscope, dual trace
Signal Generator, sine wave

NOTES

Exercise 1 – PAM Signal Generation

EXERCISE OBJECTIVE

When you have completed this exercise, you will be able to:

- describe the process and circuits for generating PAM signals.
- describe the characteristics of PAM signals.
- demonstrate the effect of sampling pulse rates on PAM signals.
- calculate the power of a PAM signal.

You will use an oscilloscope to make observations and measurements.

DISCUSSION

- In pulse-amplitude modulation (PAM), a pulse signal periodically samples an analog message signal.
- The result is a train of constant-width pulses with amplitudes proportional to the message signal amplitude at the time of sampling.
- A sample signal causes an electronic switch to close and open.
- When the top of the PAM pulse follows the curvature of the message signal, the process is natural sampling. When a sampler closes, it produces a natural PAM sample.
- To produce flat-top PAM signals, a sample/hold circuit holds the amplitude of each pulse at a constant level.
- Staircase PAM signals are output from the sample/hold circuit; the sampler circuit converts them to flat-top PAM signals.
- The sample pulse frequency (f_s) must be more than the Nyquist rate to produce a PAM signal that clearly represents the message signal. When f_m is 3.5 kHz, f_s must be 8 kHz to produce a PAM signal that clearly represents the message signal.
- If PAM pulses are narrow, little power is required for transmission. Power in a PAM signal that represents a 2 kHz sine wave signal is about 15 percent of the sine wave power.
- The power (P_s) that a sine wave signal produces equals its rms voltage [$V_{rms\ (s)}$] squared, divided by the impedance (Z) where: $V_{rms\ (s)} = 0.707 \times V_{peak\ (s)}$
- Voltage $V_{rms\ (p)}$ of a pulse signal is pulse voltage (V_p) times the square root of the ratio of the PAM signal pulse width (PW) divided by its period (T). $V_{rms\ (p)} = V_p \times PW/T$.
- Theoretical rms value of a PAM signal formed from a cyclical signal is calculated as $V_{rms\ (p)} = PW/T \times V_{rms\ (s)}$
- The rms value [$V_{rms\ (p)}$] of the PAM signal for $V_{rms\ (s)}$ equals $V_{rms\ (p)} = PW/T \times V_{rms\ (s)}$; the theoretical power of a PAM signal formed from a sine wave is PW/T times the power of the sine wave (P_s): $P_p = PW/T \times P_s$
- Measure voltages and the PW/T fraction of the PAM pulses and calculate pulse rms values to experimentally determine the rms voltage of a PAM signal
- The rms voltage of a PAM signal equals the square root of the sum of the squares of the individual pulse rms values.

NOTES

Exercise 2 – PAM Signal Demodulation

EXERCISE OBJECTIVE

When you have completed this exercise, you will be able to:

- describe the demodulation of a PAM signal.
- demonstrate the effect of the PAM sampling rate on the recovered message signal.
- demonstrate the effect of the filter's cutoff frequency on the recovered message signal.

You will use an oscilloscope to make observations and measurements.

DISCUSSION

- A technique for demodulating PAM signals is low-pass filtering.
- A low-pass filter has the effect of averaging or "smoothing out" the received PAM signal.
- When the sample pulse frequency is greater than the Nyquist rate, a good reconstruction of the original message signal is output from a low-pass filter with the proper frequency response characteristics.
- To improve the recovery of the message signal, the PAM receiver may include a sample/hold circuit before the low-pass filter.
- A PAM signal and a sample pulse, synchronized to the transmitter sample pulse, are applied to a sample/hold circuit.
- The sample/hold circuit forms a staircase signal, which is applied to a low-pass filter.
- PAM frequency-domain signals help illustrate how a low-pass filter recovers, or reconstructs, the message signal.
- A PAM signal-spectrum contains both the message signal frequency and message-signal replica frequencies.
- A low-pass filter must remove all the replica frequencies in the PAM frequency spectrum to demodulate (reconstruct) a PAM signal.
- Proper demodulation of the PAM signal is possible when these conditions are met: (1) The message signal frequency is limited to some maximum value (f_m). (2) The pulse sample frequency (f_s) is greater than the Nyquist rate. (3) The filter cutoff frequency is greater than the maximum message frequency (f_m), and the first replica is sufficiently attenuated.
- If f_s is less than twice f_m, a condition known as **aliasing,** or fold-over distortion, occurs.
- Aliasing causes severe distortion of the recovered message signal.

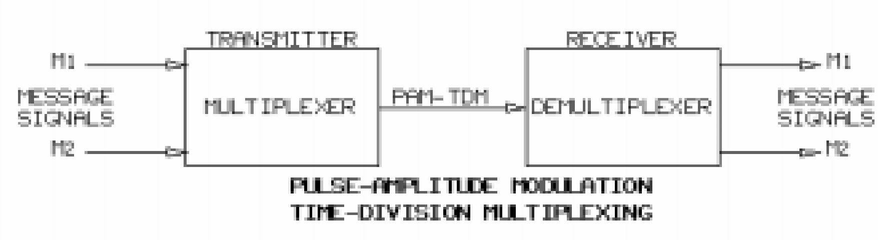
NOTES

UNIT 3 – PAM TIME-DIVISION MULTIPLEXING

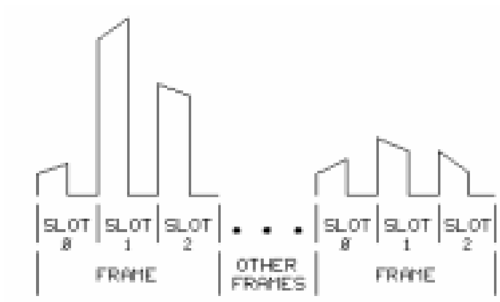
UNIT OBJECTIVE

At the completion of this unit, you will be able to describe how signals can be time multiplexed. You will use the PAM-TDM circuit block on the circuit board to multiplex and recover two PAM signals.

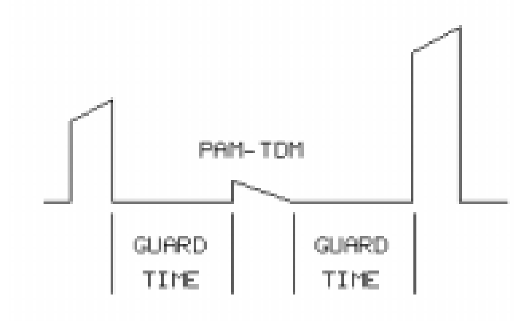
UNIT FUNDAMENTALS



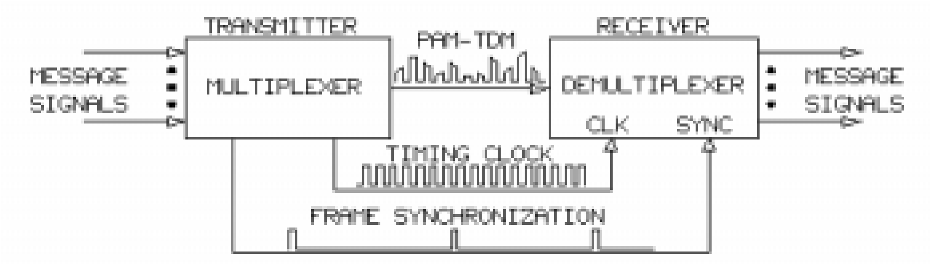
Time-division multiplexing (TDM) simultaneously sends multiple message signals over a single channel. The transmitter, or **multiplexer**, combines the message signals into a single pulse-amplitude modulated (PAM) TDM signal. The receiver, or **demultiplexer**, separates the PAM-TDM signal back into message signals.



The TDM channel is divided into time periods called **time slots**. Time slots have a fixed order that repeats once each **frame**. Each message signal is assigned to one of the time slots. As the frame repeats, one PAM sample from each message signal is placed in its assigned time slot.

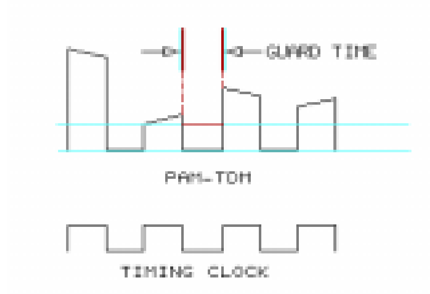


The space between the PAM-TDM pulses is called **guard time**. The guard time prevents interference between adjacent pulses.

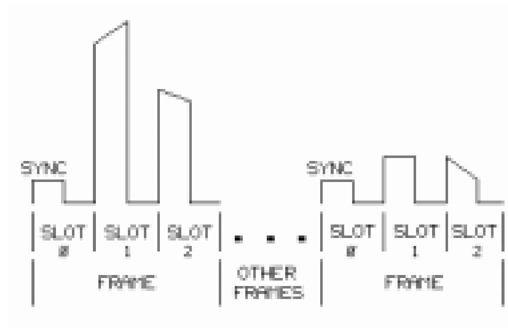


The transmitter **interleaves**PAM pulses into the TDM channel and provides timing and frame **synchronization** information for the receiver. The receiver uses a timing clock to identify when the PAM pulses are valid. Synchronization aligns the receiver with the transmitted TDM frames.

To allow TDM transmission over long distances, timing and control information is usually multiplexed along with the message signals. The receiver contains circuits that recover the timing and synchronization information from the PAM-TDM.



A timing clock can be extracted using the PAM pulses and the guard time spaces. A minimum PAM pulse height, or **pedestal**, ensures that a stable timing clock can be recovered from the PAM-TDM. The receiver uses the timing clock to identify when the PAM pulses are valid.



The PAM-TDM receiver also needs synchronization information to locate the time slots. The transmitter tags the beginning of each frame with a synchronization signal. The receiver uses the synchronization tag and the time slot assignment to identify each of the PAM signals.

The basic concepts of timing and synchronization are applicable to all TDM systems. Each type of TDM channel uses its own **protocol** that defines the channel's data format, timing, and synchronization. The format for a TDM channel is called the **physical layer protocol**.

NEW TERMS AND WORDS

demultiplexer - a device that separates two or more signals previously combined by a multiplexer and transmitted over a single channel (also called receiver).
time-division multiplexing (TDM) - two or more message signals sent or received over the same line by using a different time interval for each signal.
multiplexer - a circuit that combines multiple signals into a single output for later recovery (also called transmitter).
time slots - fixed periods of time assigned in a specific order within a repeating frame (see frame).
frame - a repeating period of time equal to the sampling period that is equally divided into smaller time assignments called time slots.
guard time - the time between pulses to prevent interference.
interleaves - the act of mixing items from one sequence ordered so that they alternate with items from another sequence.
synchronization - a means of aligning the transmitter and receiver to send and receive data at the correct times.
pedestal - a fixed height pulse inserted under a varying height pulse that guarantees a minimum pulse height.
protocol - required procedure or set of rules.
physical layer protocol - a format used to send information over a communication media.
state machine - a logic circuit that uses its present status to determine its next state.

EQUIPMENT REQUIRED

In order to complete the following exercises, you will need:
F.A.C.E.T. base unit
DIGITAL COMMUNICATIONS 1 circuit board
Multimeter
Oscilloscope, dual trace
Generator, sine wave

NOTES

Exercise 1 – PAM-TDM Transmission

EXERCISE OBJECTIVE

When you have completed this exercise, you will be able to:

- describe how PAM signals can be time-division multiplexed
- explain the synchronization used in the PAM-TDM circuit block
- explain the S1 and S2 sample pulse timing relationships

You will use an oscilloscope to verify the timing on the circuit board.

DISCUSSION

- The PAM-TDM transmitter combines message signals into a single channel by interleaving PAM samples. This process is called time-division multiplexing.
- The PAM transmitter contains two samplers, a slot counter, and an adder.
- The message signals are sampled during their assigned time slots.
- S1 samples the M1 message signal.
- S2 samples the M2 message signal.
- The S1 and S2 sample pulses are in time slots 1 and 2 respectively.
- The PAM pulse generated by S1 occurs when the output of the other sampler is at zero. The same is true for S2.
- The ADDER sums the SAMPLE 1, SAMPLE 2, and SYNC outputs.
- SAMPLE 1 and SAMPLE 2 provide PAM pulses in slots 1 and 2, respectively.
- SYNC (slot 0) is normally at ground generating an empty time slot.
- The transmit clock (CLKT) is subtracted to generate a pedestal.
- The ADDER receives one pulse at a time from its inputs and transmits these pulses one at a time.
- The SLOT COUNTER, or **state machine**, generates the S1, S2, and CLKT pulses.
- The SLOT COUNTER divides the transmit frame into three time slots.
- The SLOT COUNTER ensures that only one of its outputs is high at a time.

NOTES

Exercise 2 – PAM-TDM Reception

EXERCISE OBJECTIVE

At the completion of this exercise you will be able to

- explain how the clock signals are recovered from the PAM-TDM
- explain how the receiver time slots are synchronized with the transmitter time slots
- describe how the PAM-TDM signal is demultiplexed and how the message signals are recovered.

DISCUSSION

- The PAM-TDM receiver (demultiplexer) receives the time-division multiplexed PAM signal from the transmitter and recovers the individual message signals (M1 and M2).
- In order to recover M1 and M2, the receiver must recover the transmitted clock signals, synchronize its time slots with the transmitter time slots, demultiplex the PAM signals into individual staircase signals, and reconstruct the message signals.
- The SAMPLE HOLD 1 & 2 circuits demultiplex the PAM-TDM into staircase signals.
- FILTERs 1 & 2 reconstruct the message signals from the staircase signals.
- The SAMPLE HOLD circuits are sampled with the H1 and H2 signals, which are generated by the SLOT COUNTER.
- The receiver's sample/hold pulse (H1) occurs at the same time as the transmitter's sample pulse (S1).
- The M1 message signal sampled by S1 is recovered by the sample/hold circuit that receives the H1 pulse.
- M2 is sampled by S2 and is recovered using H2 during time slot 2.
- H1 is narrower than the PAM-TDM samples to ensure that only the PAM pulses are sampled
- The receiver's SLOT COUNTER generates the H1 and H2 signals from the timing and synchronization contained in the PAM-TDM signal.
- COMP- and COMP+ interface the SLOT COUNTER with the PAM-TDM signal.
- The SLOT COUNTER uses the information from COMP+ and COMP- to generate H1 and H2 at the correct times.
- The receiver uses the same time slot assignments as the transmitter.
- Time slot 0, which contains only a pedestal, is used to align or synchronize the receiver's time slots with the transmitter’s time slots.

- Time slots 1 and 2 contain the PAM samples from M1 and M2, respectively.
- The positive comparator (COMP+) indicates when the PAM-TDM signal is above 1.5V.
- The SLOT COUNTER uses the COMP+ output to detect when its time slots are out of synchronization with the transmitter's time slots.
- The transmitter ensures that time slot 0 contains only a pedestal (0V).
- The COMP+ block output will be active when a PAM-TDM pulse is above 1.5V.
- The COMP- block recovers the timing clock (CLKT) from the PAM-TDM signal.
- The SLOT COUNTER in conjunction with the phase-locked loop (PLL) synthesizes clocks that are phase-aligned with the transmitter's clocks.
- The transmitter sends the CLKT timing clock between 0 Vdc and -5 Vdc.

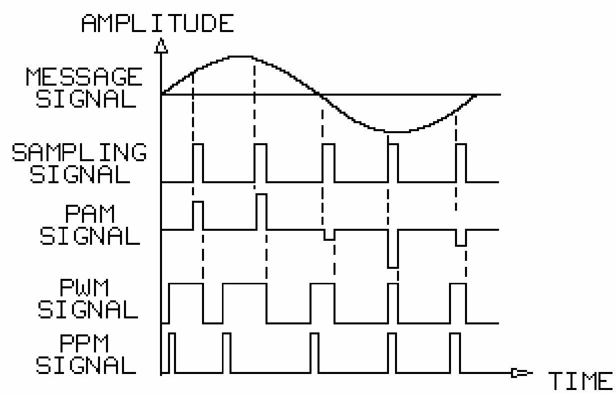
NOTES

UNIT 4 – PULSE-TIME MODULATION (PTM)

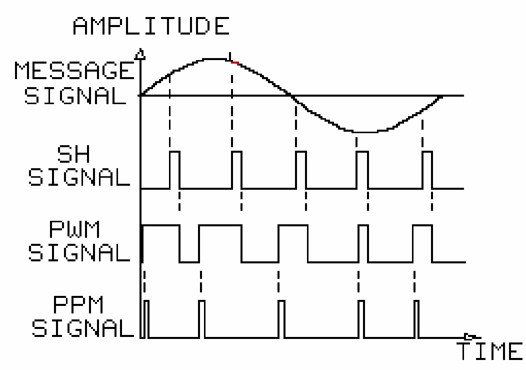
UNIT OBJECTIVE

At the completion of this unit, you will be able to describe how to convert a message signal into a pulse-time modulation (PTM) signal and how to demodulate a PTM signal by using the PTM circuit block on the DIGITAL COMMUNICATIONS 1 circuit board.

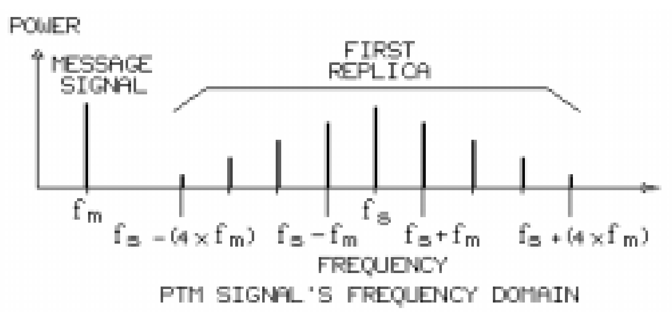
UNIT FUNDAMENTALS



There are different methods for pulse modulating a signal. Each method gets its name from the way pulses are varied to represent the message signal. Two common types of pulse modulation include pulse-amplitude modulation (PAM) and **pulse-time modulation (PTM)**.
Two types of PTM are **pulse-width modulation (PWM)pulse-position modulation (PPM)**. In PTM, the timing of the pulse is the variable, not the amplitude as in PAM.



PWM is sampling the message at regular intervals with a sample hold (SH) signal and producing pulse widths proportional to the message signal's amplitude at the time of sampling. A pulse edge is modulated (the edge position varies). The unmodulated edges are equally spaced. The PWM pulses have equal amplitudes. Pulse-position modulation (PPM) signals are generated from PWM signals. PPM signals are narrow, fixed-amplitude pulses with a varying position proportional to the message signal's amplitude at the time of sampling. The position of the PPM pulse is relative to the SH signal, and the distance between PPM pulses varies with the message signal's amplitude.



The frequency domain of a PTM signal is very complex. In addition to containing the message signal frequency, the PTM first replica may contain the sampling frequency (f_s) and several sidebands separated by a frequency equal to the message signal's frequency (f_m). Because of the additional lower sidebands in the first replica, the sampling frequency (f_s) must be higher than is necessary in PAM. In order to have sufficient spacing between the message and lowest sideband frequencies, a PTM signal should be sampled at a rate (f_s) several times the maximum message signal frequency (f_m).

A PTM signal is demodulated by low-pass filtering.

NEW TERMS AND WORDS

pulse-time modulation (PTM) - a type of pulse modulation where the timing of the pulse varies with the message signal's amplitude.
pulse-width modulation (PWM) - a type of PTM where the pulse width varies with the message signal's amplitude.
pulse-position modulation (PPM) - a type of PTM where the pulse position varies with the message signal's amplitude.

EQUIPMENT REQUIRED

In order to complete the following exercises, you will need:
F.A.C.E.T. base unit
DIGITAL COMMUNICATIONS 1 circuit board
Oscilloscope, dual trace
Signal generator, sine wave

NOTES

Exercise 1 – PTM Signal Generation

EXERCISE OBJECTIVE

When you have completed this exercise, you will be able to

- describe the characteristics of PTM signals.
- describe the circuits and signals used to generate PTM (PWM and PPM) signals.

You will use an oscilloscope to make observations and measurements.

DISCUSSION

- Two classes of pulse-time modulation (PTM) are pulse-width modulation (PWM) and pulse-position modulation (PPM).
- The PWM pulse width is directly proportional to the amplitude of the message signal.
- The PPM pulse position is directly proportional to the amplitude of the messge signal.
- On the circuit board, the SAMPLE/HOLD, RAMP GEN (ramp generator), ADDER, and COMPARATOR circuits generate the PWM signals.
- The PULSE LENGTH circuit converts an inverted PWM signal to a PPM signal.
- M1, the message signal, is a sine wave of about 5 $V_{pk\text{-}pk}$ at 1 kHz.
- The SH signal samples the message signal. In the PROCEDURE, you will use a 16 kHz SH signal.
- To understand how the PWM and PPM signals are generated, you must understand the relationship of the SH signal to the signals from the SAMPLE/HOLD, RAMP GEN, ADDER, COMPARATOR, and PULSE LENGTH circuits.
- The sample/hold circuit converts the message signal to a staircase PAM signal as the first step in generating a PWM signal.
- The sample/hold circuit generates the step at the positive edge of the SH signal.
- Every time the SH signal samples the message signal, a step is output from the sample/hold circuit.
- The step is equal to the message signal's amplitude at the time of sampling.
- The ramp generator forms a ramp signal synchronized to the SH signal.
- The SH signal causes the amplitude of the ramp signal to return to 0 Vdc and begin to increase at a fixed slope.

- The ramp signal's frequency equals the SH frequency.
- Because they are initiated by the SH pulses, step and ramp pulses occur at about the same time.
- The adder circuit sums the voltages of a step pulse and a ramp pulse.
- The average value of the adder's output matches the message signal waveform.
- The comparator compares the adder's output signal to a reference voltage.
- A PWM signal is output from the comparator.
- Because the average value of the adder's output has the message signal's waveform, the widths of the PWM pulses from the comparator will vary in proportion to the message signal's amplitude.
- The pulse length circuit forms the PPM signal from the inverted PWM output of the comparator.
- The PPM pulse occurs at the positive edge of the PWM signal.
- The distance from the PPM pulse to the SH pulse and the distance between PPM pulses are proportional to the message signal's amplitude.

NOTES

Exercise 2 – PTM Signal Demodulation

EXERCISE OBJECTIVE

When you have completed this exercise, you will be able to

- explain demodulation of PTM signals.
- describe the effect of message and sampling frequencies on the recovered message signal.
- describe the effect of the PPM pulse width on the recovered message signal.

You will use an oscilloscope to make observations and measurements.

DISCUSSION

- The PTM signal is demodulated by the low-pass filter.
- In most receivers, the PTM signal passes through a limiter circuit to eliminate noise before the message signal is reconstructed.
- In this unit, the PTM signal is noise free. The unit on "Channel Effects" explains and demonstrates how the limiter removes noise from a PTM signal.
- The analog prefilter is the first filter stage; it essentially removes frequencies higher than 16 kHz from the PTM signal and partially reconstructs and amplifies the message signal.
- A prefilter is necessary to recover a clear message signal from a PPM signal.
- PWM signals can be demodulated without a prefilter.
- A PTM signal's frequency domain is more complex than a PAM signal's frequency domain.
- The first replica in a PTM signal may contain the sampling frequency (f_s) and several sidebands separated by a frequency equal to the message signal's frequency (f_m).
- The analog prefilter has the following characteristics: attenuates 10 kHz signals by about 15 dB; attenuates frequencies higher than 14 kHz by over 20 dB; amplifies frequencies less than 3 kHz.
- When demodulating PPM signals, amplification of the message frequency is necessary because of the low power level of the narrow PPM pulses.
- The prefilter connects to the switched capacitor low-pass filter in the PAM circuit block.
- The filter completely removes the lower sideband and sampling frequencies that are in the prefilter's output signal.
- The reconstructed message signal is output from the filter.
- Because more than one lower sideband frequency may be present in a PTM signal, f_s should be several times the maximum message frequency (f_m) for a practical low-pass filter to output a clear reconstructed message signal.
- If the first replica contains four lower sideband frequencies separated by a frequency equal to f_m, f_s should be greater than four times f_m to properly reconstruct the message signal with a low-pass filter.

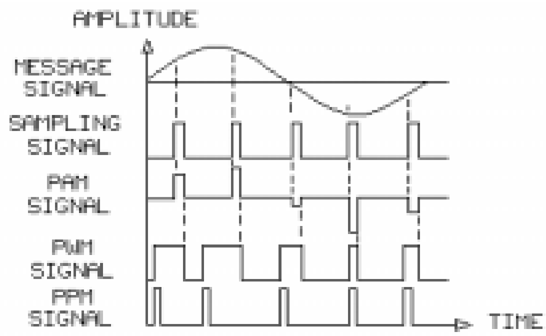
NOTES

UNIT 5 – PULSE-CODE MODULATION (PCM)

UNIT OBJECTIVE

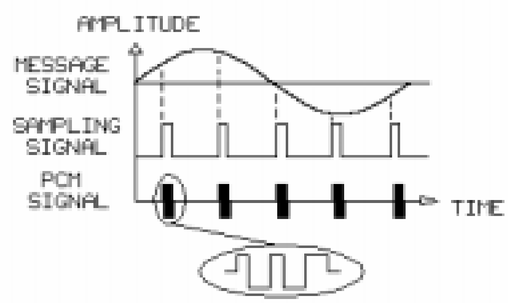
At the completion of this unit, you will be able to demonstrate pulse-code modulation (PCM) and time-division multiplexing (TDM) by using the PCM circuit block on the DIGITAL COMMUNICATIONS 1 circuit board.

UNIT FUNDAMENTALS



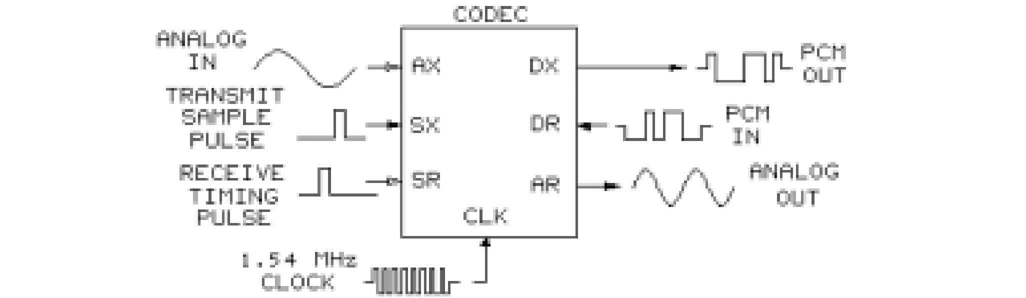
PAM, PWM, and PPM are pulse signals that represent the analog signal by a change in pulse amplitude, width, or position.

Although these signals are sequential pulses, they are not binary digital codes.



Pulse-code modulation (PCM) is sampling the voltage of an analog signal and converting each sample into an 8-bit serial digital code.

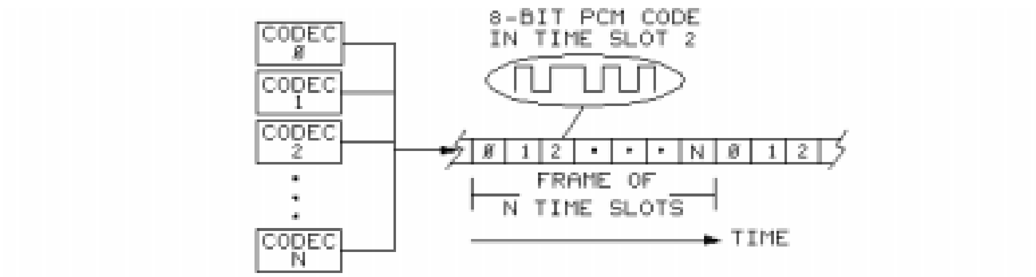
PCM signals are a series of 8-bit codes that represent the analog signal.



A **CODEC (encoder/decoder)** IC performs the analog-to-digital conversion (encoding) for the transmission of PCM signals.

The CODEC simultaneously performs the digital-to-analog conversion (decoding) on the received PCM signals.

Digital communications systems, such as telephone switching and transmission networks, use CODECs extensively for pulse-code modulation and demodulation of voice signals.



Time-division multiplexing (TDM) is the process by which many channels of PCM signals representing different messages are transmitted over the same line.

The outputs of each CODEC, which may be located in different communication devices, can be connected to a common transmission line.

From each CODEC, a series of 8-bit PCM signals representing each message signal is output.

The PCM signal's assigned time period on the transmission line is called a **time slot**.

A **frame** is a group of repeating PCM time slots, which are transmitted over the same line.

Some advantages of PCM transmission are:

- lower cost
- ease of multiplexing
- ease of switching
- less noise problems

Overall, the advantages outweigh the disadvantages. The industry trend is toward 100% digital communications networks.

NEW TERMS AND WORDS

pulse-code modulation (PCM) - a modulation process that produces binary serial codes for amplitude samples of the analog message signal.
CODEC (encoder/decoder) - a device that can simultaneously perform analog-to-digital conversion (encoding) and digital-to-analog conversion (decoding).
time-division multiplexing (TDM) - a process that transmits two or more message signals over the same line by using a different time interval for each signal.
time slot - period of time within a frame that is equal to the period of the samples.
frame - a period of time, equal to the sampling period, that is divided into smaller equal periods called time slots.
quantum - a number assigned to a particular quantized value.
companding - a compression and expansion process that improves the overall signal-to-noise ratio during PCM encoding and decoding
quantization - the process of converting a sampled amplitude into a numbered level called a quantum based on the number of bits in the PCM signal.
companding law - a compression and expansion process that improves the overall signal-to-noise ratio during PCM encoding and decoding a logarithmic relationship used for companding.
full duplex - a transmission system that permits communication signals to flow simultaneously in both directions
µ-law - a companding law used in North America and Japan.
A-law - a companding law used mainly in Europe.
local loop - the two-wire loop that connects the telephone set to the central office.
simplex - a transmission system that restricts communication signals to only one direction.

EQUIPMENT REQUIRED

In order to complete the following exercises, you will need:

F.A.C.E.T. base unit
DIGITAL COMMUNICATIONS 1 circuit board
Oscilloscope, dual trace
Signal Generator, sine wave

NOTES

Exercise 1 – Generation and Demodulation

EXERCISE OBJECTIVE

When you have completed this exercise you will be able to

- describe the process for converting analog signals to PCM signals.
- describe how PCM signals are demodulated.
- read and decode PCM signals.

You will verify your results by observing and measuring the signals with an oscilloscope.

DISCUSSION

- An analog message signal (AX) is encoded into a digital PCM signal (DX) by a CODEC.
- The acronym CODEC means encoder/decoder
- There is an 8 kHz transmit sample (timing) pulse (SX), which occurs one clock cycle before a time slot assigned to the transmitted PCM signal, that keys a CODEC to sample its message signal and generate a PCM signal.
- The CODEC decodes the received digital PCM signal (DR) into a recovered analog message signal (AR).
- There is an 8 kHz receive timing pulse (SR), which occurs one clock cycle before a time slot assigned to the received PCM signal that, keys a CODEC to decode an 8-bit PCM code.
- Encoding sections consist of a transmit filter, a sample/hold circuit, an ADC, a register, and a parallel-to-serial converter.
- The timing and control circuit synchronizes the CODEC's operation.
- SX is the 8 kHz transmit sample pulse.
- SR is the 8 kHz receive timing pulse, which usually occurs in a time slot different from SX.
- The analog message signal is applied to the transmit filter at AX.
- The sample/hold circuit samples the amplitude of the filtered analog signal 8000 times per second.
- The sample/hold circuit output is a staircase PAM signal.
- Quantization simplifies the conversion of the step voltage into a binary code.
- Overall signal-to-noise ratio is improved during quantization by companding (compressing/expanding).

- The higher signal-amplitudes are compressed during encoding and expanded during decoding.
- With S-shaped companding, smaller step-voltages receive higher quantum values than higher step voltages.
- A CODEC uses a companding law to determine the output code where the MSB is the sign bit. The MSB for a positive step voltage is 1. The MSB for a negative step voltage is 0. The remaining 7 bits designate one of the 128 quantum levels.
- An ADC encodes quantum values into an 8-bit parallel digital code, which is temporarily stored in a register.
- Parallel code from the output register is converted to an 8-bit serial digital code that is output, one bit at a time, by a 1.54 MHz clock.
- Operation of decoder circuitry is the reverse of the encoder.

NOTES

Exercise 2 – Time-Division Multiplexing

EXERCISE OBJECTIVE

When you have completed this exercise, you will be able to

- describe time-division multiplexing (TDM) of PCM signals.
- explain the relationship of the transmit sample (SX), receive timing (SR), and multiplexing PCM signals.
- demonstrate full-duplex transmission with PCM signals.

You will verify your results by observing and measuring the signals with an oscilloscope.

DISCUSSION

- Multiplexing is the method by which many channels of telephone conver- sations (message signals) are simultaneously carried over the same line (full duplex).
- At a central office, CODECs convert analog voice signals to PCM signals that are multiplexed and transmitted long-distance.
- PCM signals are easier to multiplex than analog signals.
- PCM signals are multiplexed using time-division multiplexing (TDM).
- PCM signals from a group of CODECs connect to a single transmission line.
- PCM signals from a single CODEC represent digitally coded analog message signal samples.
- Full duplex transmission of PCM signals can occur with TDM.
- PCM signals from CODECs are assigned a time slots on the transmission line.
- After the PCM signal from the last CODEC (CODEC n) is placed in its time slot, the TDM process repeats with a PCM signal from the first CODEC (CODEC 0).
- During a time slot, the PCM signal appears only one byte at a time on the transmission line.
- A group of repeating time slots is called a frame.
- In a decoder that receives a PCM signal, the SR (receive timing) pulse occurs at the same time an SX pulse enables message signal sampling.
- Because voice PCM signals have a relatively low frequency (8 kHz), many PCM channels can be sent in series on one high-frequency digital transmission line.
- The 1.536 MHz clock rate used for CODECs in the PCM circuit block is about the same as the clock rate for commercial 1.544 MHz T1 carrier systems used for digital networks of 50 miles or less.

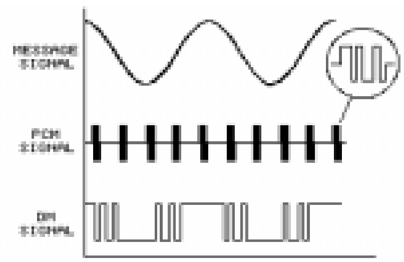
NOTES

UNIT 6 – DELTA MODULATION (DM)

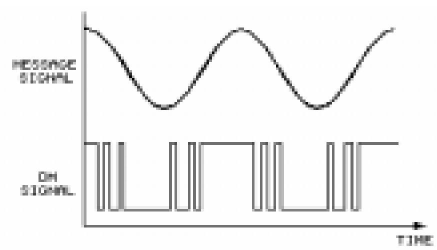
UNIT OBJECTIVE

At the completion of this unit, you will be able to demonstrate delta modulation and demodulation by using the DELTA circuit block on the DIGITAL COMMUNIATIONS 1 circuit board.

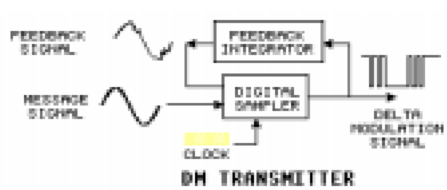
UNIT FUNDAMENTALS



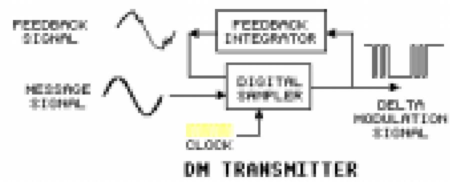
Like PCM, **delta modulation (DM)** is a digital encoding process. PCM converts each message-signal amplitude-sample into an 8-bit code. DM converts the change in the message-signal amplitude to a 1-bit code: logic 1 for an increase, and logic 0 for a decrease.



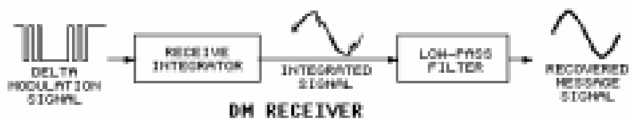
DM can be considered as a 1-bit differential pulse-code modulation system. A logic 0 means that the message signal's amplitude is decreasing. A logic 1 means that the message signal's amplitude is increasing.



Common DM sampling frequencies (CLOCK) are from five to ten times the Nyquist rate. Because the DM code is 1 bit, the bit rate equals the sampling frequency. The normal 32 kHz sampling frequency can be changed to 8 kHz, 16 kHz, or 64 kHz by CM switches.

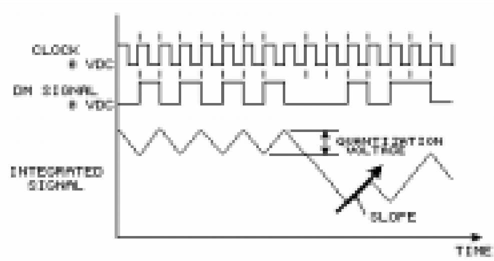


A **digital sampler** generates the DM logic state. The digital sampler is a compare/latch circuit that compares the message signal's amplitude with a reconstructed amplitude from a **feedback integrator**.



The transmitted DM signal is a continuous series of bits indicating an increase or decrease in the message signal's amplitude. The DM receiver contains a **receive integrator** (similar to the transmitter's feedback integrator) and a low-pass filter. The receive integrator produces a sawtooth reproduction of the message signal from the DM code.

The low-pass filter smooths out the sawtooth reproduction to produce the recovered message signal. Delta modulation decoding systems are neither distortion free nor noise free. The integrated signal can be distorted by **quantization errors**, which cause a distorted recovered message signal.

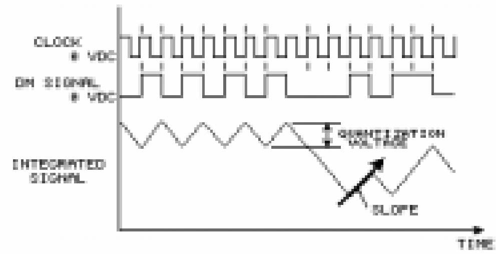


The **quantization voltage** is the amplitude of the integrated signal for one lock (sampling) period. The sampling rate and integrator slope determine the quantization voltage, which affects the quantization errors.

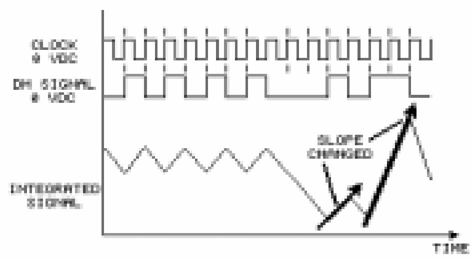
Quantization errors include:

- slope overload
- quantization noise
- idling noise

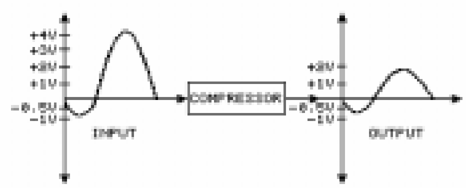
You will observe these quantization errors in Exercise 2.



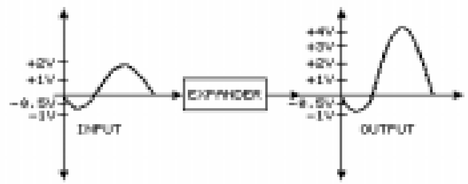
The DM transmitter described produces **linear DM**, because it has a fixed slope. **Continuously variable slope delta modulation (CVSD)** is another DM method. CVSD monitors the signal for signs of distortion.



When the quantization errors are detected, the slope of the feedback integrator output is increased or decreased to reduce the errors. CVSD is widely accepted as being potentially a very technically attractive method of digital transmission. The dynamic input signal range of a delta modulation system is limited by slope overload and quantization noise. This dynamic input range can be increased by **companding**.



Companding is accomplished by using a compander, which consists of a **compressor circuit** and an **expander circuit**. During compression, the compressor attenuates the higher amplitude portions of a signal more than the lower amplitude portions.



During expansion, the expander amplifies the larger signal amplitudes more than the lower signal amplitudes. Companding improves the system performance since a larger range of input signals can be transmitted and received.

NEW TERMS AND WORDS

delta modulation (DM) - a 1 bit encoding process that outputs a logic 1 bit for an increase in the message signal's amplitude and a logic 0 bit for a decrease.
digital sampler - a circuit composed of a comparator and D-type flip flop that compares the message signal amplitude with the amplitude of an integrated feedback signal to output a DM signal.
feedback integrator - an integrator circuit in a DM transmitter that integrates the DM signal and outputs a sawtooth reproduction whose amplitude is compared to the message signal's amplitude.
receive integrator - an integrator circuit in a DM receiver that integrates the DM signal and outputs a sawtooth reproduction to a low-pass filter.
quantization errors - DM signal errors (slope overload, idling noise, and quantization noise) that cause distortion in the recovered message signal.
quantization voltage - the amplitude of the integrated signal for one clock (sampling) period.
linear DM - a DM system that has a fixed quantization voltage; the slope of the integrated signal is constant.
continuously variable slope delta modulation (CVSD) - a DM system that detects quantization errors and changes the slope of the integrated signal to reduce the errors.
companding - a process that contains a compressor circuit and an expander circuit to improve the dynamic range of a DM system.
compressor circuit - a circuit, usually in a transmitter, that attenuates the higher amplitude portions of a signal more than the lower amplitude portions.
expander circuit - a circuit, usually in a receiver, that amplifies the higher amplitude portions of a signal more than the lower amplitude portions.

EQUIPMENT REQUIRED

F.A.C.E.T. Computer-Interface
Base Unit
DIGITAL COMMUNICATION 1 Circuit Board
Signal Generator, (sine/square)
Multimeter
Oscilloscope, dual trace

NOTES

Exercise 1 – DM Transmitter

EXERCISE OBJECTIVE

When you have completed this exercise you will be able to

- describe how DM signals represent the analog message signal.
- demonstrate the operation of a digital sampler.
- describe the function and operation of a feedback integrator.
- explain how clock frequency affects the DM bit rate and the feedback integrator's output signal.

You will verify your results by observing and measuring the signals with an oscilloscope.

DISCUSSION

- To produce a DM signal, a digital sampler compares the message signal's amplitude with the amplitude of a feedback signal, which is a partially reconstructed message signal from a feedback integrator.
- A DM sample of the message signal's increase or decrease contains 1 bit.
- A comparator and a D-type flip flop constitute the digital sampler in the DELTA circuit block.
- The feedback integrator is the integrate 1 circuit.
- The comparator compares the amplitude of the message and feedback signal amplitude to determine if message signal amplitude is increasing or decreasing.
- Ideally, the feedback amplitude should be a close approximation of what the message-signal amplitude was at the previous clock pulse.
- When the message signal is greater than the feedback signal, the comparator output is high, which indicates an amplitude increase.
- When the feedback signal is greater than the message signal, the comparator output voltage is low.
- The comparator output is the D input of the flip flop.
- The 32 kHz CLK signal, the sampling signal, is the flip flop's CLK input.
- If the comparator output (flip flop D input) is high at the rising edge of the CLK signal, the Q output of the flip flop becomes logic 1.
- If the comparator output is low at the rising edge of the CLK signal, the Q output of the flip flop becomes logic 0.
- The flip flop's Q output is latched to a logic state until the next rising edge of the signal occurs?

- The logic state of the flip flop Q output will remain the same or change at every rising edge of the clock signal depending upon the voltage at the D input.
- The flip flop Q output is the delta modulation (DM) signal.
- The INTEGRATE 1 circuit integrates the DM signal to produce the feedback signal.
- The REF voltage determines how closely the feedback signal resembles the message signal. Adjust the REF voltage input with the POSITIVE SUPPLY voltage potentiometer on the base unit.
- An RC time constant sets the positive and negative slope of the integrator output voltage.
- Each logic 1 DM signal causes the integrator output voltage to increase by a fixed amount.
- A logic 0 DM signal causes the integrator output voltage to decrease.
- Baud is the number of signal-changes per second and is equal to the reciprocal of the period of the smallest signaling element.
- In DM, baud equals the number of bits per second.
- The DM signal is output from the digital sampler at a bit rate equal to the frequency of the clock signal.
- If the clock frequency to the digital sampler is 32 kHz, the bit rate in baud is 32,000.

NOTES

Exercise 2 – DM Receiver and Noise

EXERCISE OBJECTIVE

When you have completed this exercise you will be able to:

- describe how DM signals are demodulated.
- identify and explain the cause of slope overload, idling noise, and quantization noise.
- describe how companding improves the dynamic range of DM.

You will verify your results by observing and measuring the signals with an oscilloscope.

DISCUSSION

- Compare the M1 message signal to the INTEGRATE 2 and recovered message signals.
- The delta modulation receiver contains simple hardware for decoding the DM signal.
- Only a receive integrator and low-pass filter compose the receiver.
- On the circuit board, the receive integrator is the INTEGRATE 2 circuit, and the low-pass filter is the FILTER 1 circuit in the PAM-TDM circuit block.
- The receive integrator converts the DM signal into an integrated signal that approximates the original intelligence signal.
- The low-pass filter recovers the message signal from the integrated signal by removing the high frequency components.
- Clock and synchronization signals are not required for the DM receiver to recover the message signal?
- Delta modulation communication systems are not distortion or noise free.
- The recovered message signal can be distorted by quantization errors including slope overload, quantization noise or granular noise, and idling noise
- The exercise procedure explains and demonstrates each of these quantization errors.
- Companding, which permits an increase in the dynamic range (greater change of amplitude) of the DM message signal and reduces quantization errors, is discussed and demonstrated in this procedure.
- Companding permits greater changes in the amplitude of the DM message signal?

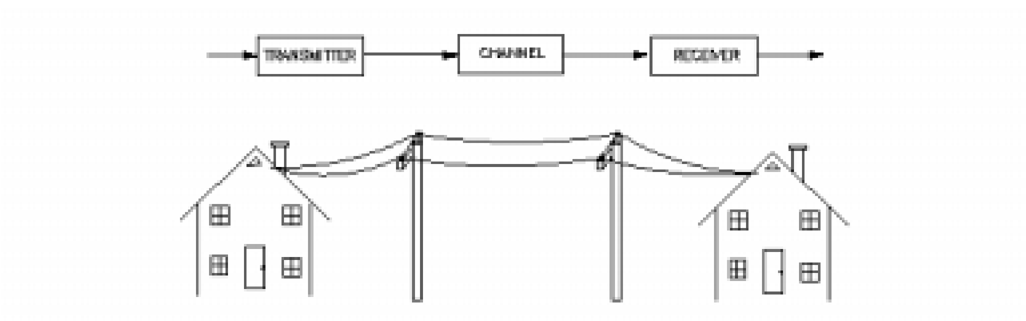
NOTES

UNIT 7 – CHANNEL EFFECTS

UNIT OBJECTIVE

At the completion of this unit you will be able to demonstrate how a communications channel affects pulse and digital signals.

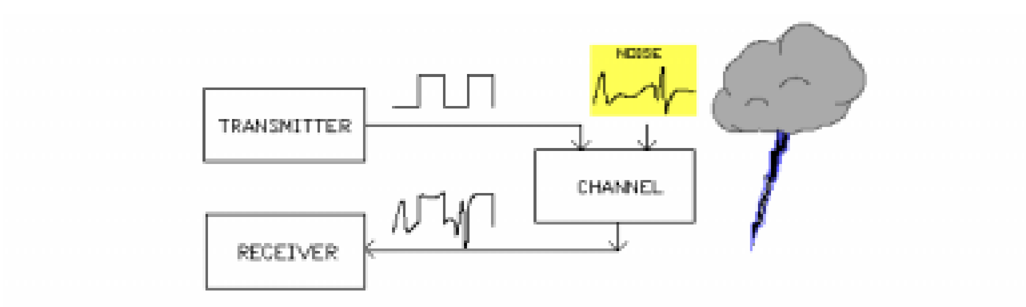
UNIT FUNDAMENTALS



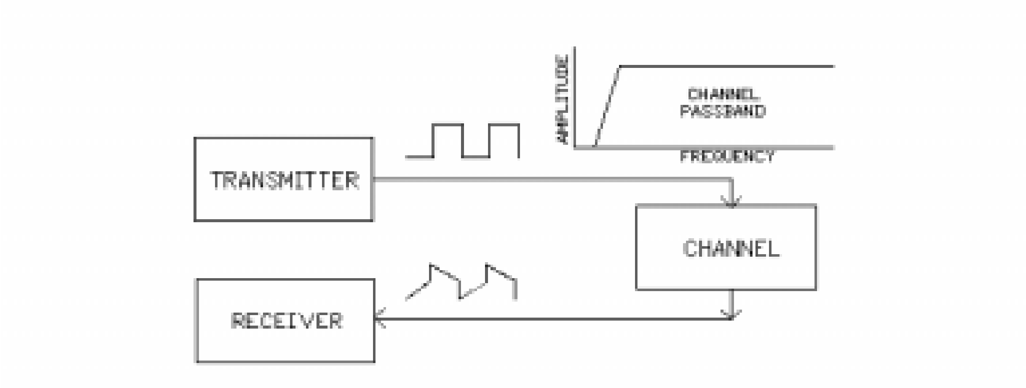
The channel is a path used to pass signals between the transmitter and receiver. A **digital communications channel** carries digital signals (pulses) which represent the message being transferred. A digital channel can use any **media** to carry the pulses including: twisted wire, fiber optics, or coaxial cable. All channels distort the signals they carry.

Channel characteristics that can cause distortion are:

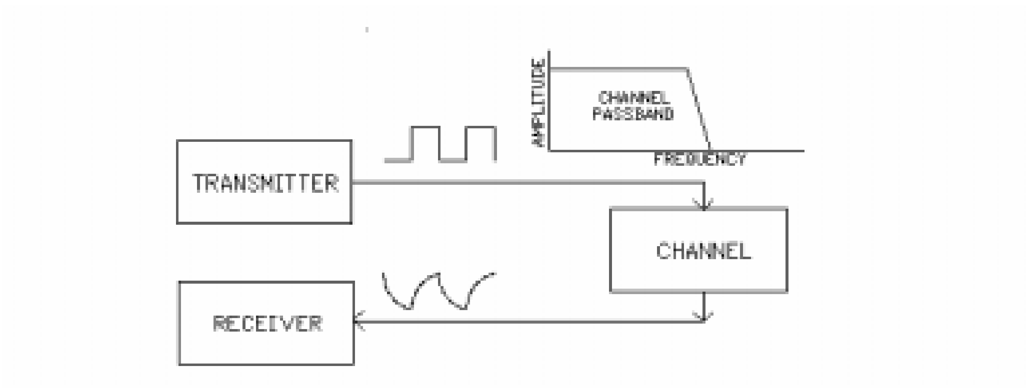
- noise susceptibility
- channel bandwidth
- channel timing variations
- channel resonance



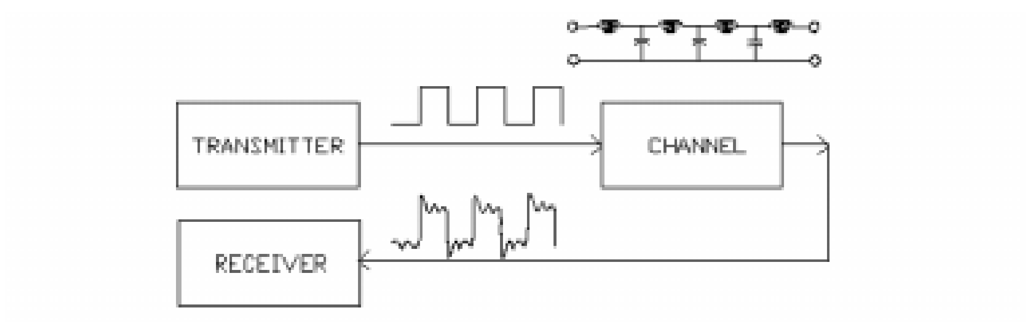
Noise is an interference signal which the channel adds to the transmitter signal. The noise signal affects the amplitude and possibly the logic state of the receiver's input pulses.



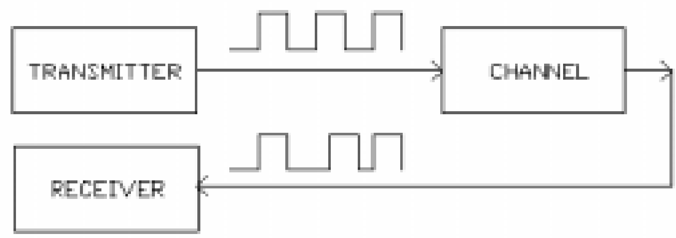
A channel with a poor low-frequency characteristic can affect the top and bottom of a pulse. The pulses do not maintain a stable voltage level, making them difficult to detect.



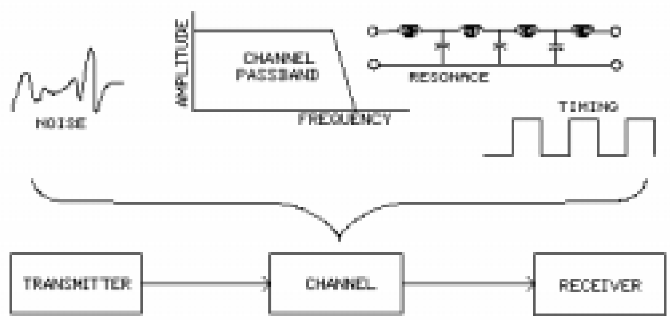
A channel with a poor high-frequency characteristic affects the pulse rise and fall times. Limited bandwidth increases the time a pulse needs to attain the amplitude required by the receiver.



The channel can contain reactances that will oscillate. The oscillations cause **ringing** at the channel's output.



If variations exist in the channel timing characteristics, the channel's time delay will be inconsistent. Timing variations affect the received signal by delaying it for varying amounts of time; this is called **jitter**. Channel noise and bandwidth limitations will also cause jitter.



Matching transmitter signal characteristics to channel characteristics minimizes channel effects.

NEW TERMS AND WORDS

digital communications channel - a path used to carry digital signals.
media - the physical path used to transmit the signal to the receiver.
ringing - decaying oscillations that occur after a signal transition.
jitter - abrupt signal changes.
significant frequency components - frequencies that, when removed, make a substantial difference in the resulting waveform and represent most of a signal's power.
harmonics - signals with frequencies that are integral multiples of the fundamental frequency.
Intersymbol interference - when a given pulse (symbol) is contaminated with energy from previous pulses.
random noise - as used in this units, an interference signal which is instantaneously unpredictable.
signal-to-noise ratio - the ratio of desired signal power to noise power; expressed in decibels.
pseudo random - a repetitive pattern that appears to be unpredictable.
symmetrical - mirrored about a line, balanced.

EQUIPMENT REQUIRED

F.A.C.E.T. base unit
DIGITAL COMMUNICATIONS 1 circuit board
Multimeter
Oscilloscope, dual trace
Generator, sine wave

NOTES

Exercise 1 – Channel Bandwidth

EXERCISE OBJECTIVE

When you have completed this exercise you will be able to:

- Measure the bandwidth of the CHANNEL SIMULATOR.
- Demonstrate how a pulsed signal is affected by a channel's passband.
- Describe the effect of channel resonance and ringing.

You will verify your results by observing and measuring signals with an oscilloscope.

DISCUSSION

- Communication systems consist of three main parts:
  transmitter
  transmission medium (channel)
  receiver.
- The transmitter's signal must pass through the channel to reach the receiver.
- For a pulse to be detected at the receiver, the significant frequency components of the pulse must pass through the channel.
- A pulsed signal consists of a large number of frequency components (harmonics) located at multiples of the fundamental pulse frequency ($f_o = 1/T$).
- Practical channels have a limited bandwidth.
- Frequency components inside the channel's passband are passed to the receiver.
- The elimination of the higher frequency components changes the shape of the pulse delivered to the receiver.
- The formula for the bandwidth (bw) of a single-pole RC low pass filter is:
- **bw = $1/(2\pi RC)$**
- One RC time constant ($\tau = RC$) is required for the circuit's output to reach 63% of the input voltage.
- The mathematical relationship between a single-pole filter's bandwidth and its time constant.
  bw = $1(2\pi\tau)$
- Rise time (t_r) is the time required for a pulse to rise from 10 to 90 percent of its final amplitude.
- By substituting $t_r/2$ for τ, a practical relation between bandwidth and rise time can be determined. bw = $1/(\pi t_r)$
- The minimum rise time (t_r) for a given channel bandwidth (bw) can be approximated using the formula: $t_r = 1/\pi$bw
- For a pulse to reach its full amplitude, the pulse width must be greater than the rise time.
- The channel bandwidth limits the rise time and therefore also determines the minimum pulse width.

NOTES

Exercise 2 – Channel Noise

EXERCISE OBJECTIVE

In this exercise you will explore the effect of noise on pulse and digital circuits by:

- Measuring signal to noise ratio.
- Demonstrating the effect of noise on PAM signals.
- Demonstrating the effect of noise on PCM signals.
- Comparing the noise sensitivity of PAM and PCM signals.

DISCUSSION

- A signal passing through a channel can be affected by either internal or external noise sources.
- Two examples of external noise sources are lightning and solar radiation. The transmission media determines how much of the external noise will be coupled to the channel.
- Two examples of internal noise sources are circuit switching and thermal noise from circuit components.
- The channel adds noise to the transmitter's signal.
- The channel output signal is a composite containing both the random noise and the transmitter's pulsed signal.
- If the noise power is large in comparison to the transmitter signal power, the receiver will not be able to reconstruct the message.
- The ratio of the signal power to noise power is an indication of how easily the pulses can be detected.
- The signal-to-noise ratio [SNR(dB)] is the standard comparison between the signal average power (P_s) and the noise average power (P_n). [$SNR(dB) = 10 \times \log(P_s/P_n)$]
- An alternate form of the SNR(dB) formula uses the square of the voltage ratio to find the power ratio.
- Power can be expressed as rms voltage squared divided by the resistance. $P = V^2/R$
- Substituting (V^2/R) in place of P in the SNR(dB) formula:
  $SNR(dB) = 10 \times \log((V_s^2/R)/(V_n^2/R))$

- The resistance can be removed from the power ratio by multiplying both (V_s^2/R) and (V_n^2/R) by R. SNR(dB) = 10 x log(V_s^2/ V_n^2)
- The square of the rms voltage ratio can be found by multiplying the log of the ratio by 2. log(y^2) = 2 x log(y)
- The PAM, PCM, and NOISE signals are not simple sine waves.
- The formula for finding the rms voltage of a sine wave does not apply to pulsed waveforms.
- The PCM and NOISE signals can be analyzed using a standard formula for pulses of constant amplitude. $V_{rms} = A \times (T_{pw}/T)^{1/2}$
- The PAM signal may have several pulses. Each pulse has its own pulse amplitude. The pulses are analyzed separately and summed using the square root of the sum of the squares. $PAM_{rms} = (V_{rms1}^2 + V_{rms2}^2 + \ldots)^{1/2}$

NOTES

APPENDIX A – SAFETY

Safety is everyone's responsibility. All must cooperate to create the safest possible working environment. Students must be reminded of the potential for harm, given common sense safety rules, and instructed to follow the electrical safety rules.

Any environment can be hazardous when it is unfamiliar. The F.A.C.E.T. computer-based laboratory may be a new environment to some students. Instruct students in the proper use of the F.A.C.E.T. equipment and explain what behavior is expected of them in this laboratory. It is up to the instructor to provide the necessary introduction to the learning environment and the equipment. This task will prevent injury to both student and equipment.

The voltage and current used in the F.A.C.E.T. Computer-Based Laboratory are, in themselves, harmless to the normal, healthy person. However, an electrical shock coming as a surprise will be uncomfortable and may cause a reaction that could create injury. The students should be made aware of the following electrical safety rules.

1. Turn off the power before working on a circuit.
2. Always confirm that the circuit is wired correctly before turning on the power. If required, have your instructor check your circuit wiring.
3. Perform the experiments as you are instructed: do not deviate from the documentation.
4. Never touch "live" wires with your bare hands or with tools.
5. Always hold test leads by their insulated areas.
6. Be aware that some components can become very hot during operation. (However, this is not a normal condition for your F.A.C.E.T. course equipment.) Always allow time for the components to cool before proceeding to touch or remove them from the circuit.
7. Do not work without supervision. Be sure someone is nearby to shut off the power and provide first aid in case of an accident.
8. Remove power cords by the plug, not by pulling on the cord. Check for cracked or broken insulation on the cord.