Experiment
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Pulse Amplitude Modulation

Hardware Experimentation

Objectives

The main objectives of this lab are:

1) To gain a practical understanding of the concepts of analog signal sampling, their usefulness and limitations.

2) To learn about Pulse Amplitude Modulation (PAM) as a transmission scheme.

3) To experiment with the realization of PAM modulation & demodulation using the DCB.

Pre-Lab Work

1) Explain the main result of the Sampling Theorem, and give a simple graphical proof for it (there is no need for full analytical details).

2) Discuss why is the sampling of analog signals quite useful in many applications?

Overview

An analog message signal, representing voice for example, has continuous amplitude and frequency values that vary with time. Analog communication systems transmit the complete analog waveform. But instead of transmitting the analog signal, it is possible to transmit discrete pulses (or samples) that represent some parameter of the message signal's waveform at regular intervals in time.

In the case of pulse amplitude modulation (PAM), the amplitude of the analog waveform is sampled at discrete (i.e., discontinuous) time instants, and transmitted as a sequence of pulses. Notice, however, that this sampling produces amplitudes that can still take *any value*, hence the scheme is not fully digital (unlike PCM which will be studied in a subsequent lab, and which uses quantization to represent the discrete samples by a finite number of bits).

As long as the sampling of the analog signal is taken with a sufficiently high frequency (higher than the minimum Nyquist rate of twice the signal largest frequency), it can be shown that there is *no loss* in information as a result of taking discrete samples. In fact, the PAM receiver simply passes the received samples through a low-pass interpolating filter to fully recover the analog waveform. Notice also that the results of the sampling theorem, which are derived for the case of ideal sampling with Dirac delta functions, are equally applicable for practical sampling with pulses that have a finite duration (refer to your textbook for more details).

Different types of sampling can be distinguished. With "natural" PAM signals, the amplitude of each pulse follows the amplitude of the message signal for the duration of the pulse. Another type of sampling results in "flat-top" PAM signals where a sample&hold circuit holds the sampled amplitude at a constant level between the sampling times, thus resulting in a stair-case PAM signal shape. This is sometimes used at the receiver end prior to low-pass filtering since it helps increase the amplitude of the reconstructed signal.

There are a number of advantages for transmitting PAM pulses rather than complete analog signals. For example, 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 full analog signal. For a sampling pulse train with a duty cycle fraction *PW/T* (i.e., the sampling pulse duration is *PW*, and its period is *T*), it can be shown that the power of the sampled PAM signal P_p is only a fraction of the total analog signal power P_s, given by: $P_p = (PW/T) \times P_s$.

Another advantage of PAM has to do with the ability to multiplex (or combine) several different signals and transmit them on the same communication channel. Although this is also possible with analog communication (using for example frequency division multiplexing-FDM), it is much simpler and more economical to implement with digital or discrete systems like PAM using what's known as time division multiplexing (TDM). Since PAM sends amplitude pulses of a given signal at discrete periodic time slots (or intervals), it is then possible to assign the remaining time slots for other signals, thereby maximizing the utilization of the channel. There is obviously a need for maintaining adequate synchronization at the transmitter and receiver levels to distinguish the different signals. This type of TDM transmission is very efficient and widely used in practical communication systems such as telephone networks, particularly in combination with Pulse Code Modulation-PCM (which is similar to PAM, but uses additional amplitude quantization as will be seen in a following lab).

Lab Work

2) **PAM Signal Generation**

- Use a two-post connector to apply the M2 message signal to the SAMPLER input. Measure the amplitude and frequency of this signal on the oscilloscope.

- Check the sampling pulse frequency on port SP of the DCB using the oscilloscope (it should be 8KHz). Is this sampling rate sufficient for the given input signal?

- Apply the input analog signal and resulting PAM signal from the SAMPLER output to channels 1 & 2 of the oscilloscope. Observe and sketch both signals. Is the re-construction quality good?

- What type of sampling is being used: natural sampling, or sample & hold?

3) **Power Calculation**

- Use the oscilloscope to measure the pulse width PW of the PAM signal, and the period of the sampling pulse train T_p.

- Calculate the pulse duty cycle PW/T_p.

- Calculate the power of the analog sinusoidal signal and the discrete PAM signal across a hypothetical 1 kΩ resistor. Comment on the efficiency of the PAM scheme.

4) **PAM Signal Demodulation**

- First, you need to measure the receiver filter characteristics. Use the external signal generator to apply sinusoidal signals with 2V pk-pk and increasing frequency to obtain a rough sketch of the filter frequency response.

  Hint: The filter 3dB cutoff frequency should be around 2.6 kHz.

- Disconnect the M2 input, and use the external function generator to apply a new sinusoidal input signal to the SAMPLER. Set the frequency to 2 kHz and the amplitude to 4V pk-pk.

- Now, use a 2-post connector to apply the sampled PAM signal at the output of the SAMPLER to the FILTER input. Observe the reconstructed signal at the FILTER output, and sketch it. Comment on the quality of the reconstructed signal.

- Let the input signal frequency now be changed in the range 3 to 3.5 kHz. Is the Nyquist criterion still met (given that SP is 8kHz)? Observe the demodulated signal at the FILTER output. Explain why it is still strongly attenuated (although the sampling frequency is sufficient!)

5) **Effect of Aliasing**

- When the sampling rate is 8KHz, what is the theoretical input frequency at which aliasing starts to appear?

- Select an input message signal frequency of 5KHz. Apply the original analog input and PAM demodulated signals on both channels of the oscilloscope. Sketch the resulting demodulated waveforms and comment.

- Sketch a frequency-domain description of the problem that you just observed in time-domain on the oscilloscope. Notice that it possible to observe things in frequency-domain too (using spectrum analyzers, which are currently not available in sufficient numbers for the EE370 lab requirements).

Additional Questions

Q1. Explain why it is sufficient to use a low-pass filter at the receiver to recover the original transmitted analog signal? Hint: you can assume that sampling is ideal (i.e., with delta pulses), then use `frequency-domain diagrams to show the effect of sampling on the signal spectrum.

Q2. Consider the sampling of a signal which has some maximum frequency f_m, but that may be corrupted by high-frequency noise. What should you do first? Explain why.