Experiment
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Pulse Code Modulation and Time Division Multiplexing

Hardware Experimentation

Objectives

PCM and time-division multiplexing are widely used in modern telecommunication networks. The main objectives of this experiment are to use to DCB board in order to:

1. Gain a good understanding and hands-on experience with Pulse Code Modulation
2. Learn about non-uniform quantization and its applications (e.g., in telephony)
3. Understand and experiment with the concept of Time Division Multiplexing.

Pre-Lab Work

1. Read the relevant material in your textbook.
2. Describe the use of PCM and TDM in modern telephony: give the basic architecture of modern telephony networks from the user phone to the central office switch, discuss T1/E1 multiplexing, etc.
3. Discuss and compare the advantages & disadvantages of TDM vs. FDM (frequency-division multiplexing).

Overview

Pulse-code Modulation (PCM), like PAM, is a digital communication technique that sends samples of the analog signal taken at a sufficiently high rate (higher than the Nyquist rate). In addition, PCM differs than PAM in that it quantizes the samples by constraining them to only take a limited number of values, and then converts each value into a binary string of bits that are transmitted on the communication line. Typically, in digital telephony where PCM is widely used, the sampling rate is 8 kHz (higher than twice the voice band), and the quantization uses 256 levels (i.e., each sample is mapped into an 8-bit PCM code).

In practice, PCM is typically combined with Time Division Multiplexing (TDM), which is the process of combining many PCM signals representing different messages and transmitting them over the same channel on a time-sharing basis. Each PCM signal is assigned a time period called a slot on the transmission line, and slots are arranged in groups called frames. The main advantages of PCM transmission are: lower cost, ease of multiplexing and switching, and better noise immunity. Its main disadvantage is the stringent timing and synchronization requirements. Nowadays, PCM-TDM systems form the backbone for all digital telephony networks worldwide (refer to your textbooks for more details).

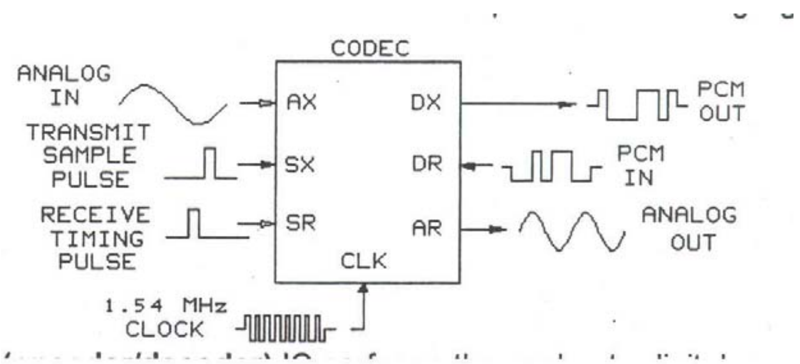
In this experiment, you will be using the PCM block in the DCB board, which is centered on a pair of voiceband CODECs (a codec is a Coder-Decoder) for PCM transmission and reception. These CODECs are IC (integrated circuit) chips that incorporate the major PCM-TDM functionalities. The basic operation of the CODEC is described below.

First, the analog signal at the AX input port is converted into a digital PCM signal at the DX output port. There is a timing signal SX giving an 8 kHz transmit pulse that occurs one clock cycle before the assigned time slot. Each pulse enables the codec to sample/encode the message signal to be transmitted in the following time slot.

At the receiver codec, the PCM signal at the DR port is recovered into an analog signal at the AR port. Similar to the transmitter codec, the receiver has an 8 kHz clock signal SR that enables the operation of the codec. In order for the receiver to decode a transmitted PCM code in a given time slot, the SX and SR pulses must be synchronized.

The transmission rate between codecs, i.e., the PCM line rate, is governed by a 1.54 MHz clock (typically referred to as T1 rate). Notice that this is much faster than the bit stream generated from a single user (which is 8bitsx8 kHz, or 64 kbps). This is because the PCM line actually carries a number of PCM signals (a T1 line corresponds to 24 PCM signals) in a TDM fashion, as explained above, and therefore needs to run at a much higher speed than a single 64 kbps PCM stream.

The actual architecture of the codecs used in the DCB board comprises many circuit blocks. On the transmitter (encoder) side, there is a voice-band anti-aliasing filter (limited to [0.2 kHz, 3.5 kHz]), followed by an 8 kHz sample & hold circuit, an ADC (analog-to-digital converter) that implements 8-bit quantization and compressor (μ-law or A-law), and finally a parallel-to-serial converter and output register. On the receiver (decoder) path, there is an input register with serial to parallel conversion, a sample & hold circuit, an expander (to remove the transmitter compression), and a receiver interpolation filter.



Lab Work

1) Part 1: PCM Signal Modulation & Demodulation

- Use the external function generator to apply a sinusoidal signal with frequency 1 kHz and amplitude 2 V peak-to-peak to Input AX of CODEC 1.
- Connect CODEC 1 to CODEC 2 with a two-post connector. CODEC 1 is the transmitter, and CODEC 2 is the receiver.
- Use the oscilloscope to simultaneously observe the AX signal of CODEC 1 and the AR signal of CODEC 2. Is the demodulation working properly?
- Observe the signal DX of CODEC 1 (or similarly, DR of CODEC 2) on the oscilloscope. What does this signal correspond to?
- If you keep increasing the amplitude of the input AX signal, what type of distortion do you get? What is the reason for that?
- Now, try to change the input frequency for AX. If the frequency is outside the range [0.2kHz, 3.5kHz], what happens to the received AR signal? What is the reason for that?

2) Part 2: PCM Timing Signals

- Disconnect the external function generator, and use a 2-post connector to apply the signal M1 to Input AX of CODEC 1. Record its frequency and amplitude.
- Observe the input sampling signal SX of CODEC 1 on the oscilloscope. Measure its period and frequency. What is the role of this SX signal?
- Check the SR signal of CODEC 2 on the oscilloscope. Is it in synch with the SX signal? What is the reason for that (i.e., will PCM decoding work if not)?
- Observe the relative timings of the signal SX and DX of CODEC 1 (same for RX and DX of CODEC 2). Apply these signals to channels 1&2 of the oscilloscope, and record your observations. When does DX start relative to SX? Explain why?
- Now, observe the characteristics of the DX signal (or RX) on the oscilloscope (which corresponds to PCM codes). How many bits per PCM code are there? What is the duration of one PCM code? Try displaying a few different code words on the oscilloscope. Are they having the same string of bits? Why?
- Using the oscilloscope, try to measure the PCM stream bit rate. Compare this with the theoretical bit rate you expect to see.

3) Part 3: Companding and Non-Uniform Quantization

- Launch the WinFACET software, and go through the material covering the overview and description of μ-law and A-law companding.
- Explain the meaning of non-uniform quantization. What is its main advantage?

- What is the difference between μ-law and A-law companding?

4) Part 4: Time-division Multiplexing (optional part)

- Launch WinFACET and go to Exercise 2 (Time Division Multiplexing).
- Go over the overview, then start the procedure, and execute all steps up to Step 26, and report your observations and answers to questions.

Additional Questions

Q1. How is PCM different than PAM?

Q2. Does a digital PCM stream of bits representing analog voice consume more or less bandwidth than the original analog signal?

Q3. What is the main advantage of PCM-TDM?

Q4. What is the difference between simplex and full-duplex communication links?