ECE 440L

Experiment 6: Analog / Digital Interface
(1 week)

I.   OBJECTIVES

Upon completion of this experiment, you should be able to:

1.   Specify an appropriate sampling frequency and anti-aliasing filter for a given signal;

2.   Recover sampled signals and investigate distortions involved in this process;

3.   Measure the quantization noise produced by digitization of an analog signal;

4.   Measure and plot the transfer characteristics of linear and companding digitizers.

II.   INTRODUCTION

A typical example of information to be transmitted over a digital communication system is speech. The speech signal voltages are converted by an analog-to-digital converter (A/D converter) to a serial binary signal stream. At the receiver, the received binary signal stream is converted to the analog speech signal by a digital-to-analog converter (D/A converter). A *pulse-code modulation (PCM)* system results when, as discussed above, an analog signal is digitized, transmitted serially, received, converted back to a "replica" of the input.

Anti-aliasing filter. The conversion rate of the A/D system (conversions/second) must exceed twice the bandwidth of the signal to be digitized.[1] Therefore, the first stage in converting any message signal to a binary signal stream is an anti-aliasing filter; this filter ensures that unwanted signal and noise components at higher frequencies do not "alias" into lower frequencies due to the spectrum folding back.

Serial A/D converter. The first stage of the A/D converter is a sample-and-hold (S/H) system that samples the analog data at the sampling instant and holds the value until A/D conversion is complete. The output of the A/D converter is a serial stream of binary data. Linear and companding A/D converters will be investigated in this experiment.

Serial D/A converter. captures data and produces an analog voltage corresponding to binary values in the data stream. The zero-order hold (ZOH) stage in D/A converters is equivalent to the S/H stage in A/D converters. S/H and ZOH stages introduce a distortion called the aperture effect.

Reconstruction filter. The spectrum of the output of the D/A converter will have frequency components above the frequency of the desired output signal. The function of the reconstruction filter is to eliminate these components as well as possible

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[1] If the signal bandwidth is limited to $\pm W$, the minimum sampling rate (no aliasing) is 2W, the Nyquist rate.

III. PRELAB[2]

1. Review Woodward's theorem from Experiment 5.

2. Nyquist:

2 a. If the Fourier transform of

$$x_\delta(t) = x(t) \sum_{n=-\infty}^{\infty} \delta(t - nT_S)$$

is

$$X_\delta(t) = f_S \sum_{n=-\infty}^{\infty} X(f - n f_S)$$

where $f_S = \dfrac{1}{T_S}$,

what is the minimum sampling rate to be used for a signal band-limited to $(-f_0, \; f_0)$ if there is to be no aliasing? Give a reference (including page number).

2 b. To illustrate the effect of the choice of sampling frequency, f_S, sketch a plot of the Fourier transform of the sampled signal for two cases:

one with sampling frequency just sufficient to avoid aliasing
and
one in which the sampling rate is much larger than necessary.

2 c. What are the minimum sampling rate and anti-aliasing filter bandwidth needed to recover:

$$x(t) = \sin(2\pi \; 1000 \, t) + \sin(2\pi \; 2000 \, t)$$

$$x(t) = sq(2\pi \; 1000 \, t)$$ where sq is a ± 1 volt square-wave [3] and the first 3 non-zero harmonics are to be acquired?

3. If an 8-bit serial A/D converter clocks out bits at a rate of 8,000 bits/second, what is the maximum rate at which it can sample a signal?

4. Suggest a filter to reconstruct a signal sampled at the rate determined for 3 above.

5. Briefly define companding. (See pages 321-322 of your textbook.)

6. BONUS: Briefly define PCM threshold. (See pages 319-320 of your textbook.) How does the quantization noise for a 4 bit system compare to the quantization noise of a 7 bit system?

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[2] Notation is from Ziemer and Tranter, Principles of Communications, Systems, Modulation, and Noise

[3] From ECE207 and ECE208 notation.

IV.   EXPERIMENT

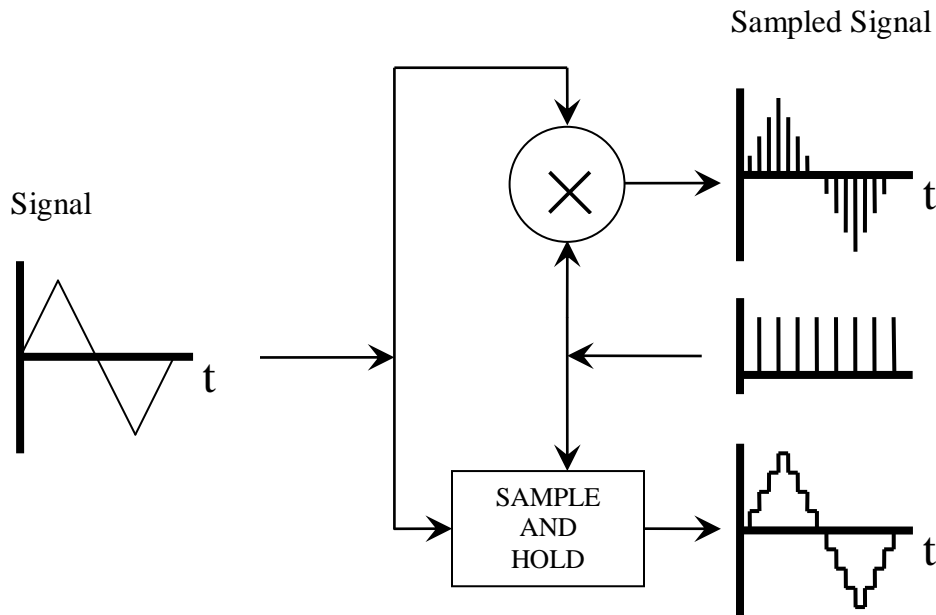### 1.   Natural Sampling and Flat-topped Sampling. (TA DEMO)

Sampled Signal

Fig. 1a. Sampling Modes

Input Signal. Any signal generator can generate the triangle wave. Start with :
> 500 Hz, 5 Vpp with no offset.

Pulse Train. The 33220A can generate the pulse train. Start with:
> 10 kHz, 0 to 10 volt, 5 μsec pulse

Sample-and-hold. S/H can be performed by using the TIMS '**INTEGRATE-AND-DUMP**' module which has built-in S/H functionality. This module has two independent channels marked as <I&D1> and <I&D2>. You may use either channel. Unplug the module and set the rotary switch corresponding to the channel you want to use to the <S&H> setting. (The other available settings are <I&D>, <I&H>, and <PWM>.)

TIMS Modules. The TIMS modules utilize the following convention: Inputs are on the left, outputs are on the right, digital signal sockets are red, and analog signal sockets are yellow.

TIMS Clock signals. TIMS modules that require a clock signal can be driven by the SYNC signal from any of the signal generators; any other *TTL* signal source with the appropriate frequency may also be used.

Signal	Sample Pulse	Oscilloscope		Spectrum Analyzer	
Type / Freq	Freq. / Width	Natural	S/H	Natural	S/H
Triangle / 500 Hz	10 kHz / 5 μsec				
,,	observe as adjust				
,,	pulse freq to 300 Hz				
Triangle / 500 Hz	5 kHz / 5 μsec				
,,	observe as adjust				
,,	width to 180 μsec				
Several / 500 Hz	5 kHz / 5 μsec				

2.   Aliasing. (TA DEMO)

2a.   To create a test signal $x(t)$ band-limited (roughly) to 2 kHz, generate a wide-band FM signal using the HP3314A VCO. Use a low frequency symmetric ramp as the input to the VCO. Observe this signal on the oscilloscope and spectrum analyzer,
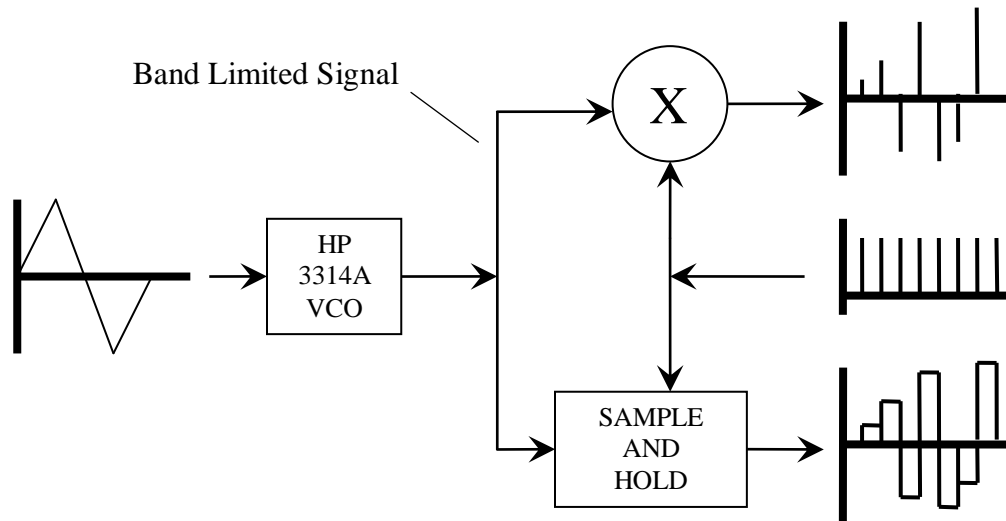

Fig. 1b. Sampling Modes

2b.   To investigate aliasing, the band-limited test signal generated using sample-and-hold and natural sampling as shown in Figure 1b.

Sample Pulse	Analyzer Span	Oscilloscope		Spectrum Analyzer	
Freq. / Width	Start / Stop	Natural	S/H	Natural	S/H
10 kHz / 5 μsec	0 Hz – 50 kHz				
adjust freq, observe	,,				
10 kHz / 5 μsec	0 Hz – 20 kHz				
adjust freq, observe	,,				
10 kHz / 5 μsec	0 Hz – 10 kHz				
adjust freq, observe	,,				
5 kHz / 5 μsec	0 Hz – 10 kHz				
adjust freq, observe	,,				
10 kHz / 5 μsec	0 Hz – 50 kHz				
adjust width, observe					

3. A PCM Communication System

A complete PCM communication system is shown in Figure 2. The modules used in this system are the TIMS PCM encoder and TIMS PCM decoder. The TIMS encoder includes the A/D converter (with an anti-aliasing filter, sample-and-hold, optional compressor, and digitizer) and parallel-to-serial converter. The TIMS decoder includes the serial-to-parallel converter and D/A converter (with an optional expander, zero-order-hold, and reconstruction filter).
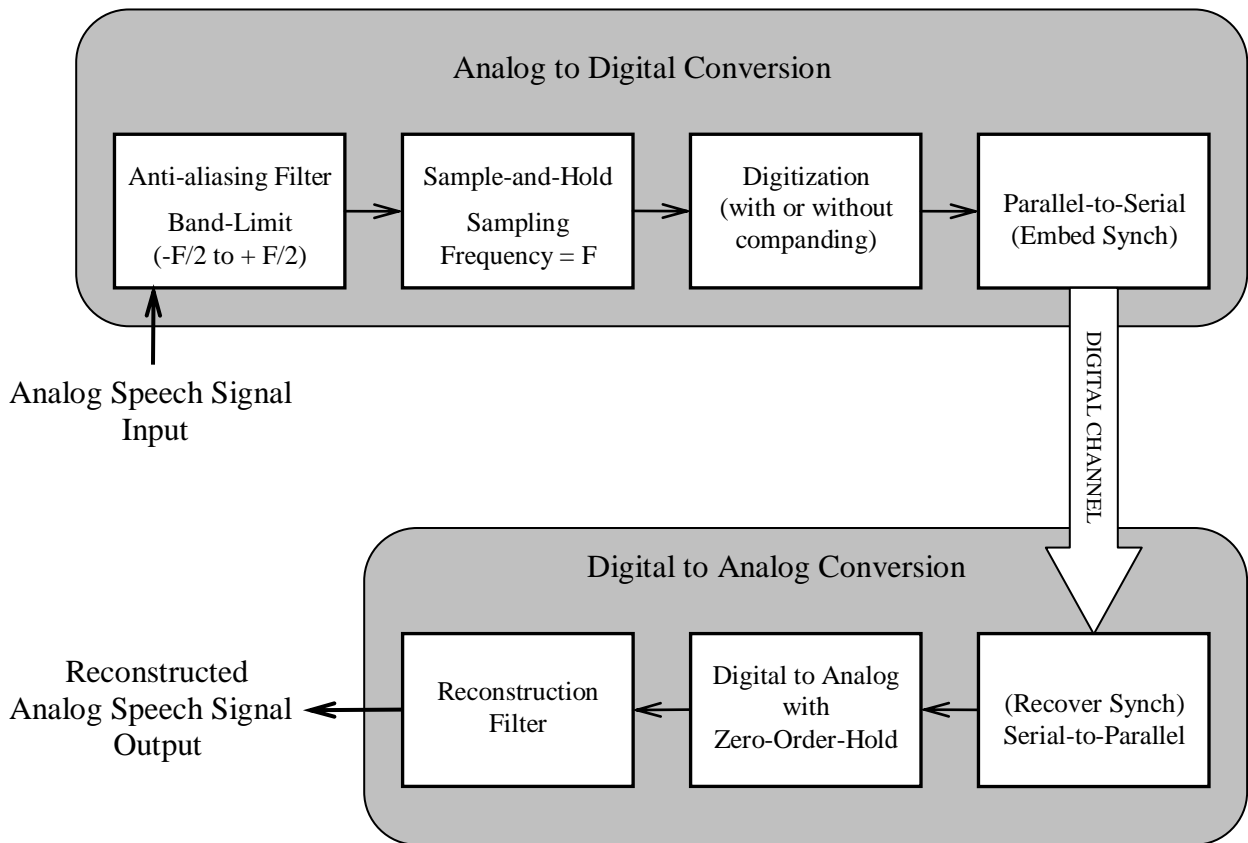

Fig. 2. Pulse Code Modulation System including A/D and D/A blocks.

3 a.   Characteristics of the PCM system.

- Set the PCM Encoder & Decoder to the '4-bit linear.' Internal switches (OFF-OFF).
- Connect CLK inputs of the PCM Encoder and Decoder to the 8.3 kHz TTL clock.
- Connect a 4 Vpp 50 Hz triangle wave to the Vin input of the Encoder.
- Connect the Encoder PCM DATA output to the Decoder PCM DATA input.
- Switch the FS SELECT switch on the PCM Decoder to EMBED.
  This setting causes the Decoder to extract FRAME SYNC from PCM DATA.
  The FRAME SYNC (FS) frequency is the conversion rate for this system.
- Observe the PCM DATA with the oscilloscope. Should be changing TTL data.
- Record the system input and output on the oscilloscope and spectrum analyzer.
- Notice that the output of the decoder has the form of 0th order S/H.
- Adjust the frequency of the input triangle wave and the time scale on the scope to obtain a display that proves that the sample rate is the same as the FS.
  Hint: Sync the scope on FS.

3.   A PCM Communication System (continued)

3 b.   Investigating the 4-bit linear A/D.

- Prepare a table mapping input voltages to output bit sequences for 4-bit linear. Sync the scope on FS. (You may have to set "Hold-Off" to get stable sync.)

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FRAME SYNC  yields:    1 0 0 0 0 0 0 0 1 0 0 0 0 0 0 0 1 0 . . .
Vin > +2.3V yields:    1 0 0 0 1 1 1 1 0 0 0 0 1 1 1 1 1 0 . . .
Vin =  0.0V yields:    1 0 0 0 1 0 0 0 0 0 0 0 1 0 0 0 1 0 . . .
Vin < -2.3V yields:    1 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 1 0 . . .
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- Apply DC voltages to the PCM encoder input and observe the (PCM) output. The table should have codes for 2^4 (that is 16) DC values.

3 c.   Add Anti-aliasing and reconstruction filters.

- Use a Kron-Hite filter as an anti-aliasing filter. This filter precedes the input.
- Set the low-pass frequency appropriately. Discuss your selection.
- Apply a 4 Vpp 100 Hz sinewave to the new system input (the anti-alias filter).
- Use a Kron-Hite filter to appropriately filter the output of the Decoder, Vout.
- Compare time and frequency domain displays of the output with/without the filter.

3 d.   Linear and Companding Transfer characteristics.

- Connect a 4 Vpp 10 Hz triangle wave to the input of the anti-aliasing filter.
- Obtain scope plots for the input triangle wave, the 4-bit linear output of the D/A, and the output of the reconstruction filter.
- Subtract the input signal from each of the output signals and determine the RMS error for each case.
- The D/A converter output vs. the system input is the system transfer characteristic. Obtain plots of "Vout vs. Vin" for 4-bit linear, 7-bit linear and 4-bit companding. State whether the plots were taken before or after the reconstruction filter.

3 e.   Play some music.

- Connect the output of the CD player to the input of the anti-aliasing filter.
- Connect the filter output to an AUX AMP.
- Connect the output of the AUX AMP to the input of the A/D, Vin.
- Adjust the CD Player and AUX AMP gains until Vin is approximately 3 Vpp.
- Connect the output of the reconstruction filter to the scope and speaker input.
- Observe and listen to the output voltage for 4-bit linear, 7-bit linear, and 4-bit companding. Don't expect high quality audio.
- Observe and listen to the 4-bit linear output voltage as you vary the reconstruction filter bandwidth from very low to very high.

V.   REPORT

Document all the readings you have obtained and any conclusions you draw in your report. Attach a copy of your lab record to the report. Answer any specific questions asked in the lab manual. Also, read the appendix and answer the following:

REFERENCES

[1] R. E. Ziemer and W. H. Tranter, *Principles of Communications*, 5th ed. Hoboken, NJ: John Wiley, 2002.