EE521 Analog and Digital Communications

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Texts:

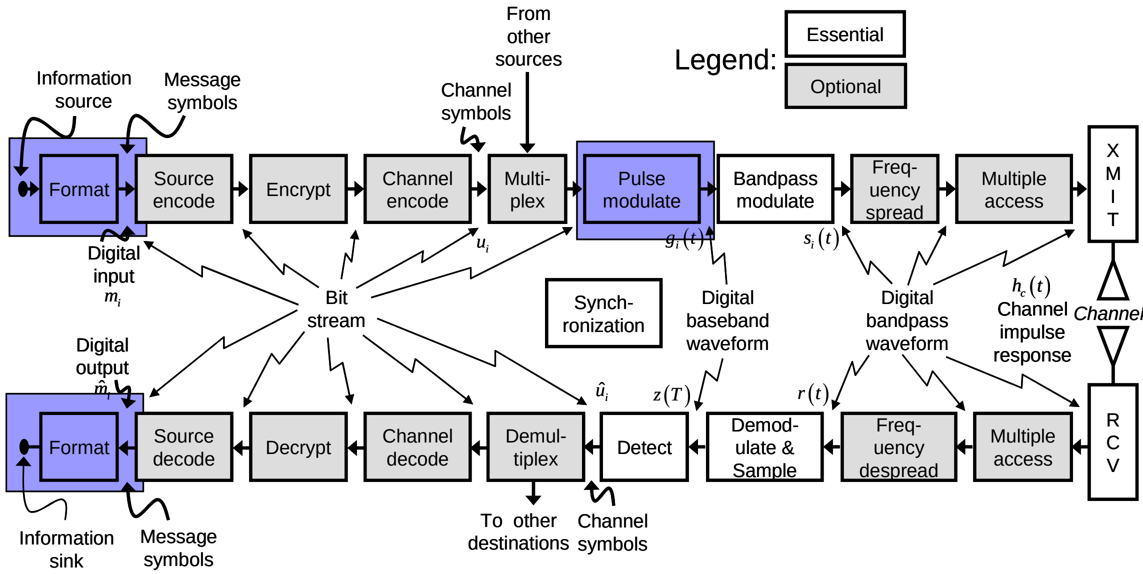
- Bernard Sklar, Digital Communications, Second Edition, Prentice Hall P T R, 2001 (2004 printing), ISBN 0-13-084788-7
- Digital Communication Systems Using SystemVue, by Dr. Silage, ISBN 1-58-450850-7

Today's Topics

- SystemView demonstration and discussion
- From Sklar Chapter 2, Formatting and Baseband Modulation
  - Sources of Corruption
  - Pulse Code Modulation
  - Uniform and Non-uniform Quantization

Overview

See block diagram below



SystemView Demonstration and Discussion

Student Version from Sklar: Short demo and discussion

Dr. Silage's Book

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

From Sklar, Chapter 2

Sampling

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.

Note the roll-off approximation in the last line, in dB, is

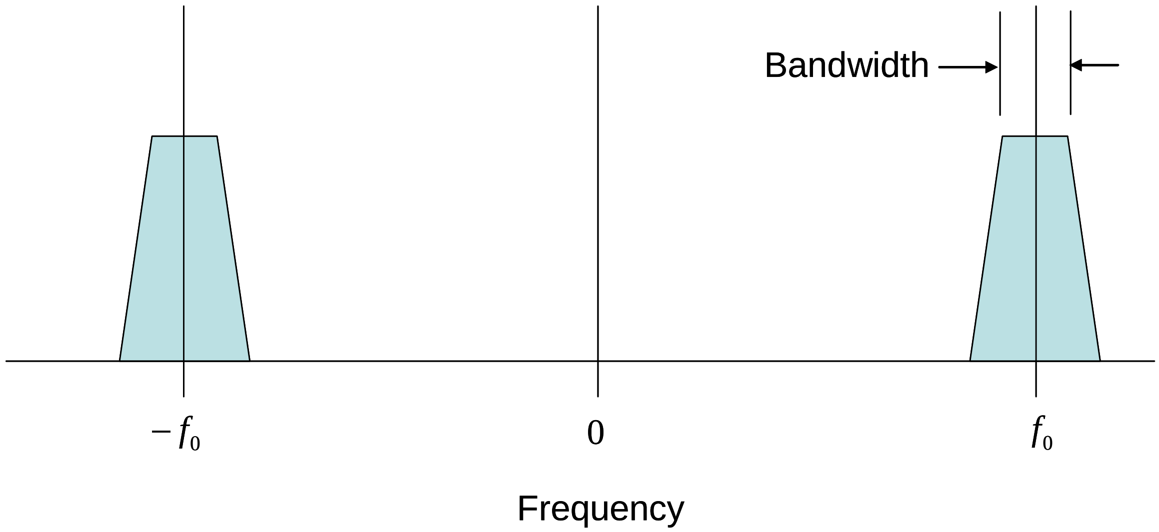
$$\begin{aligned} 20\cdot\log\left(1-\frac{(\pi\cdot f\cdot\tau)^2}{6}\right) &= 8.68588\ldots\cdot\ln\left(1-\frac{(\pi\cdot f\cdot\tau)^2}{6}\right) \\ &\approx -14.3\cdot(f\cdot\tau)^2 \text{ dB} \end{aligned}$$

Thus for a gate duty cycle of 0.1%, the roll-off in the band is $1.4\cdot 10^{-5}$ dB, or essentially negligible. You can use this equation to evaluate sampling aperture rolloff for higher effective sampling gate duty cycles.

Undersampling Signals on a Carrier (Digital Receivers)

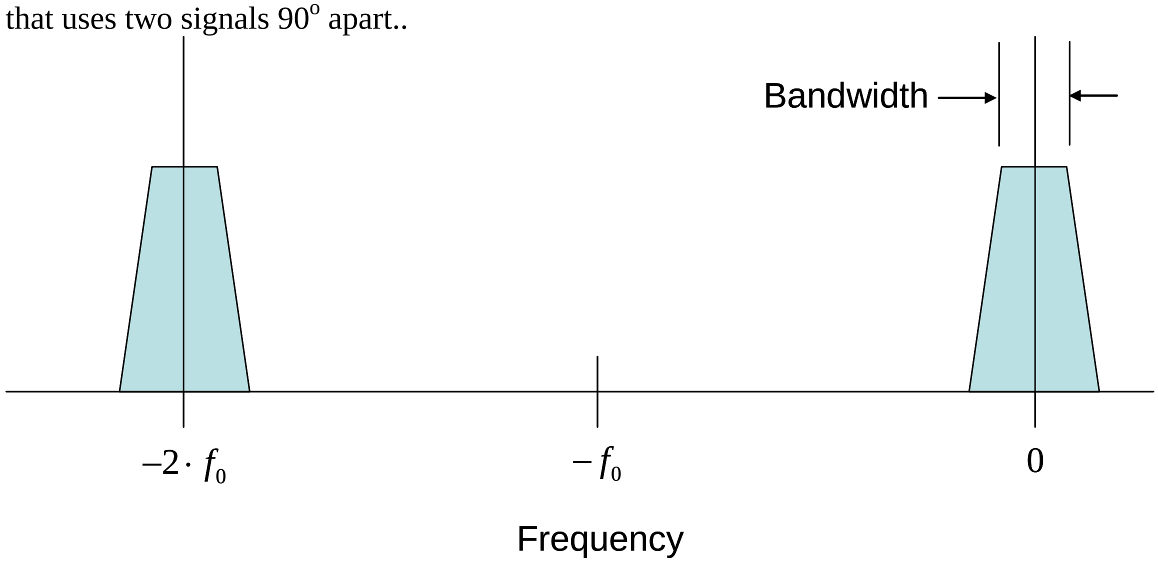
The following material is NOT in the text. It is related to carrier signals – baseband signals modulated on a carrier signal.

The signal center frequency is often much larger than the bandwidth. Nyqist criteria applies to the signal bandwidth, not the center frequency. See diagram below for a signal on a carrier at frequency f_0.



The sample rate must be twice the bandwidth, not twice the highest frequency seen in the signal. The criterion for sample rate being twice the highest frequency in the signal applies to baseband signals. The diagram shows a bandpass signal.

To sample bandpass signals, the signals are usually mixed to baseband. See the figure below for a bandpass signal mixed to baseband by a complex LO (local oscillator) – one that uses two signals 90^o apart..



The resulting signal is complex, i.e. there are two channels, and both must be sampled at twice *half the bandwidth*, or at the signal bandwidth. The two channels together provide samples at twice the signal bandwidth.

Note that positive and negative frequencies are discriminated by phase. Although $\cos(-2\pi \cdot f \cdot t)$ may not be distinguishable from $\cos(+2\pi \cdot f \cdot t)$, $\exp(-j \cdot 2\pi \cdot f \cdot t)$ is clearly distinguishable from $\exp(+j \cdot 2\pi \cdot f \cdot t)$.

The LO and mixing operation can be the sampling operation itself. This is the distinguishing characteristic of a digital receiver. Sampling operations that mix or alias the carrier center frequency, positive or negative, to zero frequency are

$$f_s = \frac{4 \cdot f_0}{2 \cdot k + 1} > B \,.$$

Note that we have added the Nyquist criterion for the lower limit.

Sources of Corruption

Additive noises include

- Quantization – effectively adds noise to the digitized signal
- Quantizer saturation – maximum possible levels from quantizer will cause clipping, or worse end-around images, of inputs that are too large
- Timing jitter – non-uniform sampling of the input due to noise on the quantizer clock, effectively frequency-modulates the signal

Channel noise includes

- Variations in amplitude and phase, or even drop-outs of signal over time
- Circuit effects such as interference from transmitter or receiver artifacts
- Intersymbol interference – trailing interference from the previously transmitted symbol(s) or from other signals that are sharing the channel

Pulse Code Modulation

This is a class of baseband signals made from the words generated by an ADC from an analog signal. This brings us to types of quantization used in ADCs.

Uniform quantization can simply be a set of uniformly spaced thresholds at voltages equal to multiples of a quantization level. For example, for a quantization level of 1 mv and 4096 quantizations (a 12-bit word), the quantization thresholds vary from -2.048 volts to +1.023 volts. There is a threshold at 0 volts. Another scheme that is useful when very small words are used is to use plus and minus half the quantization level for the smallest thresholds so that the quantization levels are symmetrical about 0 volts for an even number of thresholds; this is simply an offset of half a quantization level.

Uniform and Non-Uniform Quantization

Non-uniform quantizations are often used in communications to minimize quantization noise in small signals while allowing large signals to be accommodated without quantizer saturation. Varieties include quadratic amplitude compression, companders, and logarithmic compression-expansion.

Assignment for next time

- Read Sklar Chapter 2: 2.1, 2.3, 2.4, 2.5, 2.6, 2.7, 2.8

- Load the Student Version of SystemView included with Sklar and examine the samples and demos
- Browse appendices of text for review and supplementary material
- Look at TUARC, K3TU; websites
    - http://www.temple.edu/ece/tuarc.htm
    - http://www.temple.edu/k3tu