EE521 Analog and Digital Communications

January 25, 2006

Instructor: James K Beard, PhD
Office: Ft. Washington TBA
Email:. jkbeard@temple.edu, jkbeard@comcast.net
Office Hours: Wednesdays 3:00 PM to 4:30 PM; 5:30 PM to 6:00PM
Location: Ft. Washington 107
Time: Tuesdays 6:00 PM – 8:30 PM
Web Page: http://temple.jkbeard.com

Texts:

- Bernard Sklar, Digital Communications, Second Edition, Prentice Hall P T R, 2001 (2004 printing), ISBN 0-13-084788-7
- Digital Communication Systems Using SystemVue, by Dr. Silage, ISBN 1-58-450850-7

Today's Topics

1 Course Overview

See Synopsis (in handouts)
See Communication Block Diagram (in handouts)

2 Introduction to SystemView

See color screen shot in handouts
See PowerPoint handout

2.1 Installation and licenses:

- You have a CDROM with your copy of Sklar that includes a copy of the Student Version, limited to saving systems with no more than 10 tokens

- Dr. Sliage's book also comes with a CDROM with the Textbook Version; this is a viewer only, and will load and run only the files on the CDROM "Examples" folder
- Install and use the Student Version on your own computer until February 8
- Full versions are available on the main campus
  - A&E Building 6th floor labs
  - Ask Dr. Silage for access
- On February 8 I will give you a CDROM with
  - Full copy of SystemView
  - License code for your use in your Term Project, good for 90 days

2.2 SystemView is

- A simulation tool
- A development tool for burning FPGAs from Xilinix

2.3 SystemView Simulations

- High fidelity – actual signals represented in the simulation
- Everything is analog, sampled at the SystemView sample rate
- Data passed between tokens is always floating point at the SystemView sample rate
- Sampled data must be at a rate lower than the SystemView sample rate
- All signals are in one of three data formats:
  - Analog voltages, can be any valid floating point value, usually in range -1 to +1
  - Digital signals, may be 0 and 1, or may be -1 and +1
  - Integers, signed or unsigned, represented in 32 bits but passed as floating point values equal to integers in SystemView
  - Symbols, always in integer data format

The tokens in SystemView

- Provide a signal in one of the three basic data formats
- Range and type of output usually set in token setup parameters
- Input(s) are in one of the three basic data formats

2.4 Fundamental Principles of SystemView Token Use

- Always make sure that the inputs and outputs are in the proper data format. – it's up to you, SystemView tokens will accept the wrong data formats silently.
- Convert from one data format to another using the Function group in the standard libraries and the Comm group in the optional libraries.
- Always make the SystemView sample rate an integer multiple of the highest sample rate used in your simulation.
- Use the data sinks for output

2.5 Dr. Silage's Book

Use the book for reference on the use and functions of the tokens. This information is not available anywhere else.

3 Sklar Chapter 2, Formatting and Baseband Modulation

3.1 Baseband Signals

Definition: A signal that is not on an RF carrier is a baseband signal.
Examples: Analog voltages, digital signals, bit streams, sampled data

3.2 Messages, Characters, and Symbols

Messages, characters and symbols are baseband signals.
Definitions:

- Message – the data in the format as generated, as text strings or analog voltage representing speech. This is the input and output of the communications system.
- Characters – the groups of bits in the message that represent the granularity of the message, such as ASCII characters (7 or 8 bits) or analog-to-digital converter (ADC) words (typically 8 to 16 bits).
- Symbols – groupings of message bits. Usually the number of bits M in a character is a power of two, $M = 2^k$. The character size is characterized either by the number of bits M or the power of two k. For M=2, k=1 the characters are binary. For k>1 the characters are *M-ary*.

The number of bits per symbol is related to the design of the communications system, and is not usually related to the number of bits in the characters. See Figure 2.5, page 62, in Sklar for an example where M is much larger than the character size.

3.3 Formatting Analog Information

Analog data typically is sampled and digitized to form a character stream of ADC words. The sample rate must be at least twice the highest frequency to be reproduced or an ambiguity will result in signal reconstruction.

EXAMPLE: Sampling a Sine Wave

Sine wave frequency f

Sample rate F

The data samples are

$$y_i = E\cdot\cos\left(2\pi f\cdot\left(i\cdot\Delta T\right)\right) = E\cdot\cos\left(2\pi f\cdot\frac{i}{F}\right)$$
$$= E\cdot\cos\left(2\pi\cdot\frac{f}{F}\cdot i\right)$$

We note that the frequency f+F will give the same data samples:

$$ya_i = E\cdot\cos\left(2\pi\cdot\frac{f+F}{F}\cdot i\right) = E\cdot\cos\left(2\pi\cdot\frac{f}{F}\cdot i + 2\pi\cdot i\right)$$
$$= E\cdot\cos\left(2\pi\cdot\frac{f}{F}\cdot i\right)$$

This is a way of showing that sine waves at frequencies that differ by the sample rate F produce the same data. We say that they are *aliased* to each other. Examination of similar examples shows that the range of frequencies nearest baseband that can be reproduced without aliasing is

$$-\frac{F}{2} < f \le +\frac{F}{2}$$

This is the sampling theorem.

One design feature that results directly from the sampling theorem is the use of low-pass filters prior to a sampling operation. These filters have sufficient attenuation at frequencies of $F/2$ and higher so that high-frequency signals that are aliased back down to baseband do not interfere with the intended signal. The strength of any high-frequency signals and the tolerable levels after conversion back to the original signal determines how much attenuation is necessary. These are called anti-aliasing filters.

The Sampling Aperture and Natural Sampling

The sampling operation is analog. The input voltage is passed through a switch to a holding amplifier. The switch is on for a short time, nearly always a very small proportion of the time between samples. There are a number of ways of representing the action of the holding amplifier while the switch is on, but all of them can be characterized as the impulse response of a filter.

The principle of superposition can be used to show that sampling is equivalent to these operations in order:

1) Passing the signal through a filter that has an impulse response equal to the shaping of the sampling aperture. This is a convolution.
2) Multiplying that signal by a series of impulse functions. This is a multiplication.

Thus the effect of the sampling aperture is to apply a slight low-pass filtering to the signal.

One way of conceptualizing this process is to look at it in the frequency domain. The frequency response of the aperture function is multiplied by the energy or power spectrum of the signal, which is then convolved with the Fourier transform of the perfect impulse sampling.

The Fourier transform of an infinite series of impulse functions does not provide us with a convergent integral. The use of limiting functions and Cauchy principal values does provide us with a formal Fourier transform: an infinite series of impulses in the frequency domain, separated in frequency by *F*. Thus the sampled signal, in the frequency domain, is the original signal spectrum, sampled by the frequency response of the aperture function, then replicated at intervals of *F*.

One common way to account for the effect of the sampling aperture on an analog signal is to approximate it by a short rectangular aperture. This is called *natural* sampling. This gives us the familiar $\sin(x)/x$ frequency response. For a rectangular aperture width of τ with unit area, the frequency response is

$$\begin{aligned} G(f) &= \frac{1}{\tau}\int_{-\frac{\tau}{2}}^{\frac{\tau}{2}} \exp(-j\cdot 2\pi\cdot f\cdot t)\cdot dt \\ &= \operatorname{sinc}(\pi\cdot f\cdot\tau) \\ &\approx 1-\frac{(\pi\cdot f\cdot\tau)^2}{6} \end{aligned}$$

Since $f\cdot\tau << 1$, this is a gentle roll-off of the low-pass signal. The use of this approximation allows us to characterize this roll-off with a single parameter, τ.