

EE327 Digital Signal Processing

Convolution Properties

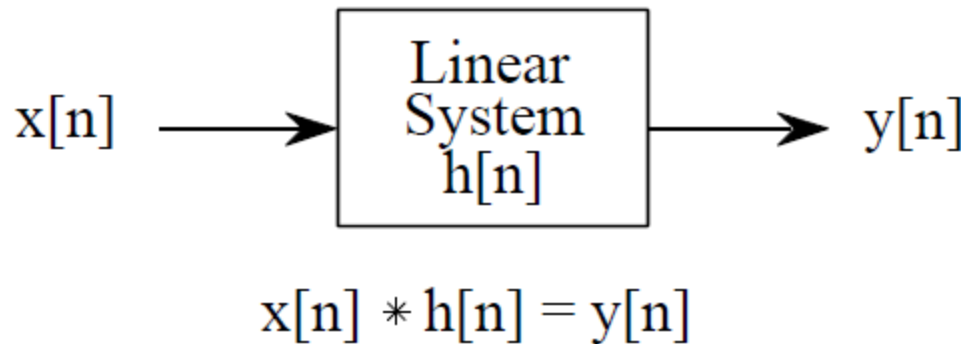
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REMINDER 1: What is convolution?

- A mathematical operation that takes two signals and produces a third one.
 - $X[n]*Y[n]=Z[n]$
- For us:
 - A way to get the output signal given the input signal and a representation of system function

From now on we will deal only with discrete signals if not otherwise specified

REMINDER 2: How to calculate the output



- Input length = N
- Impulse Response length = M
- Output length = $N+M-1$
- For example a 81 points input convolved with a 31 points impulse response gives 111 points output

REMINDER 3: Two ways to understand it

- Input Signal Viewpoint (*Input Side Algorithm*)
 - How each input impulse contributes to the output signal.
 - Good for your understanding
- Output Signal Viewpoint (*Output Side Algorithm*)
 - How each output impulse is calculated from input signal.
 - Good for your calculator

REMINDER 4: Input Side Algorithm

```
100 'CONVOLUTION USING THE INPUT SIDE ALGORITHM
110 '
120 DIM X[80] 'The input signal, 81 points
130 DIM H[30] 'The impulse response, 31 points
140 DIM Y[110] 'The output signal, 111 points
150 '
160 GOSUB XXXX 'Mythical subroutine to load X[ ] and H[ ]
170 '
180 FOR I% = 0 TO 110 'Zero the output array
190 Y(I%) = 0
200 NEXT I%
210 '
220 FOR I% = 0 TO 80 'Loop for each point in X[ ]
230 FOR J% = 0 TO 30 'Loop for each point in H[ ]
240 Y[I%+J%] = Y[I%+J%] + X[I%]*H[J%]
250 NEXT J%
260 NEXT I% '(remember, * is multiplication in programs!)
270 '
280 GOSUB XXXX 'Mythical subroutine to store Y[ ]
290 '
300 END
```

REMINDER 5: Output Side Algorithm

```
100 'CONVOLUTION USING THE OUTPUT SIDE ALGORITHM
110      '
120 DIM X[80]      'The input signal, 81 points
130 DIM H[30]     'The impulse response, 31 points
140 DIM Y[110]    'The output signal, 111 points
150      '
160 GOSUB XXXX    'Mythical subroutine to load X[ ] and H[ ]
170      '
180 FOR I% = 0 TO 110      'Loop for each point in Y[ ]
190   Y[I%] = 0           'Zero the sample in the output array
200   FOR J% = 0 TO 30    'Loop for each point in H[ ]
210    IF (I%-J% < 0)     THEN GOTO 240
220    IF (I%-J% > 80)    THEN GOTO 240
230    Y(I%) = Y(I%) + H(J%) * X(I%-J%)
240   NEXT J%
250 NEXT I%
260      '
270 GOSUB XXXX    'Mythical subroutine to store Y[ ]
280      '
290 END
```

Common Impulse Responses

- Identity System: $x[n] * \delta[n] = x[n]$
- Amplifier/Attenuator: $x[n] * k \times \delta[n] = k \times x[n]$
- Delay/Shift: $x[n] * \delta[n + s] = x[n + s]$
- Echo: $x[n] * (\delta[n] + \delta[n + s]) = x[n] + x[n + s]$

Discretizing Calculus

- First Difference :

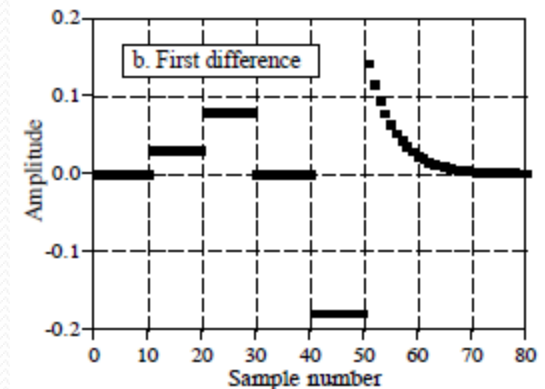
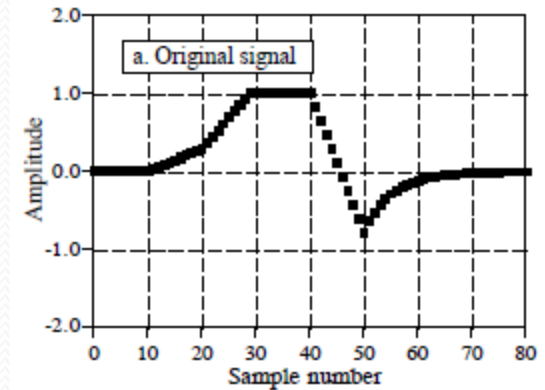
$$y[n] = x[n] - x[n-1]$$

- Discrete equivalent of differentiation
 - Discrete Derivative

- Running Sum:

$$y[n] = \sum_{i=0}^n x[i] = x[n] + y[n-1]$$

- Discrete equivalent of integration
- Discrete Integral



Properties of Convolution

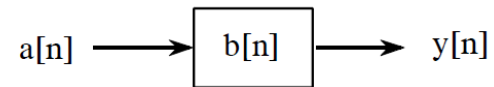
- Commutative Property:

$$a[n] * b[n] = b[n] * a[n]$$

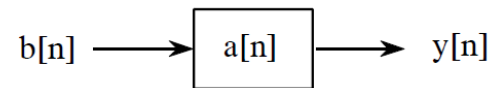
- Associative Property:

$$a[n] * (b[n] * c[n]) = (a[n] * b[n]) * c[n]$$

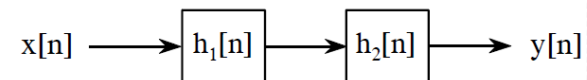
IF



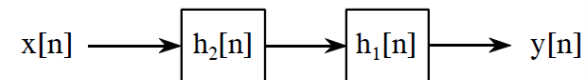
THEN



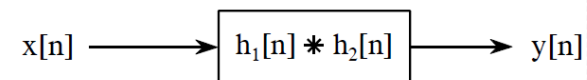
IF



THEN



ALSO



Properties of Convolution (2)

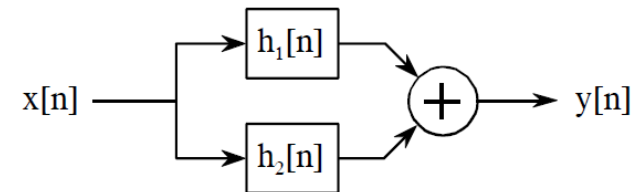
- Distributive Property:

$$a[n] * b[n] + a[n] * c[n] = a[n] * (b[n] + c[n])$$

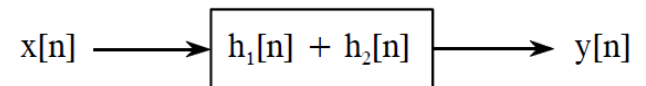
- Central Limit Theory:

$$a[n] * a[n] * a[n] * a[n] * \dots = N(\mu, \sigma)$$

IF

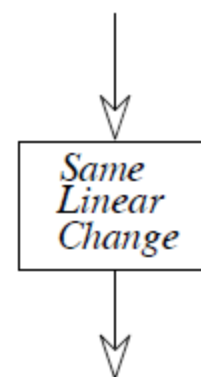
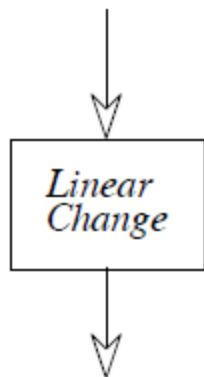
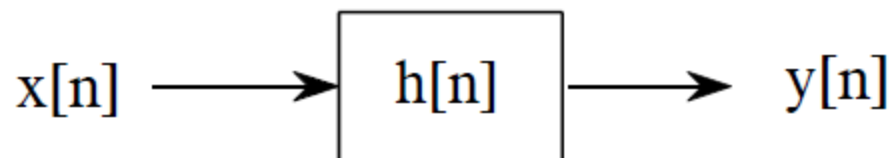


THEN

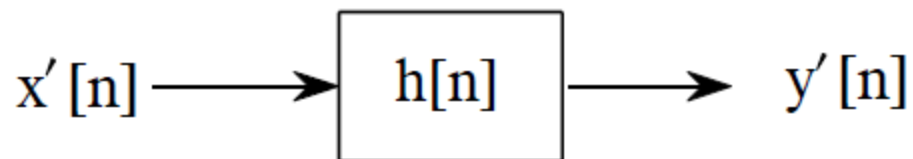


Transference Between Input and Output

IF



THEN



Correlation

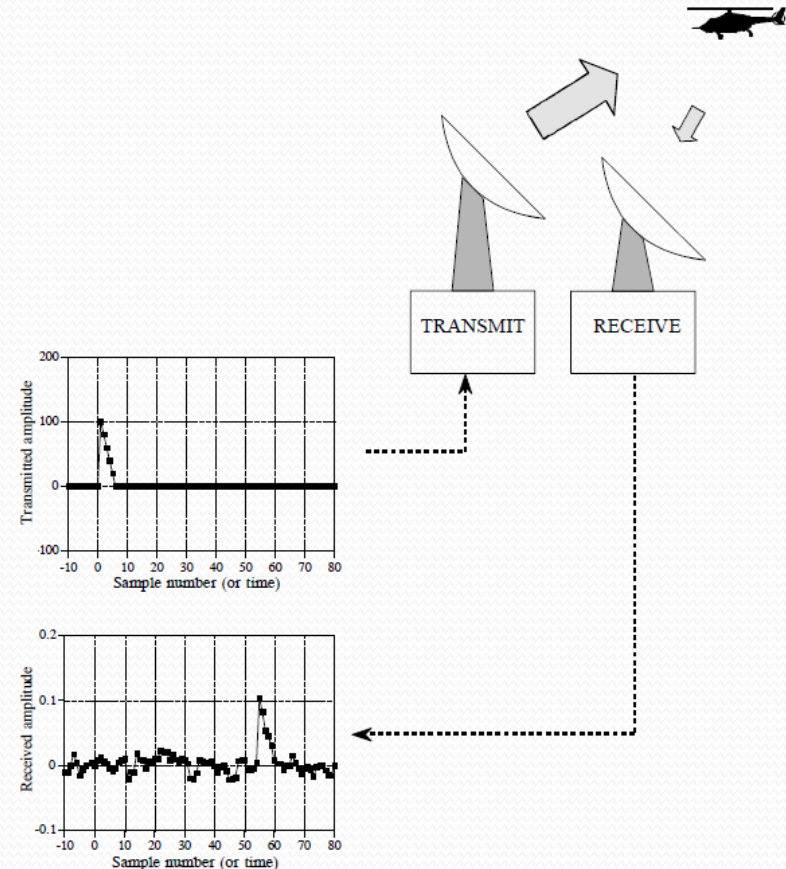
- Cross Correlation

$$c_{ab}[j] = \sum_{i=0}^{M-1} a[j]b[i-j]$$

- Self Correlation

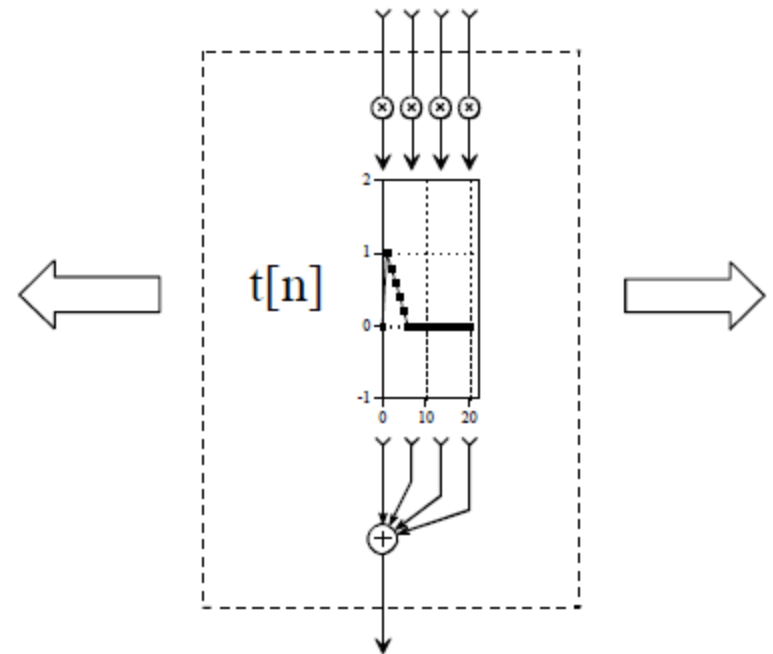
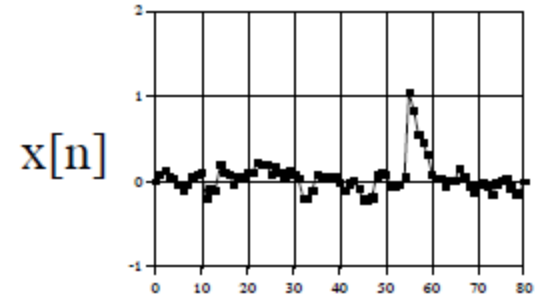
$$c_a[j] = \sum_{i=0}^{M-1} a[j]a[i-j]$$

- Measures similarity (*if correctly normalized!!!!*)



Calculating Correlation

- Convolution without flipping second signal



Just Touching Filters

- Major Types
 - Low Pass Filter
 - High Pass Filter

Low Pass Filters

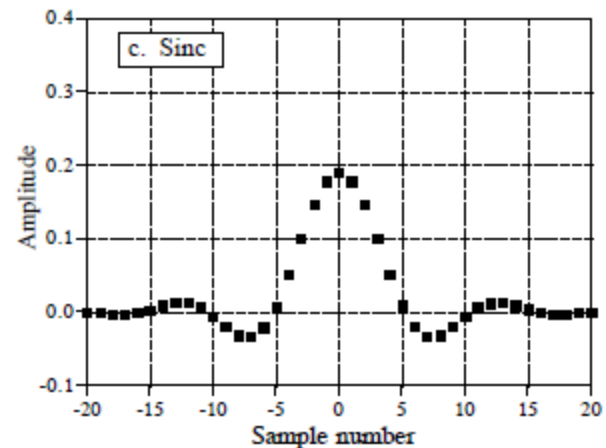
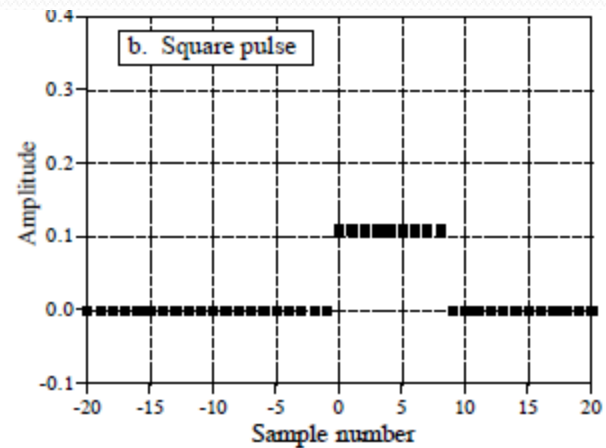
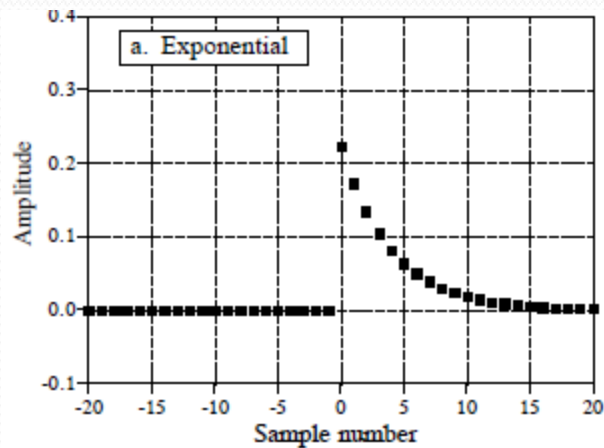


FIGURE 7-4
Typical low-pass filter kernels. Low-pass filter kernels are formed from a group of adjacent positive points that provide an averaging (smoothing) of the signal. As discussed in later chapters, each of these filter kernels is best for a particular purpose. The exponential, (a), is the simplest recursive filter. The rectangular pulse, (b), is best at reducing noise while maintaining edge sharpness. The sinc function in (c), a curve of the form: $\sin(x)/x$, is used to separate one band of frequencies from another.

High Pass Filters

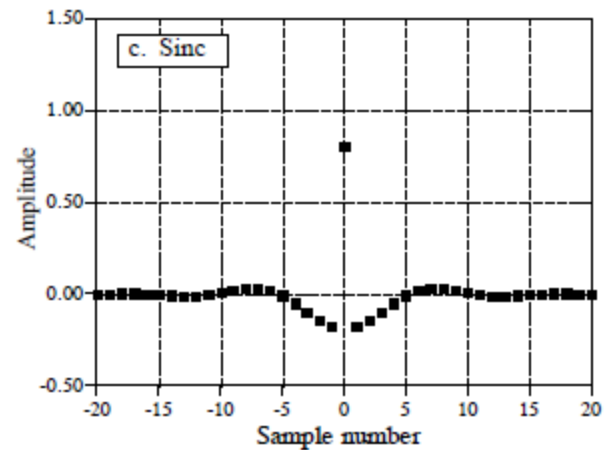
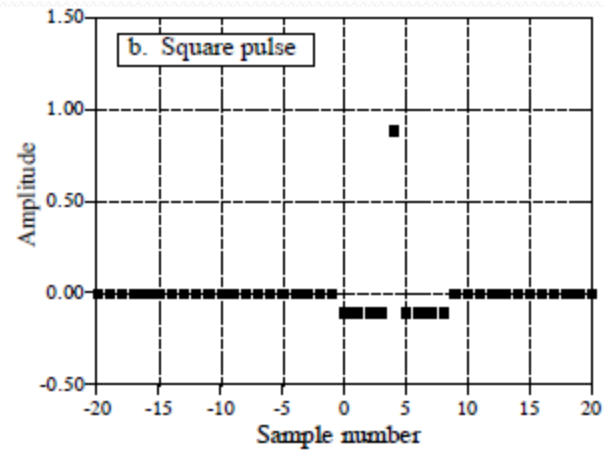
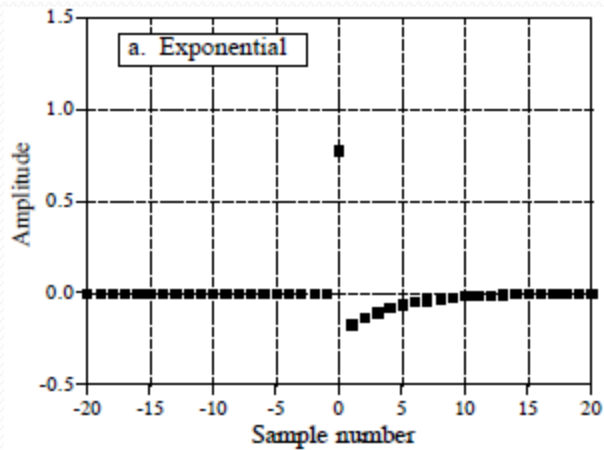


FIGURE 7-5
Typical high-pass filter kernels. These are formed by subtracting the corresponding low-pass filter kernels in Fig. 7-4 from a delta function. The distinguishing characteristic of high-pass filter kernels is a spike surrounded by many adjacent negative samples.

Causal Systems (from Impulse Response)

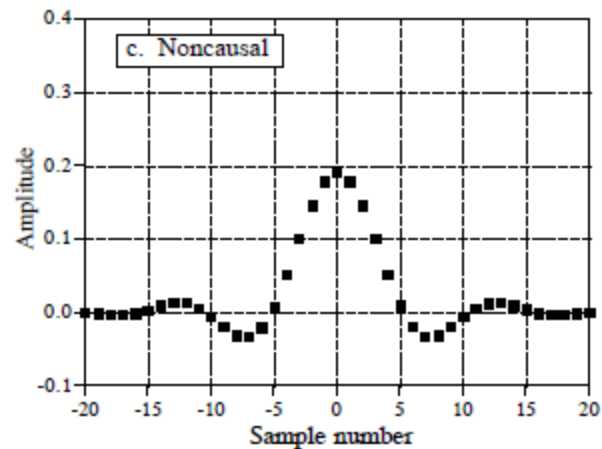
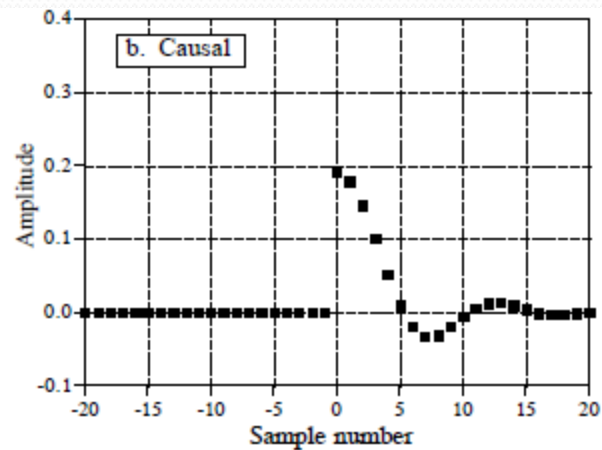
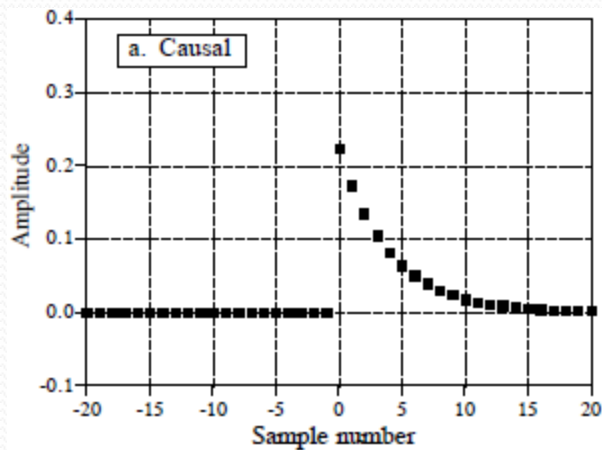
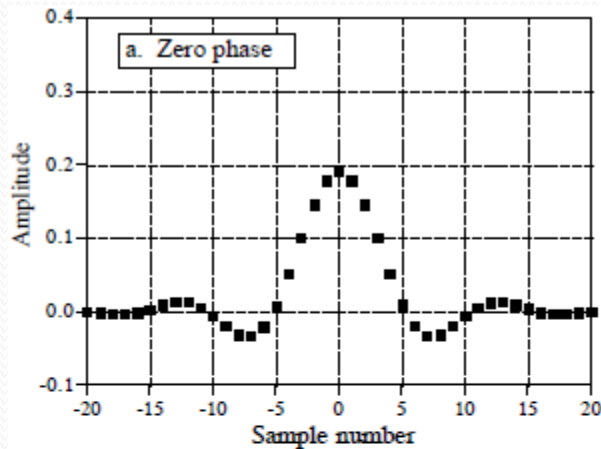


FIGURE 7-6
Examples of causal signals. An impulse response, or any signal, is said to be *causal* if all negative numbered samples have a value of zero. Three examples are shown here. Any noncausal signal with a finite number of points can be turned into a causal signal simply by shifting.

Zero, Linear and nonlinear Phase Systems

Symmetric around 0



Symmetric around another point

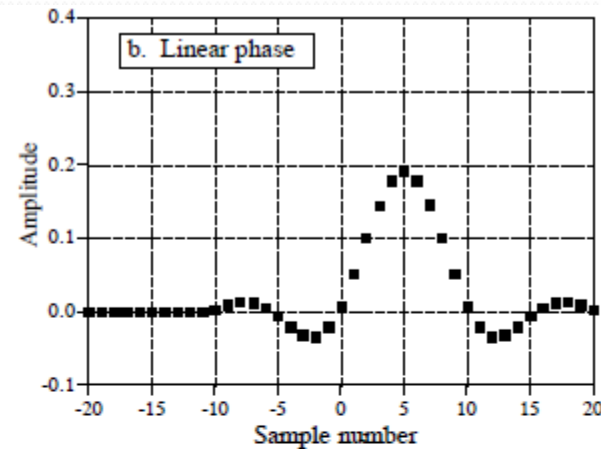
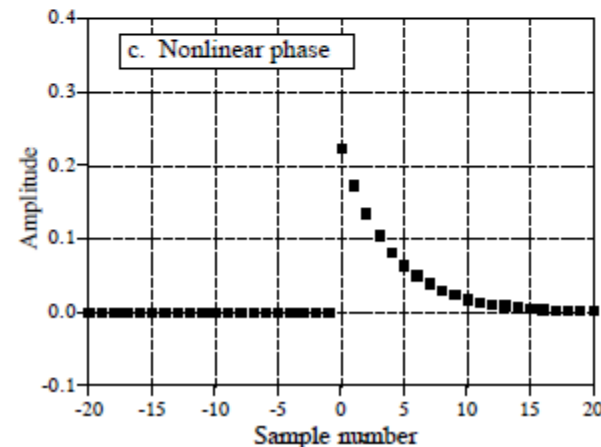


FIGURE 7-7
Examples of phase linearity. Signals that have a left-right symmetry are said to be *linear phase*. If the axis of symmetry occurs at sample number zero, they are additionally said to be *zero phase*. Any linear phase signal can be transformed into a zero phase signal simply by shifting. Signals that do not have a left-right symmetry are said to be *nonlinear phase*. Do not confuse these terms with the *linear* in linear systems. They are completely different concepts.



Asymmetric