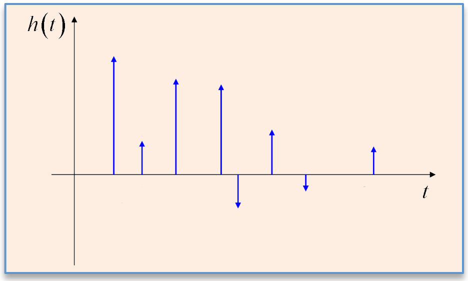
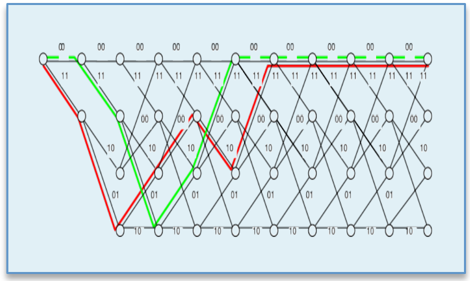
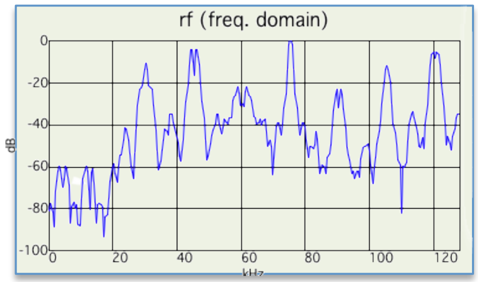
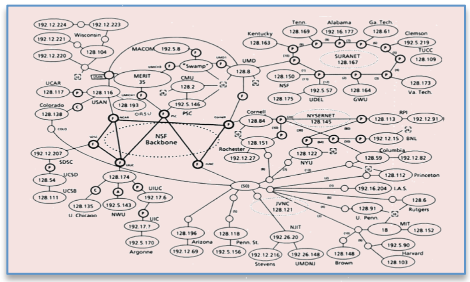








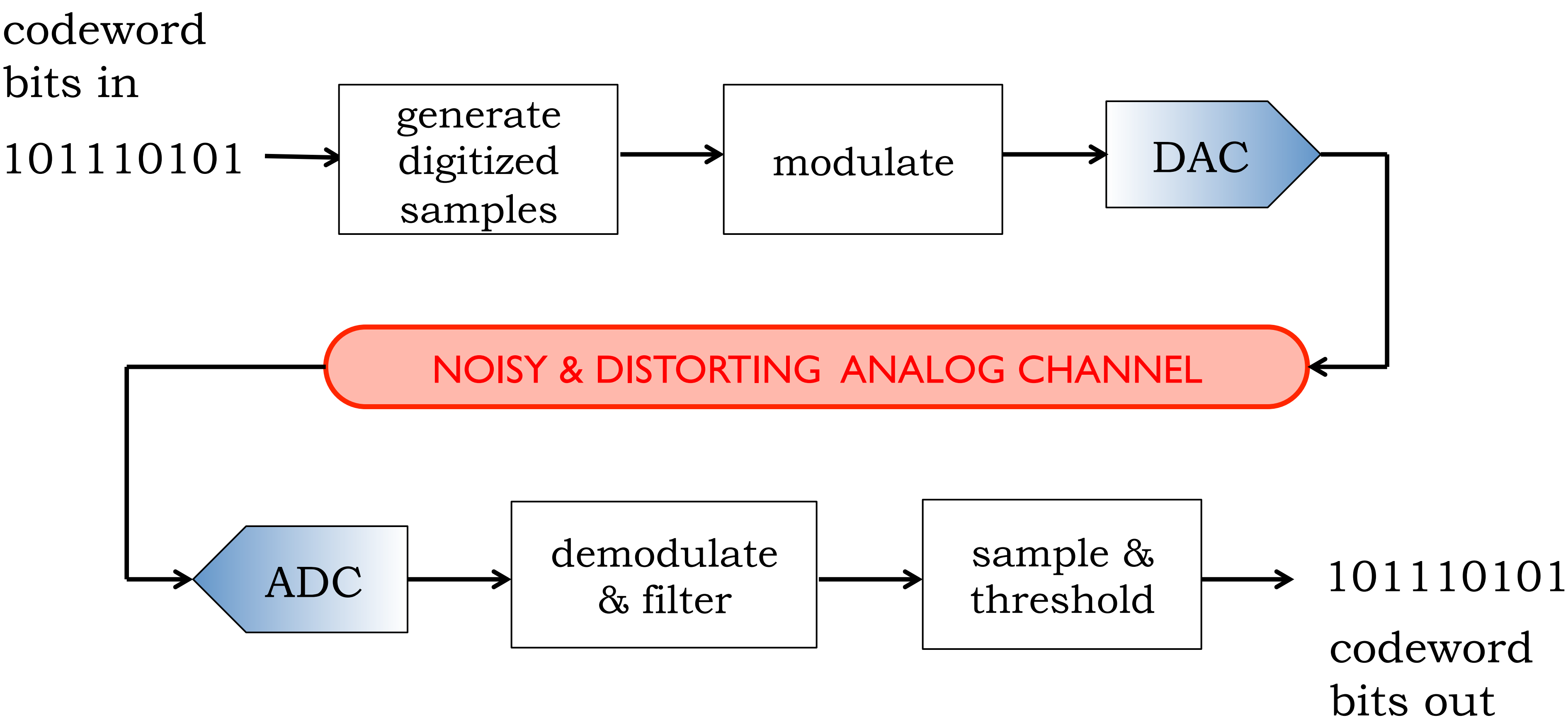
INTRODUCTION TO EECS II

DIGITAL COMMUNICATION SYSTEMS

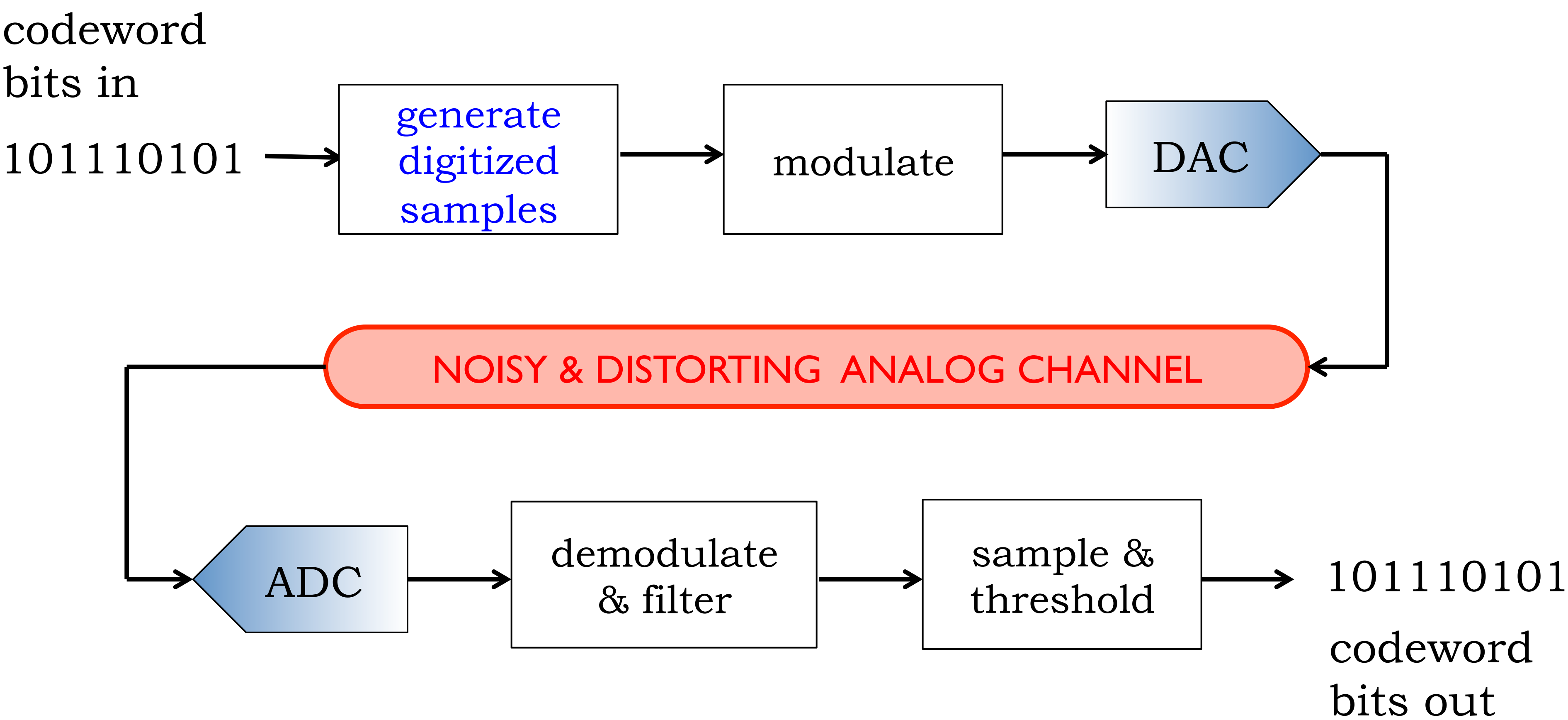
6.02 Fall 2011
Lecture #10

- Measuring and modeling channel behavior
- Input/output descriptions of systems
- Linear time-invariant (LTI) models

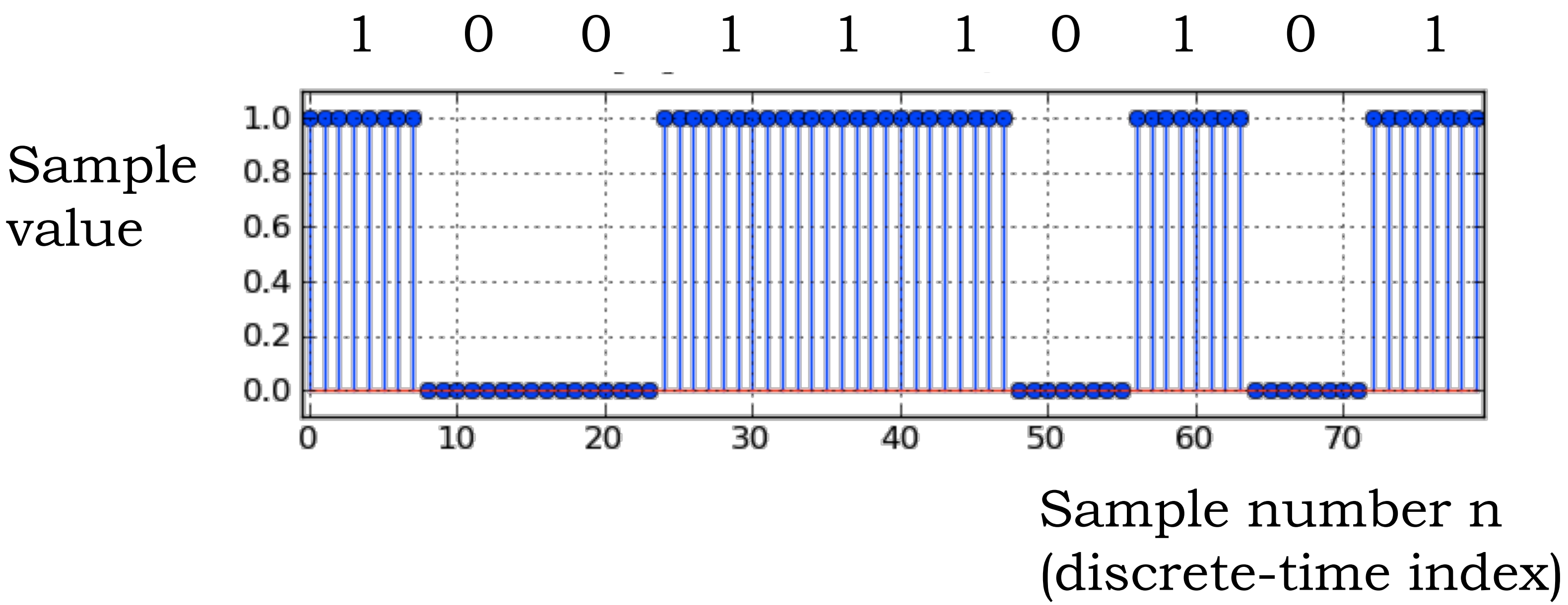
Modeling Channel Behavior



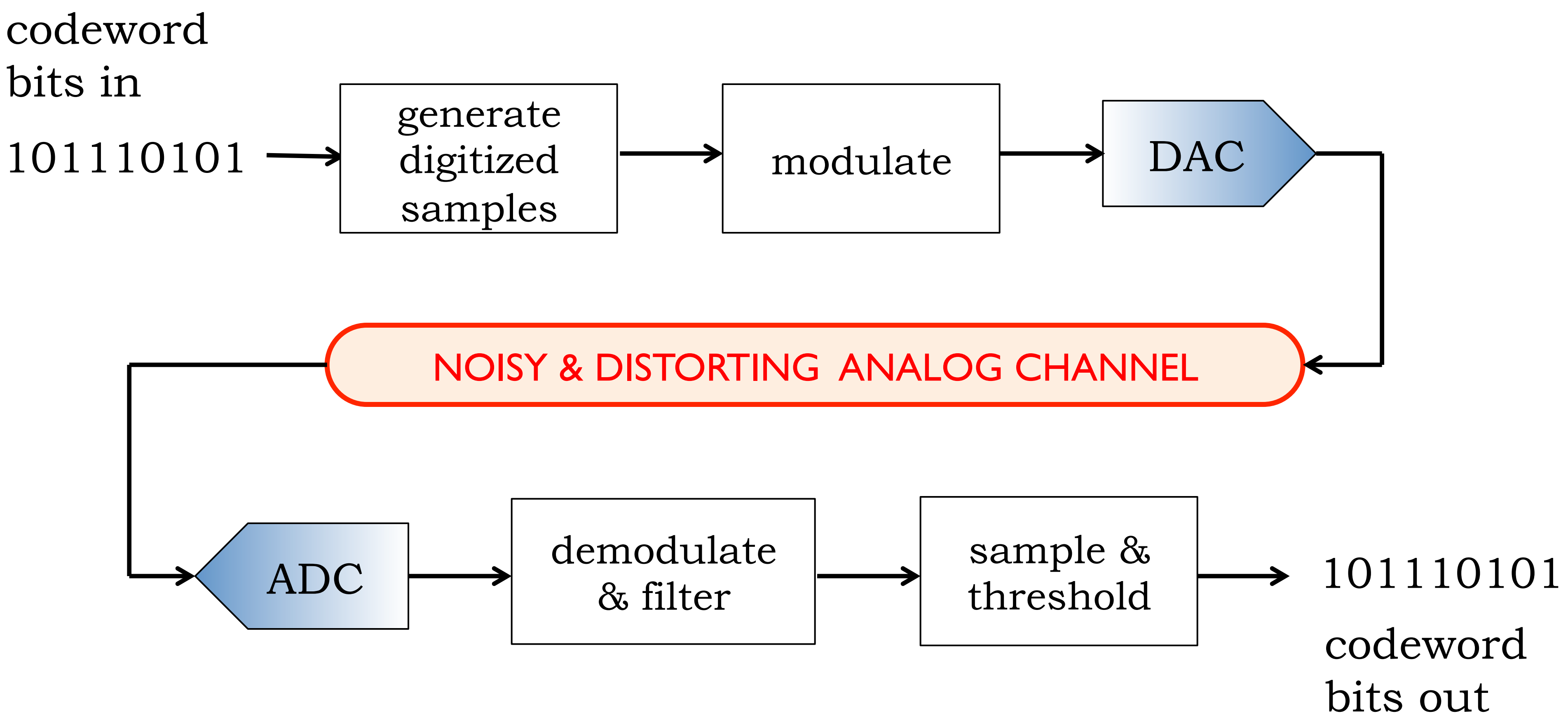
Modeling Channel Behavior



Digitized Samples

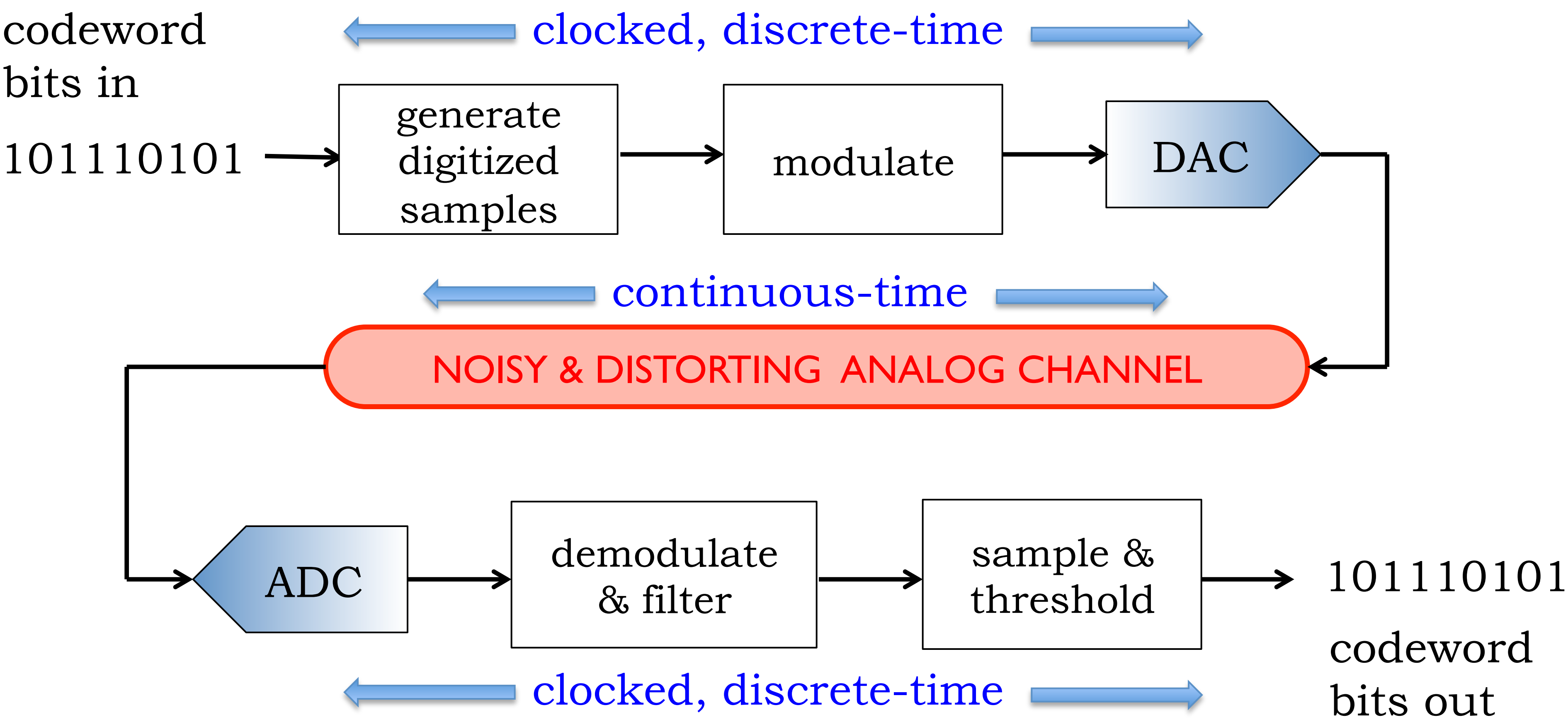


Modeling Channel Behavior

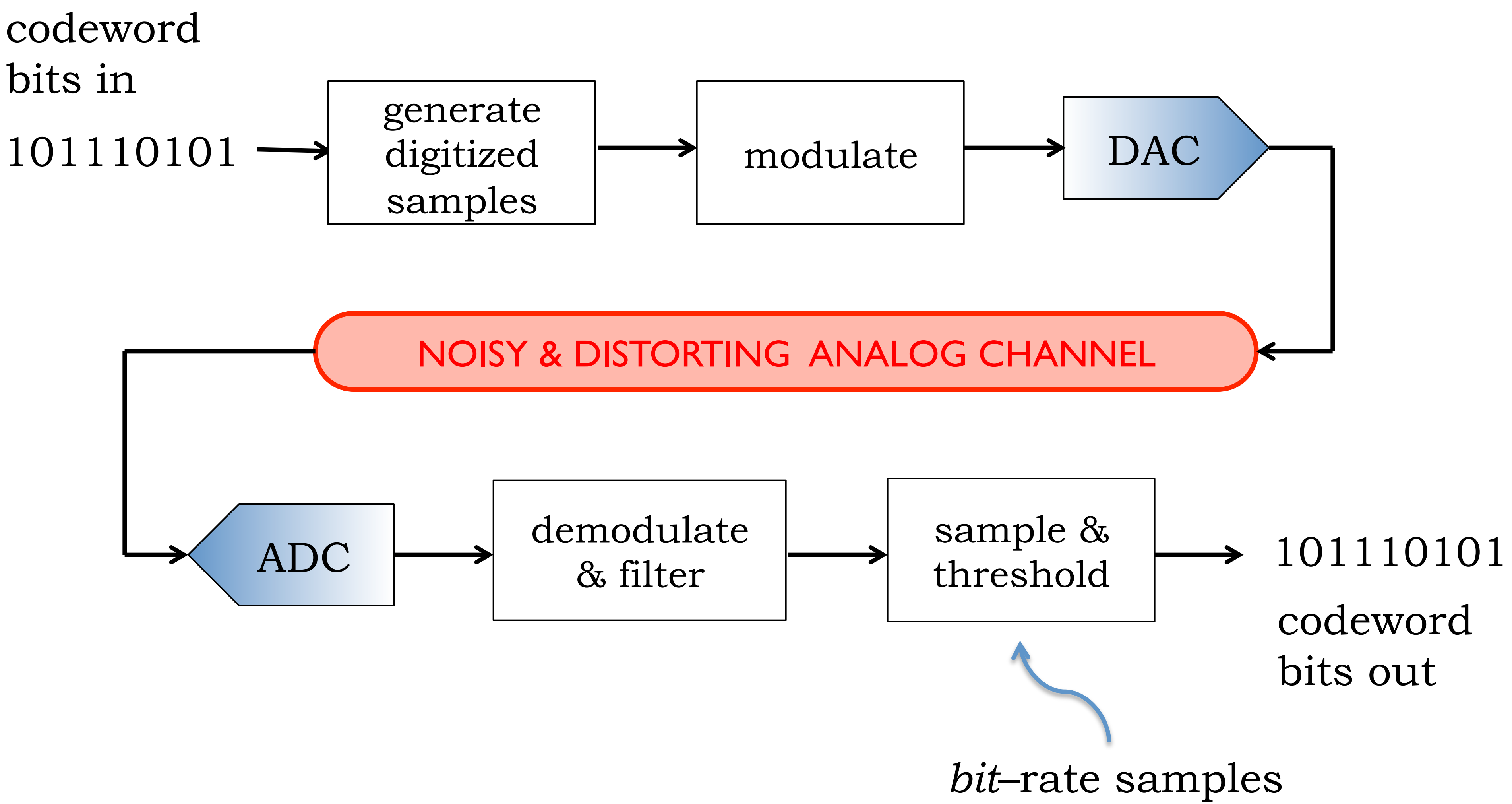


DAC: Digital-to-analog converter
ADC: Analog-to-digital converter

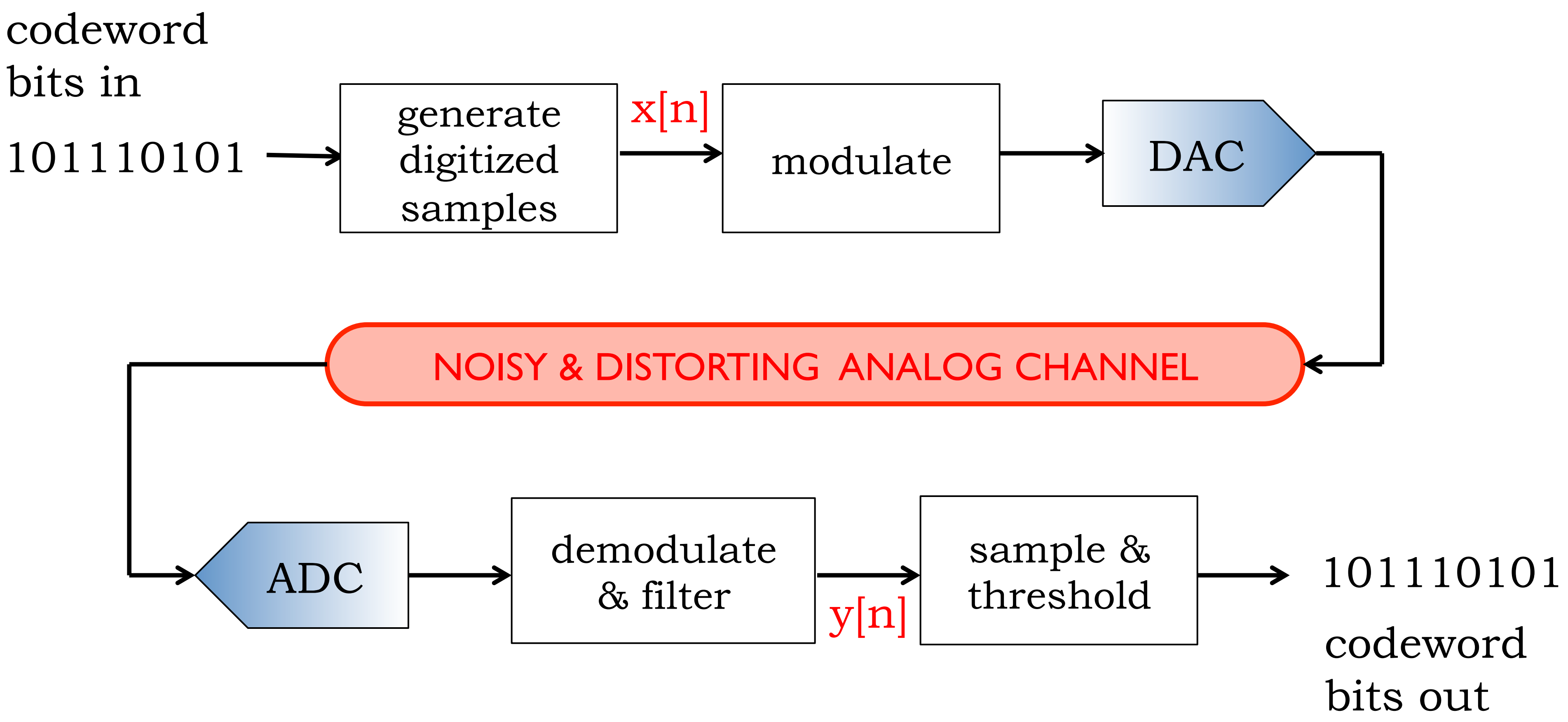
Modeling Channel Behavior



Modeling Channel Behavior

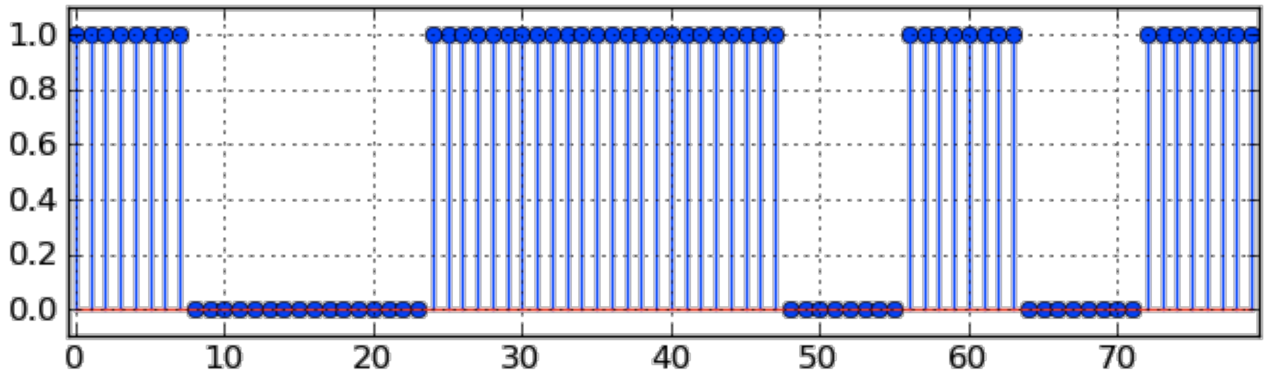


Modeling Channel Behavior

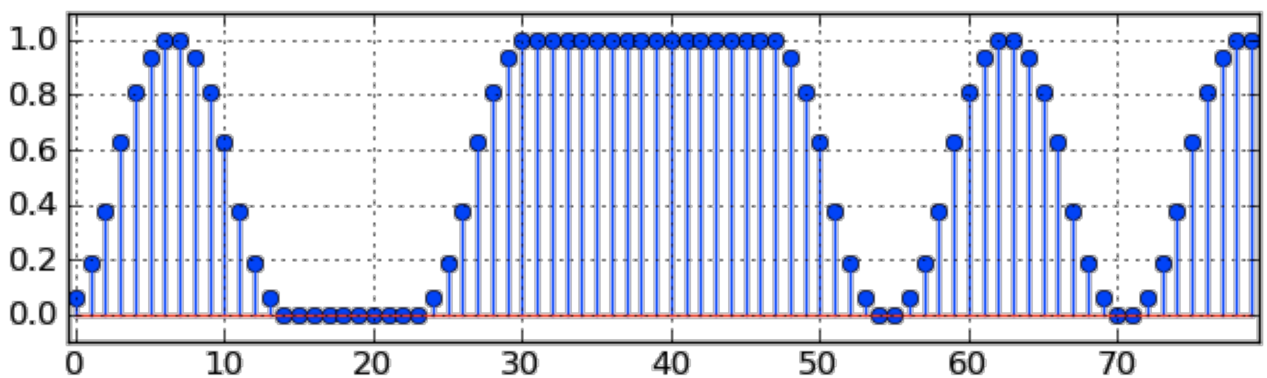


Transmission over a Channel

Signal x[n] from digitized samples at transmitter



Example of distorted noise-free signal y[n] at receiver



Modulation (at the Transmitter)

Think of this as adapting the digitized signal x[n] to the characteristics of the channel.

e.g., acoustic channel from laptop speaker to microphone is *not* well suited to transmitting *constant* levels v_0 and v_1 to represent 0 and 1. So instead transmit **sinusoidal** pressure-wave signals proportional to speaker voltages

$$v_0 \cos(2\pi f_c t) \quad \text{and} \quad v_1 \cos(2\pi f_c t)$$

where f_c is the *carrier frequency* (e.g., 2kHz; wavelength at 340 m/s = 17cm, comparable with speaker dimensions) and

$$v_0 = 0 \qquad v_1 = V > 0$$

(on-off or *amplitude* keying)

or alternatively

$$v_0 = -V \qquad v_1 = V > 0$$

(polar or *phase-shift* keying)

Could also key the *frequency*.

Demodulation & Filtering (at the Receiver)

Demodulation:

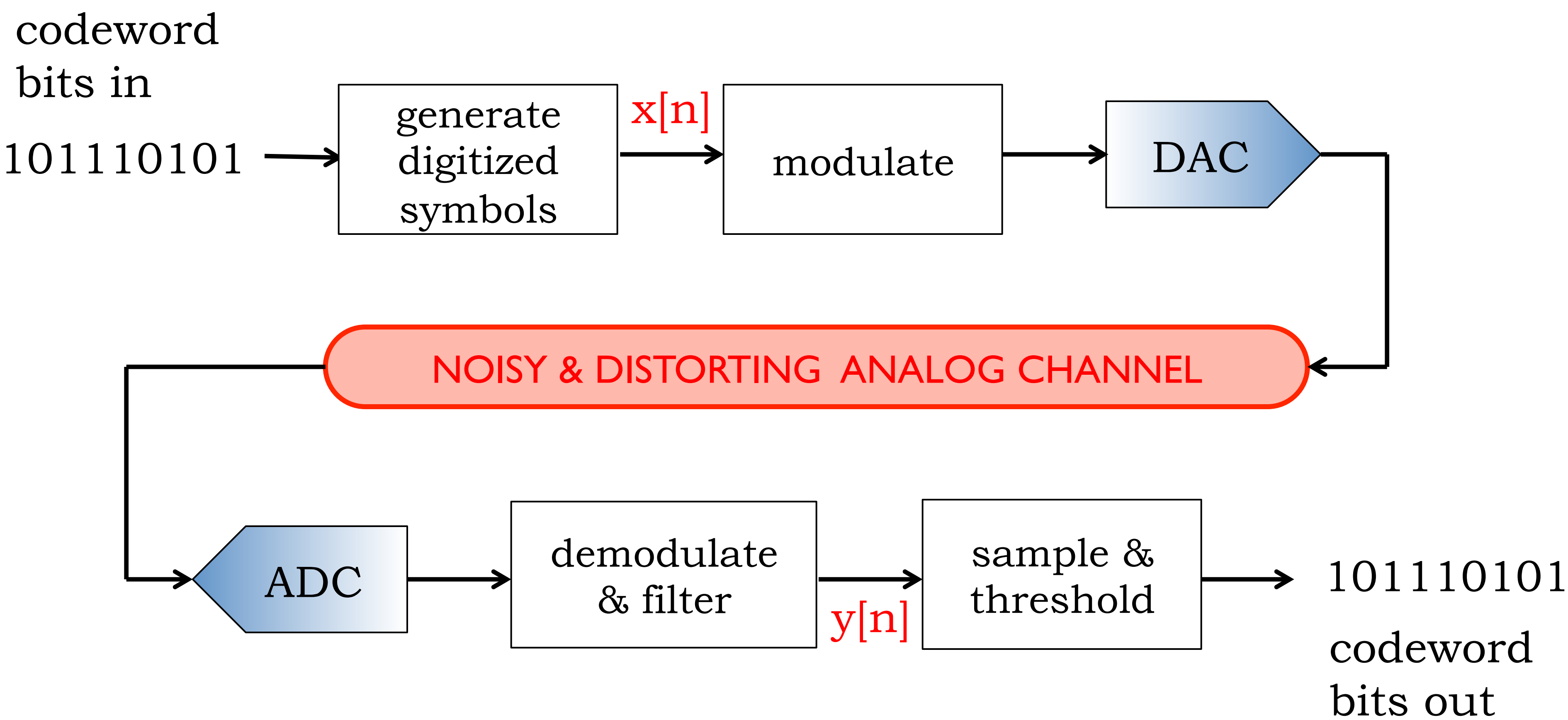
Undo the modulation.

Filtering:

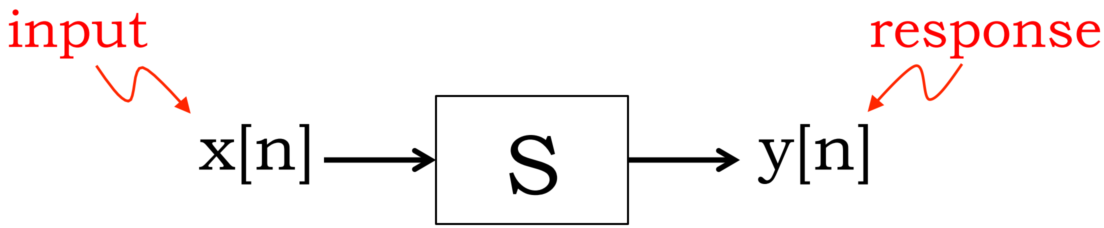
Process the received signal to separate the underlying source signal from channel noise as much as possible. Also mitigate the effects of channel distortion.

(More on these later)

Modeling Channel Behavior



System Input and Output



A discrete-time signal such as x[n] or y[n] is described by an infinite sequence of values, i.e., the time index n takes values in $-\infty$ to $+\infty$. The above picture is a snapshot at a particular time n.

In the diagram above, the sequence of *output* values y[.] is the *response* of system S to the *input* sequence x[.]

Question: Why didn't I write:

"In the diagram above, the sequence of *output* values y[n] is the *response* of system S to the *input* sequence x[n]" ??

Notation, Notation!

--We want to be clear, but being overly explicit about things leads to a lot of notational clutter. So we take shortcuts and liberties, "abusing" and "overloading" the notation, in the hope that context and other factors will make our meaning clear.
--But poor notation can also impede, mislead, confuse! So one has draw the line carefully.

Example: our hard-working discrete-time index n (in continuous-time, it's t). Specifically, x[n] can denote

(a) the value of the signal x at a particular time n

(b) the sequence of values for n in $-\infty$ to $+\infty$, i.e., the entire signal x.

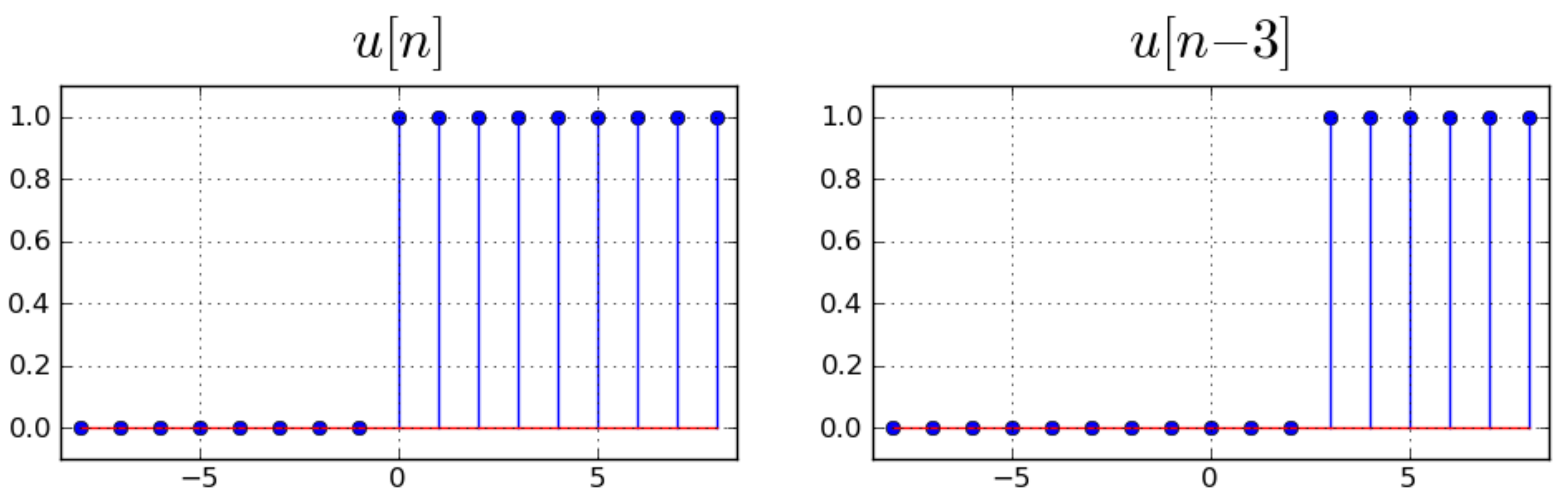
For (b), it's often clearer to write x[.] or just x --- particularly if there are multiple signals involved, because the same "dummy index" n shouldn't be used for both.

On the other hand, if you want to use x[n] for a *specific* value of time, it's sometimes clearer to write $x[n_0]$

Unit Step

A simple but useful discrete-time signal is the *unit step* signal or function, u[n], defined as

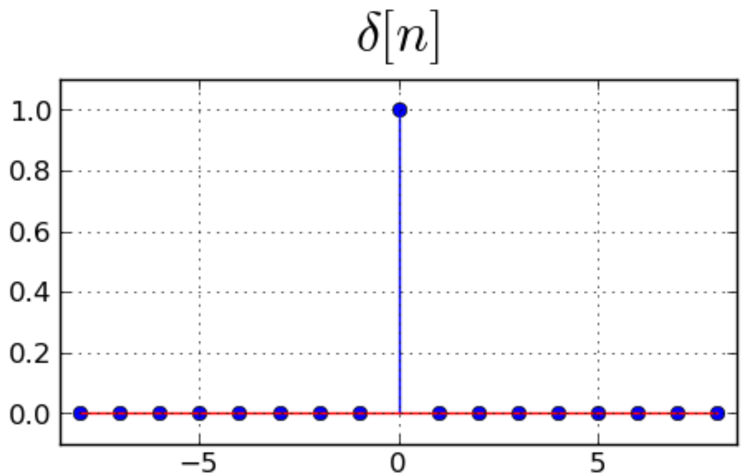
$$u[n] = \begin{cases} 0, & n < 0 \\ 1, & n \geq 0 \end{cases}$$

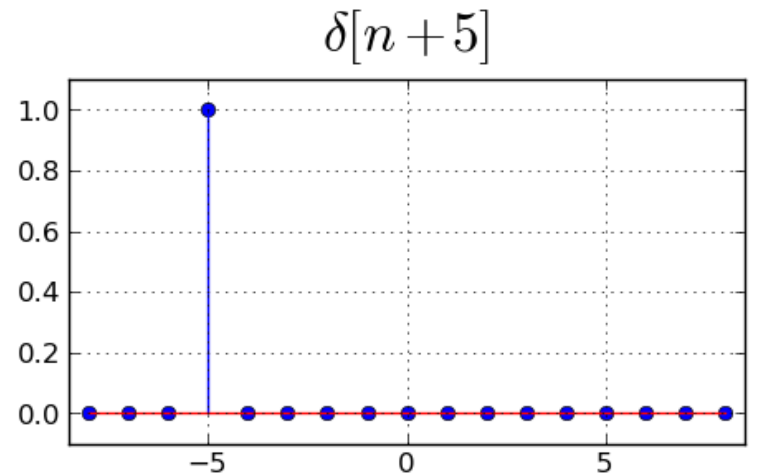


Unit Sample

Another simple but useful discrete-time signal is the *unit sample* signal or function, δ[n], defined as

$$\delta[n] = u[n] - u[n-1] = \begin{cases} 0, & n \neq 0 \\ 1, & n = 0 \end{cases}$$





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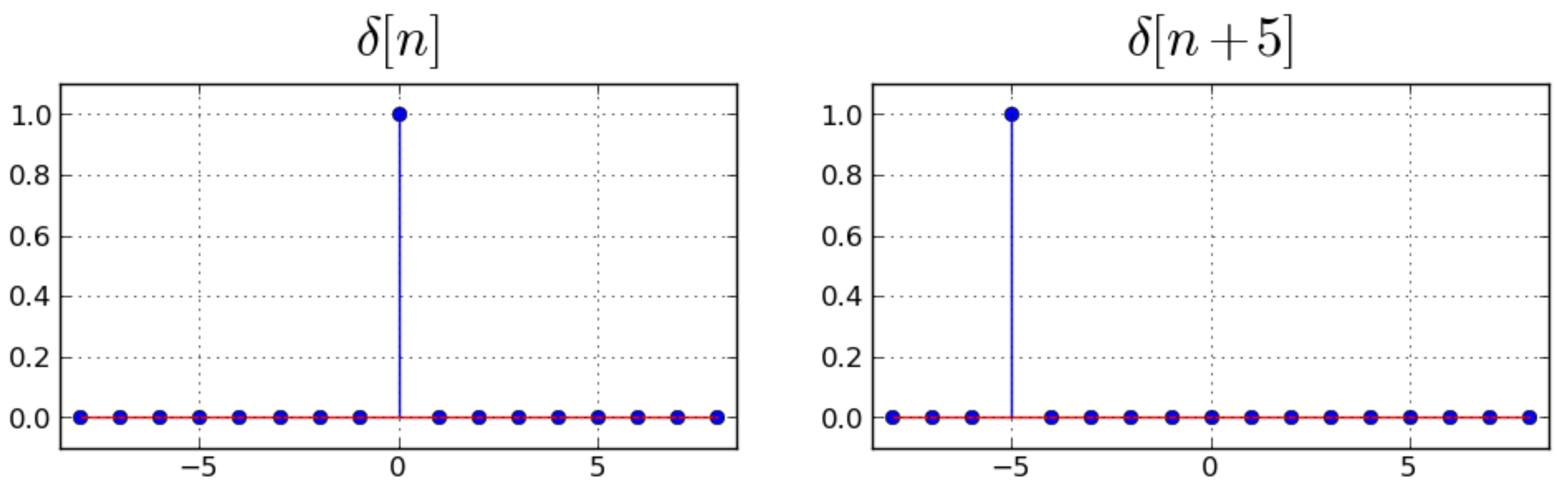
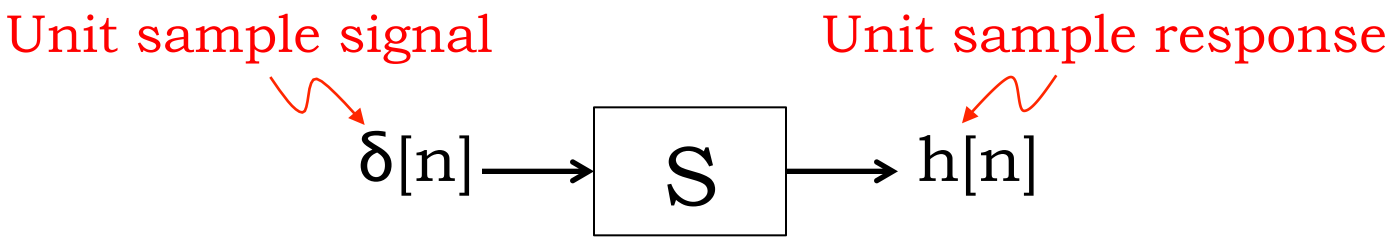


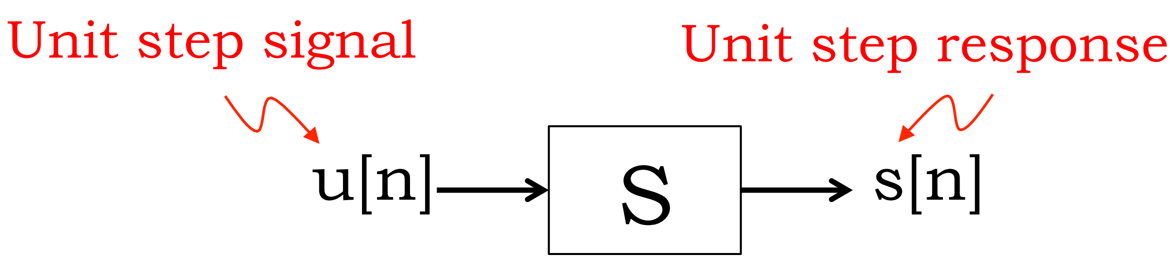
Note that standard algebraic operations on signals (e.g. subtraction, addition, scaling by a constant) are defined in the obvious way, instant by instant.

Unit Sample Response & Unit Step Response

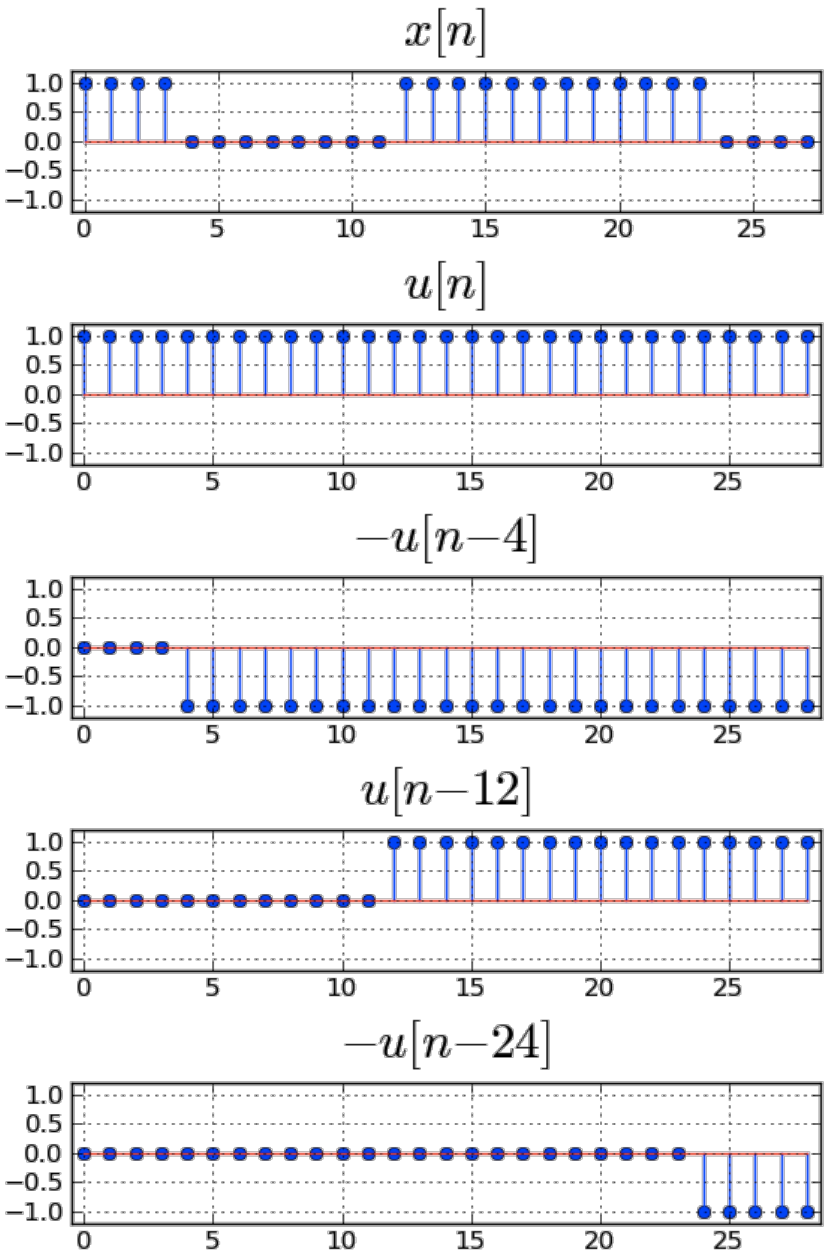


The *unit sample response* of a system S is the response of the system to the unit sample input. We will typically denote the unit sample response as h[n].

Similarly, the *unit step response* s[n]:



Unit Step Decomposition



"Rectangular-wave" digital signaling waveforms, of the sort we have been considering, are easily decomposed into time-shifted, scaled unit steps (each transition corresponds to another shifted, scaled unit step).

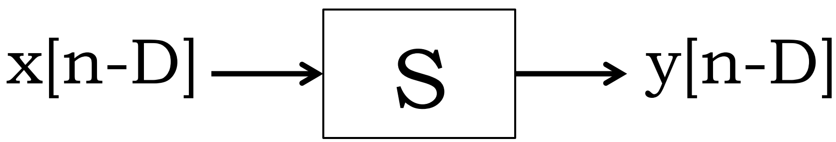
In this example, x[n] is the transmission of 1001110 using 4 samples/bit:

$$x[n] = u[n] - u[n-4] + u[n-12] - u[n-24]$$

Time Invariant Systems

Let y[.] be the response of S to input x[.]

If for all possible sequences x[n] and integers D



then system S is said to be *time invariant* (TI). A time shift in the input sequence to S results in an identical time shift of the output sequence.

In particular, for a TI system, a shifted unit sample function $\delta[n-D]$ at the input generates an identically shifted unit sample response $h[n-D]$ at the output.

Linear Systems

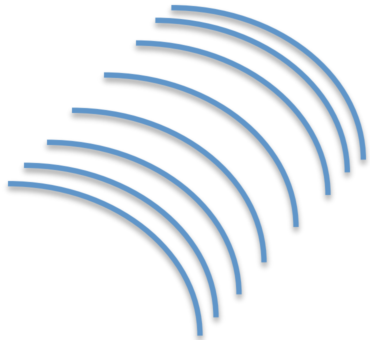
Let $y_1[.]$ be the response of S to input $x_1[.]$, and $y_2[.]$ be the response to $x_2[.]$

If the response to linear combinations of these two inputs equals the same linear combination of the individual responses, then system S is said to be *linear*.

$$a_1x_1[n]+a_2x_2[n] \longrightarrow \boxed{\;S\;} \longrightarrow a_1y_1[n]+a_2y_2[n]$$

If the input is the weighted sum of several signals, the response is the corresponding *superposition* (i.e., weighted sum) of the response to those signals.

Let's explore acoustic transmission in this room



Many thanks to **Keith Winstein** for his extensive work on the acoustic channel platform for 6.02 and for today's demo!