Signal processing

Plan

- Linear time-invariant systems
- Finite impulse response filters
- Acoustic echo cancellation
- Microphone array processing
- Beamforming

Linear system

$$y[n] = -\frac{x[n]}{2}$$

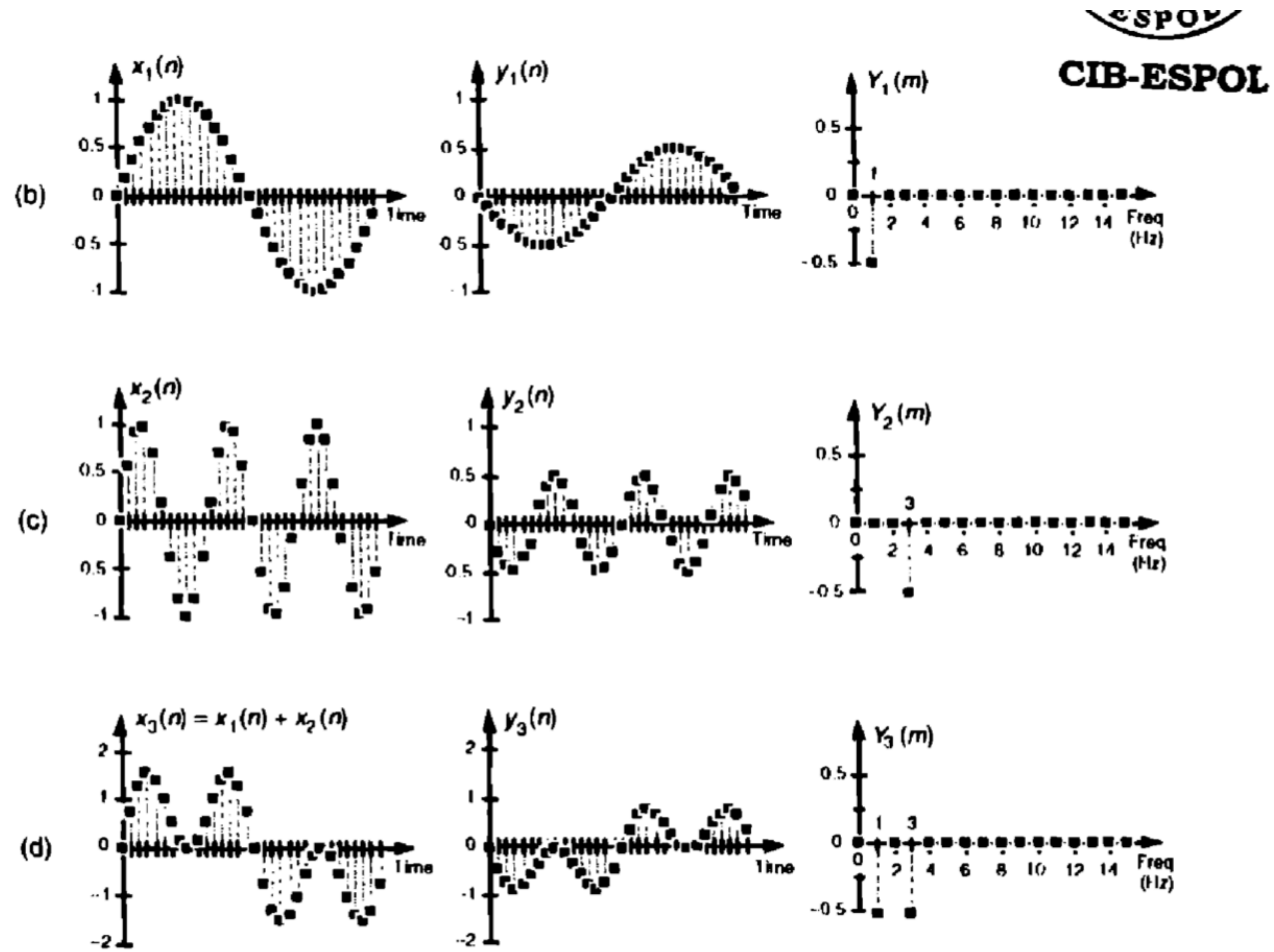


Figure 1-7 Linear system input-to-output relationships: (a) system block diagram where $y(n) = -x(n)/2$; (b) system input and output with a 1-Hz sinewave applied; (c) with a 3-Hz sinewave applied; (d) with the sum of 1-Hz and 3-Hz sinewaves applied.

Time invariant system

$$y[n] = -\frac{x[n]}{2}$$

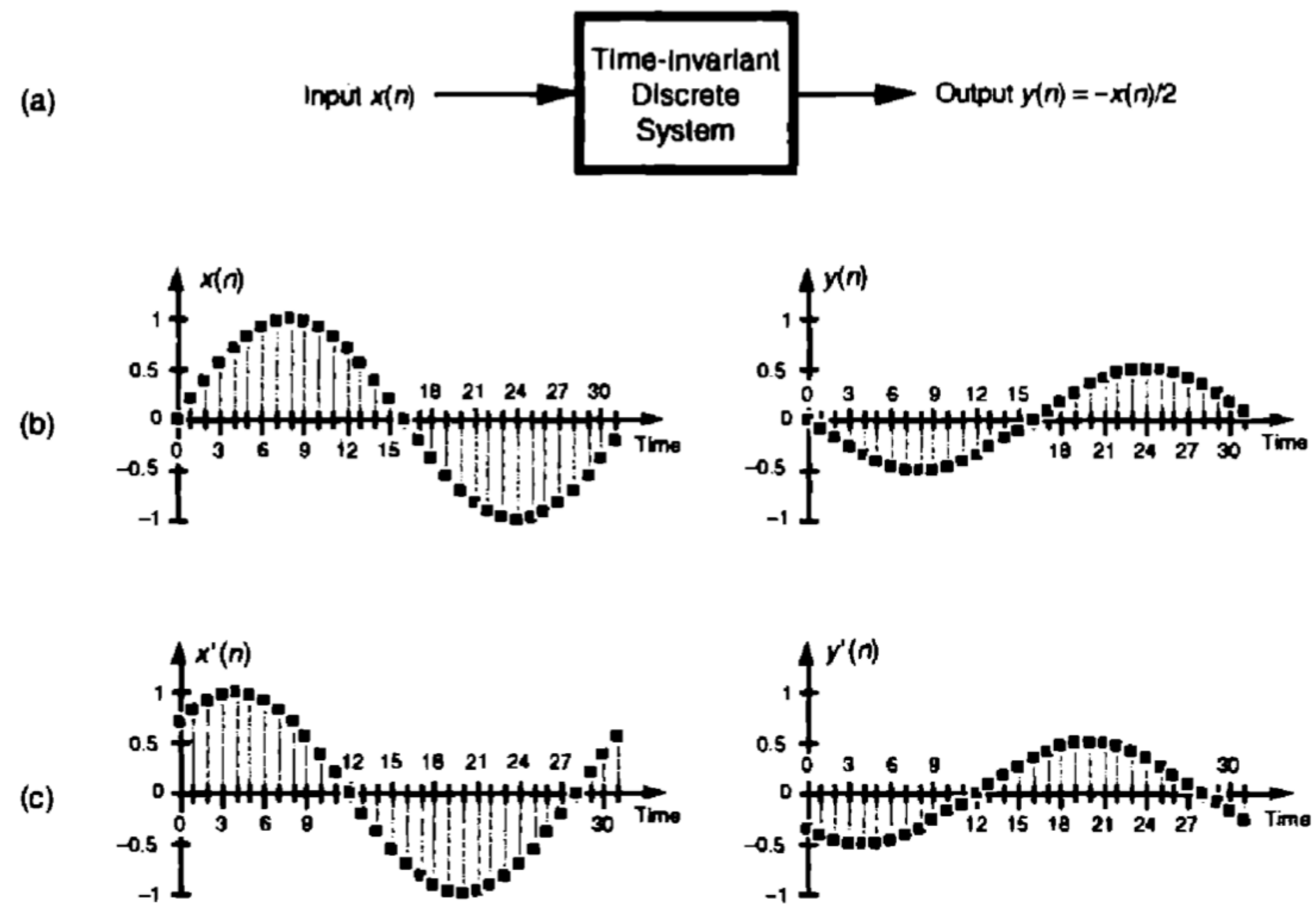


Figure 1-9 Time-invariant system input-to-output relationships: (a) system block diagram where $y(n) = -x(n)/2$; (b) system input and output with a 1-Hz sinewave applied; (c) system input and output when a 1-Hz sinewave, delayed by four samples, is applied. When $x'(n) = x(n+4)$, then, $y'(n) = y(n+4)$.

Impulse response

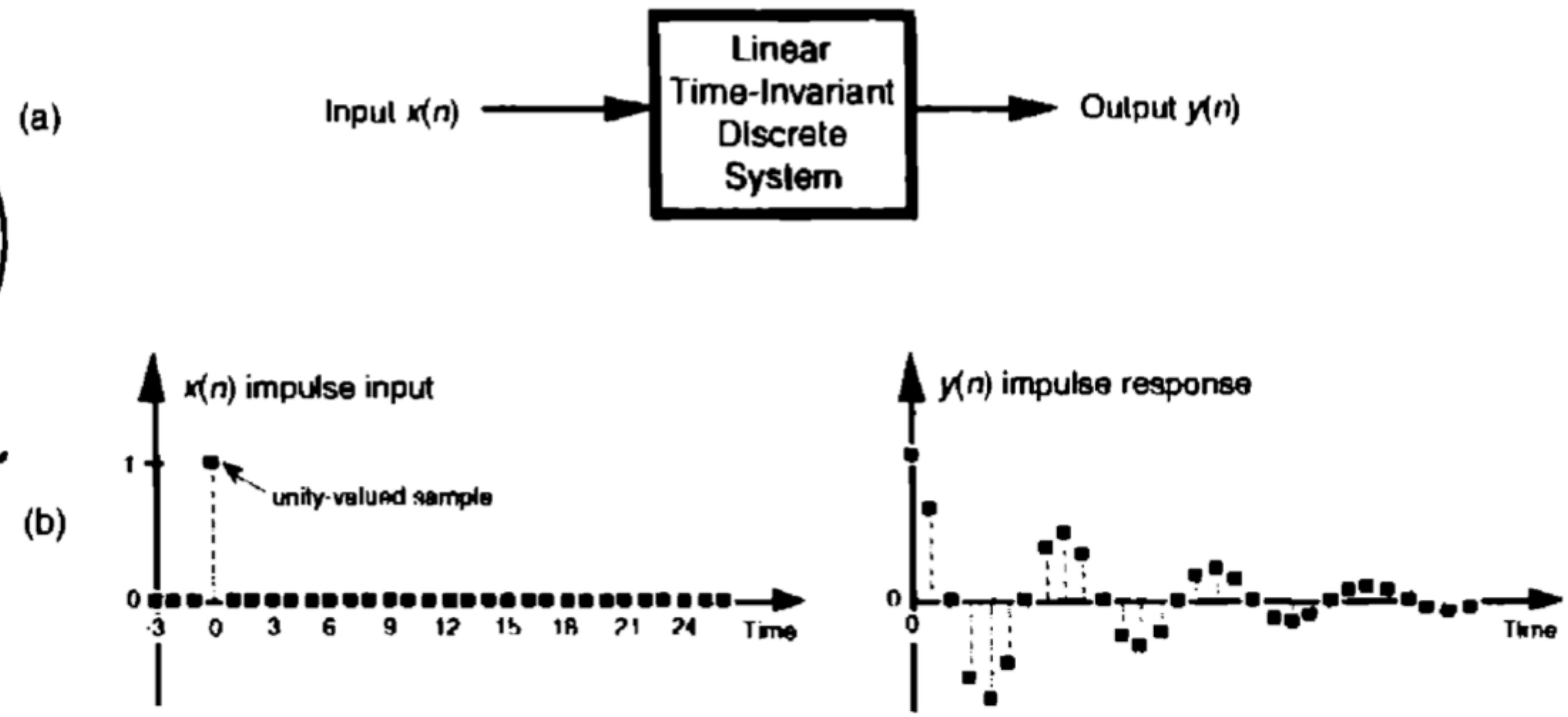


Figure 1-11 LTI system unit impulse response sequences: (a) system block diagram; (b) impulse input sequence $x(n)$ and impulse response output sequence $y(n)$.

The impulse response is sufficient for a complete description of Linear Time-Invariant System

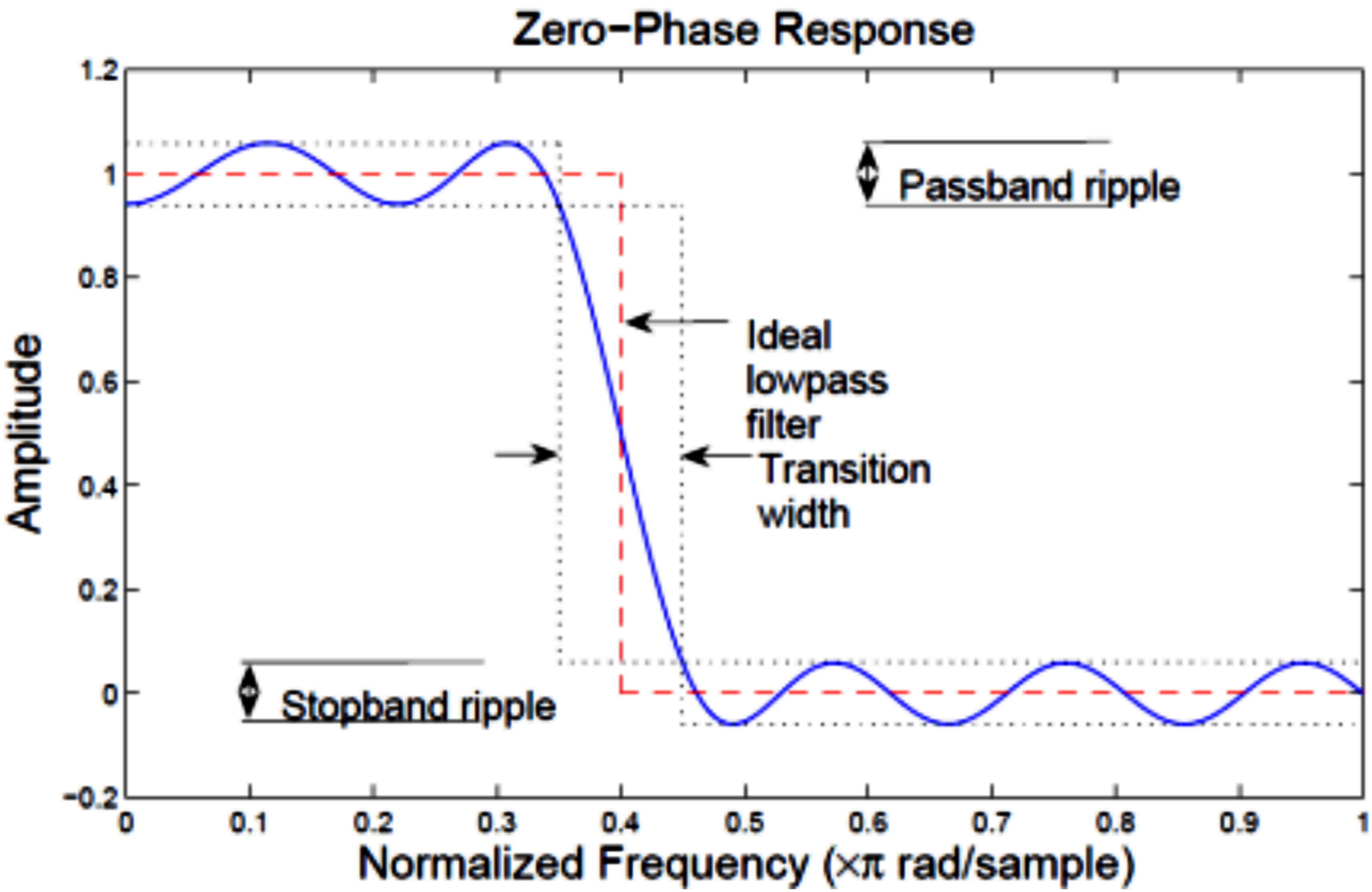
Finite impulse response filters

$$
\begin{aligned}
y[n] &= b_0 x[n] + b_1 x[n-1] + \cdots + b_N x[n-N] \\
&= \sum_{i=0}^{N} b_i \cdot x[n-i],
\end{aligned}
$$

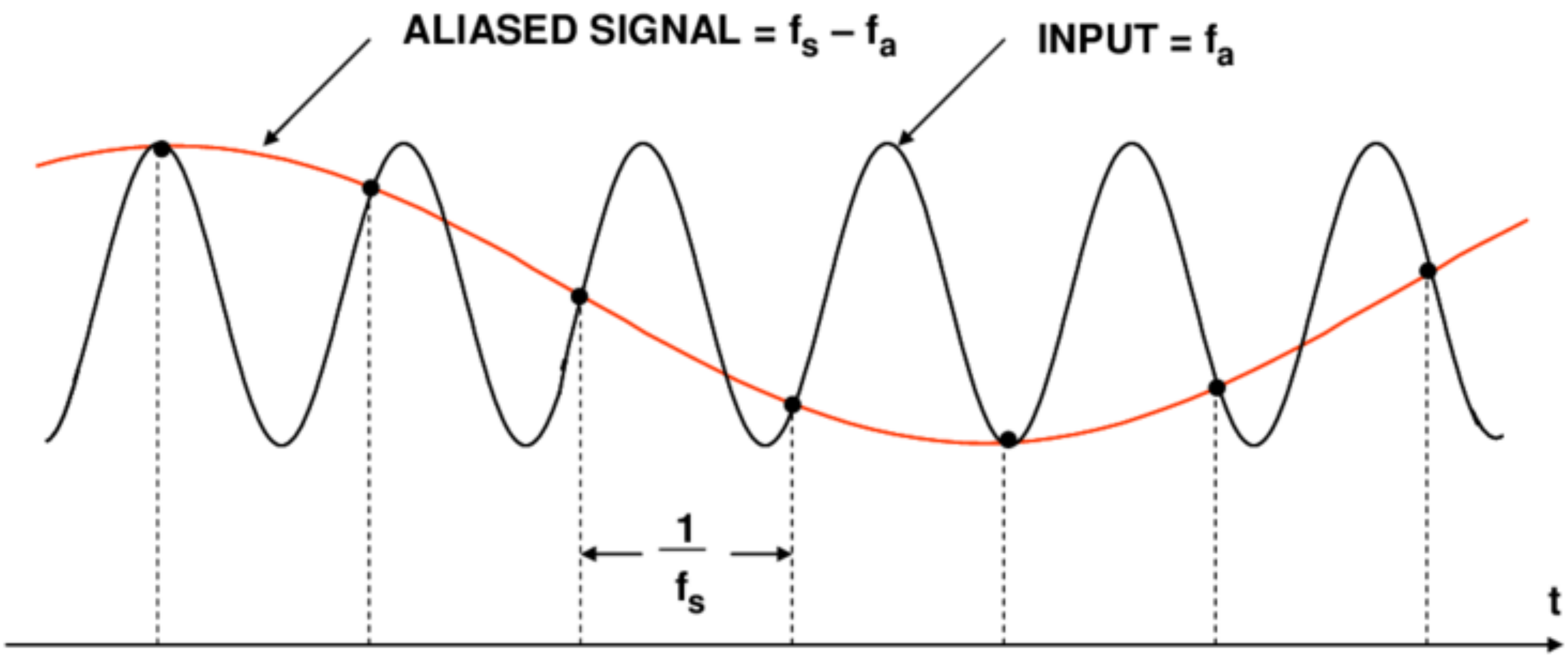
where:

- $x[n]$ is the input signal,
- $y[n]$ is the output signal,
- N is the filter order; an N^{th}-order filter has $N+1$ terms on the right-hand side
- b_i is the value of the impulse response at the *i*th instant for $0 \leq i \leq N$ of an N^{th}-order FIR filter.

Finite impulse response filters

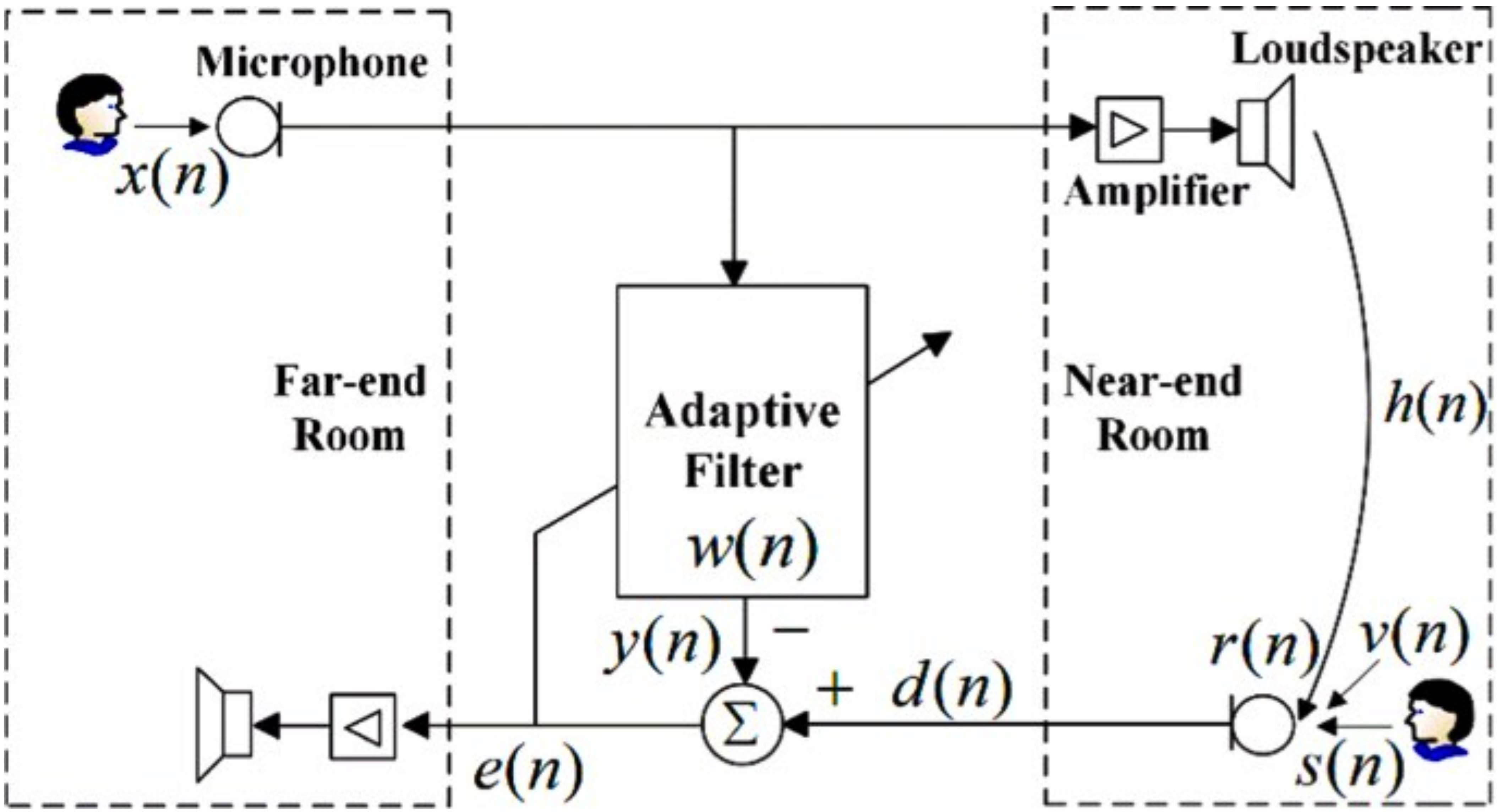


Time domain aliasing

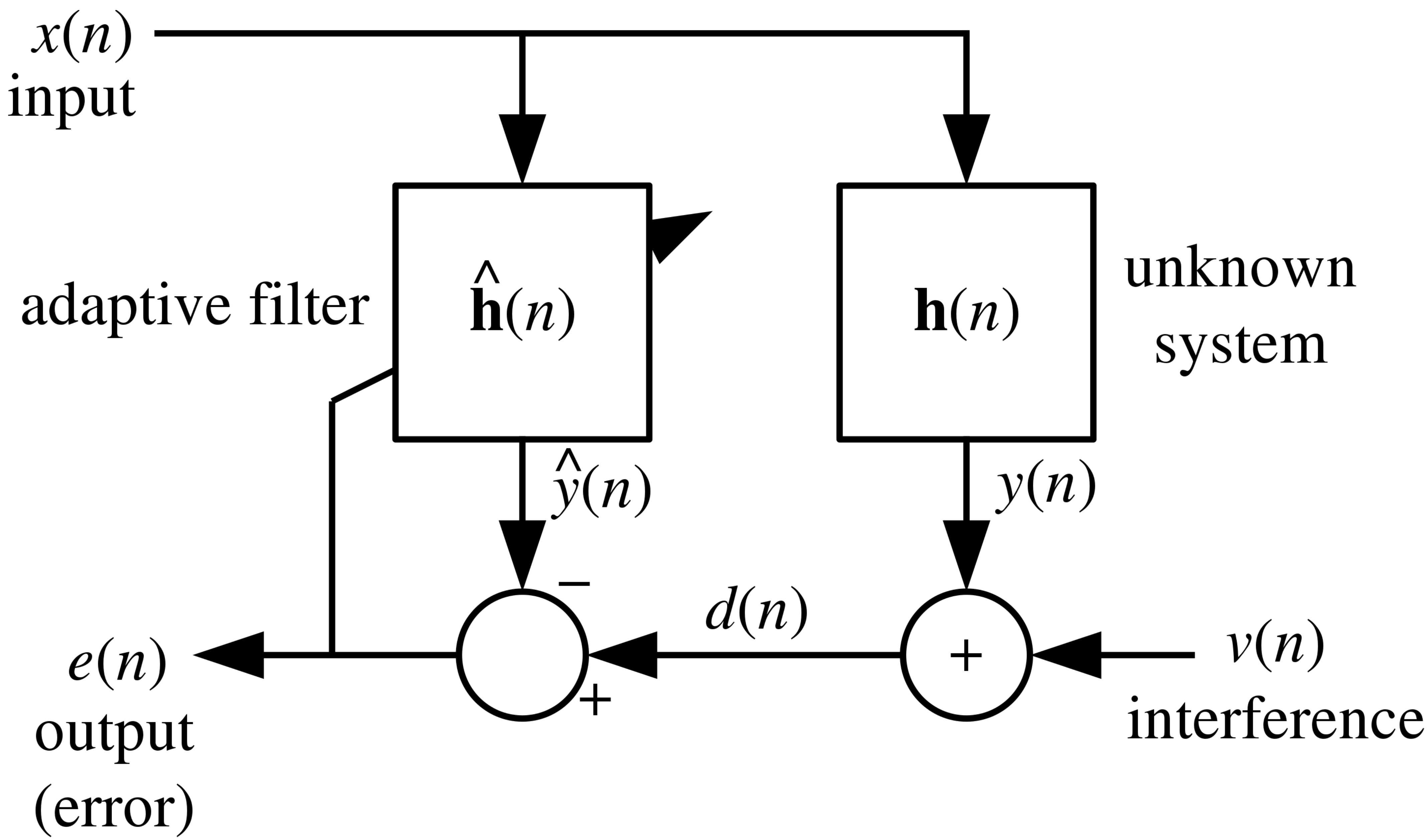


NOTE: f_a IS SLIGHTLY LESS THAN f_s

Acoustic Echo Cancellation



Least mean squares



Least mean squares

The LMS algorithm for a pth order filter can be summarized as

Parameters: $p =$ filter order

$\mu =$ step size

Initialisation: $\hat{\mathbf{h}}(0) = \mathrm{zeros}(p)$

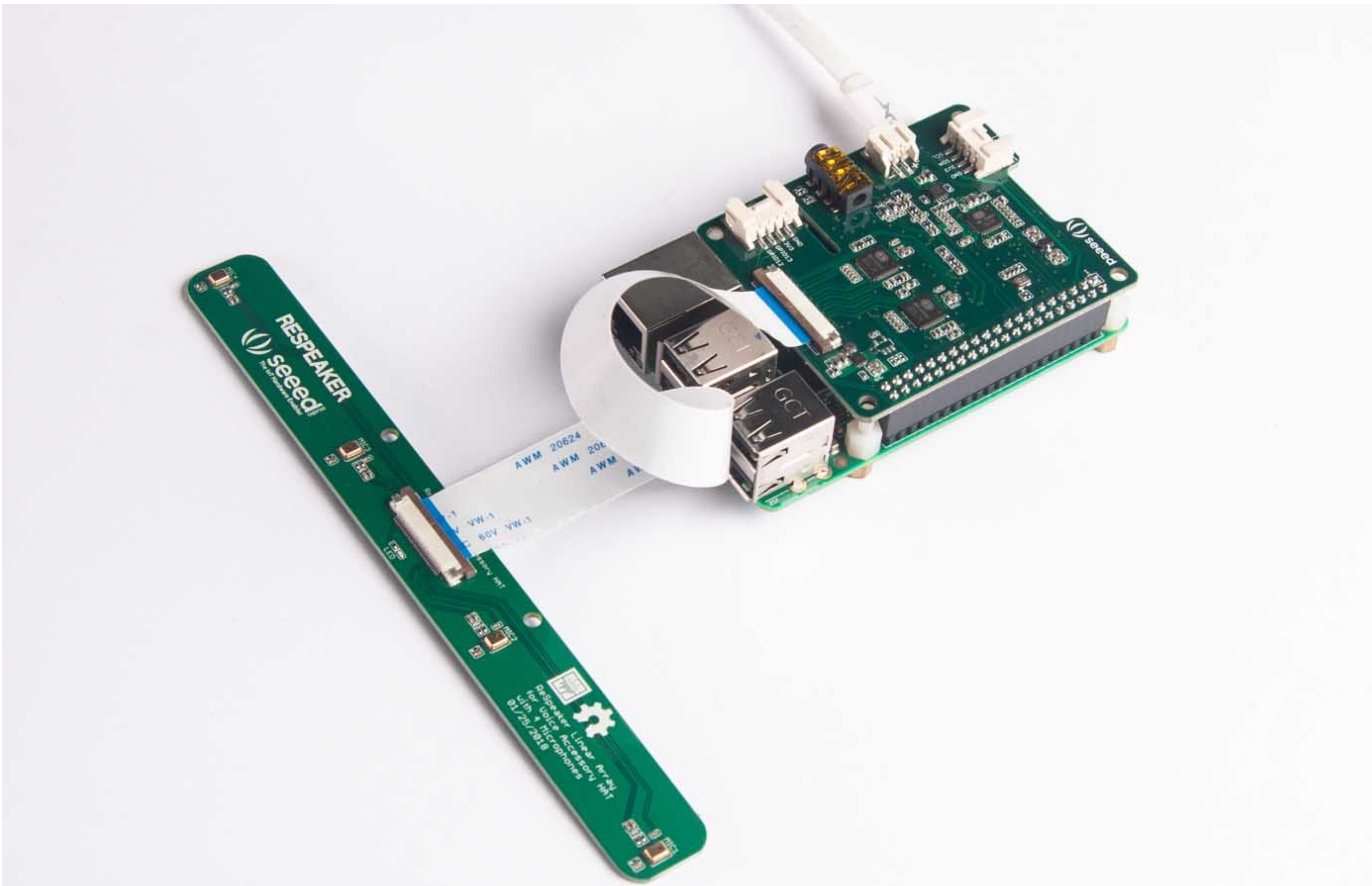
Computation: For $n = 0, 1, 2, \ldots$

$$\mathbf{x}(n) = \left[x(n), x(n-1), \ldots, x(n-p+1)\right]^T$$

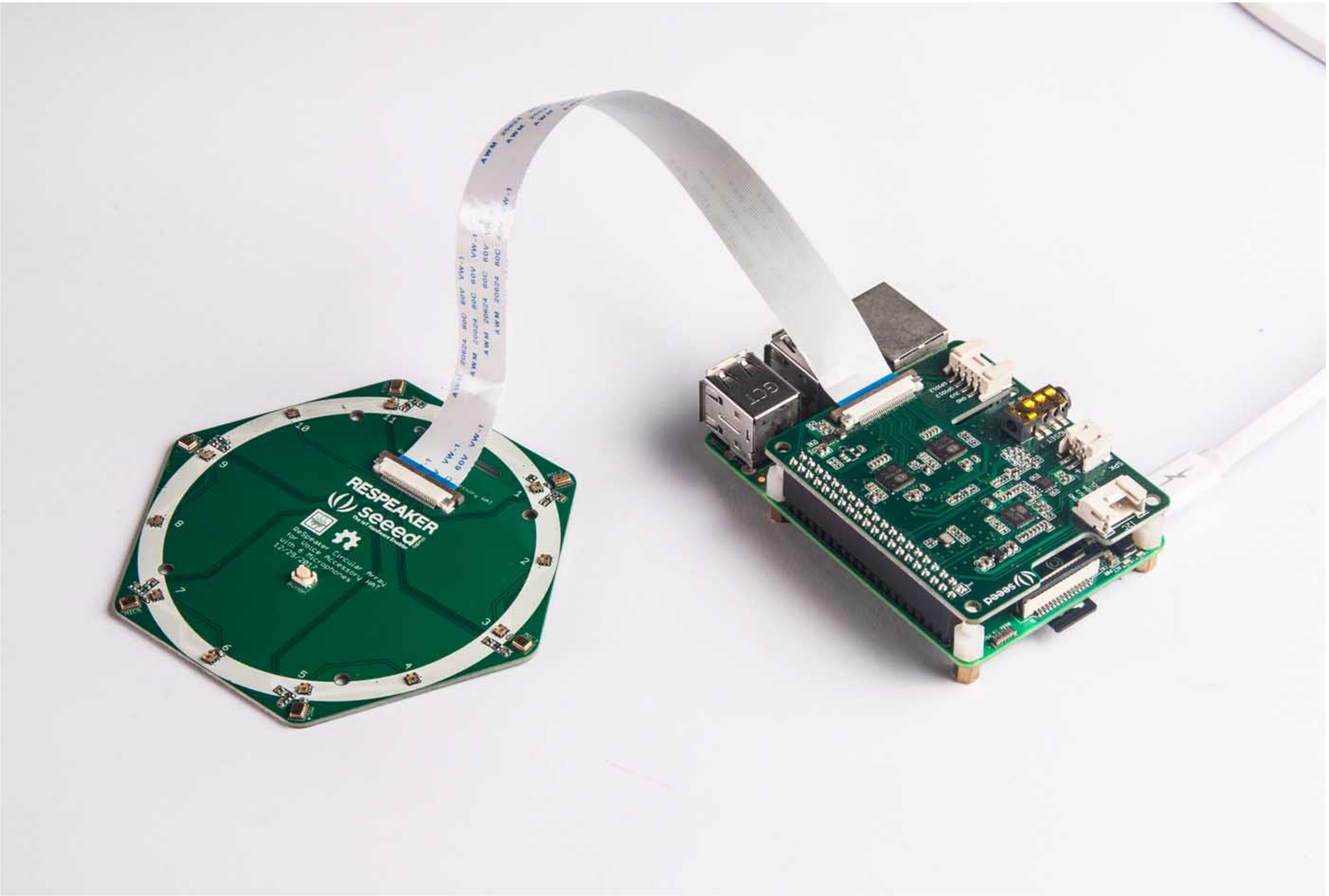
$$e(n) = d(n) - \hat{\mathbf{h}}^H(n)\mathbf{x}(n)$$

$$\hat{\mathbf{h}}(n+1) = \hat{\mathbf{h}}(n) + \mu\, e^*(n)\mathbf{x}(n)$$

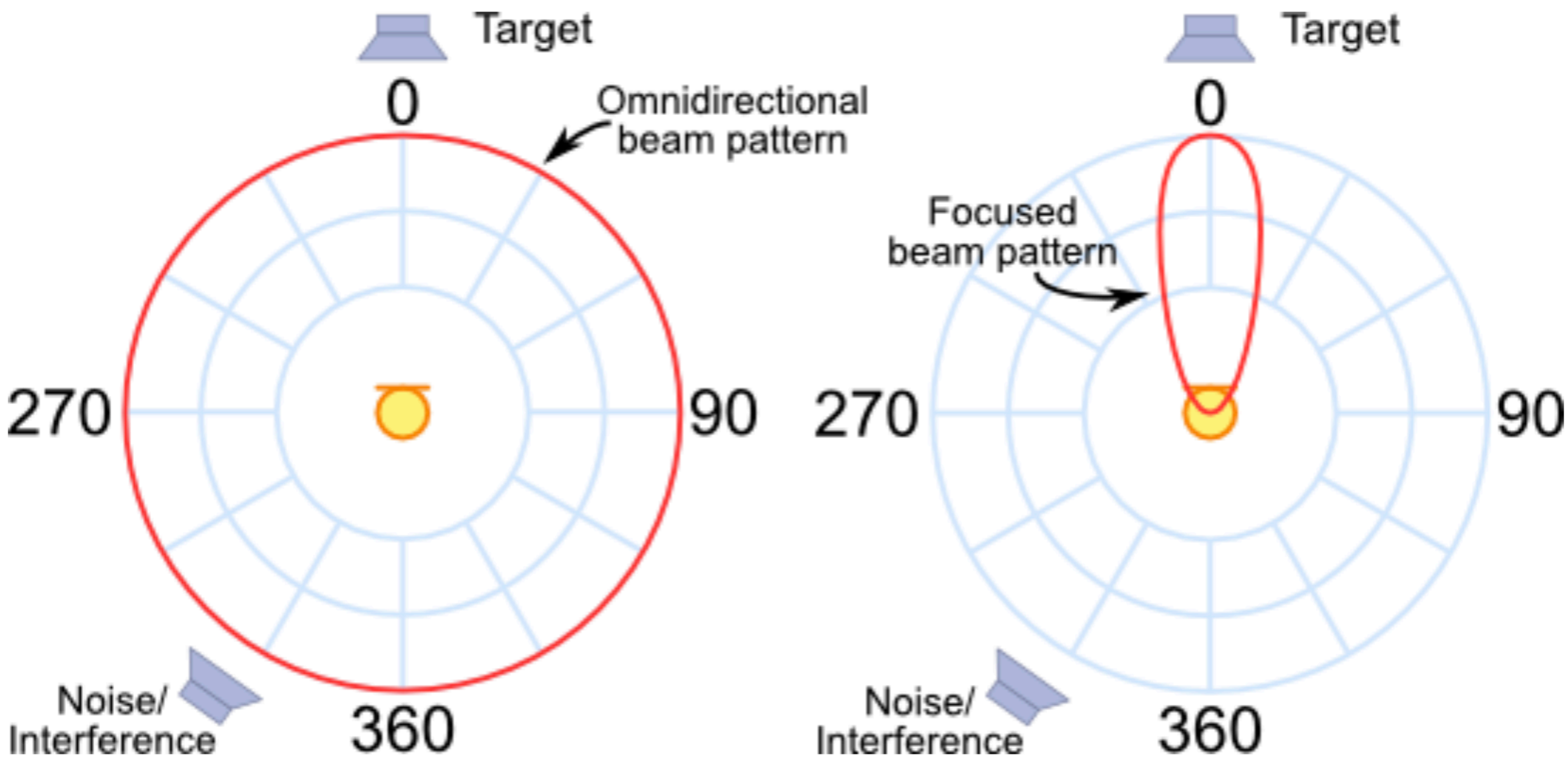
Microphone array processing



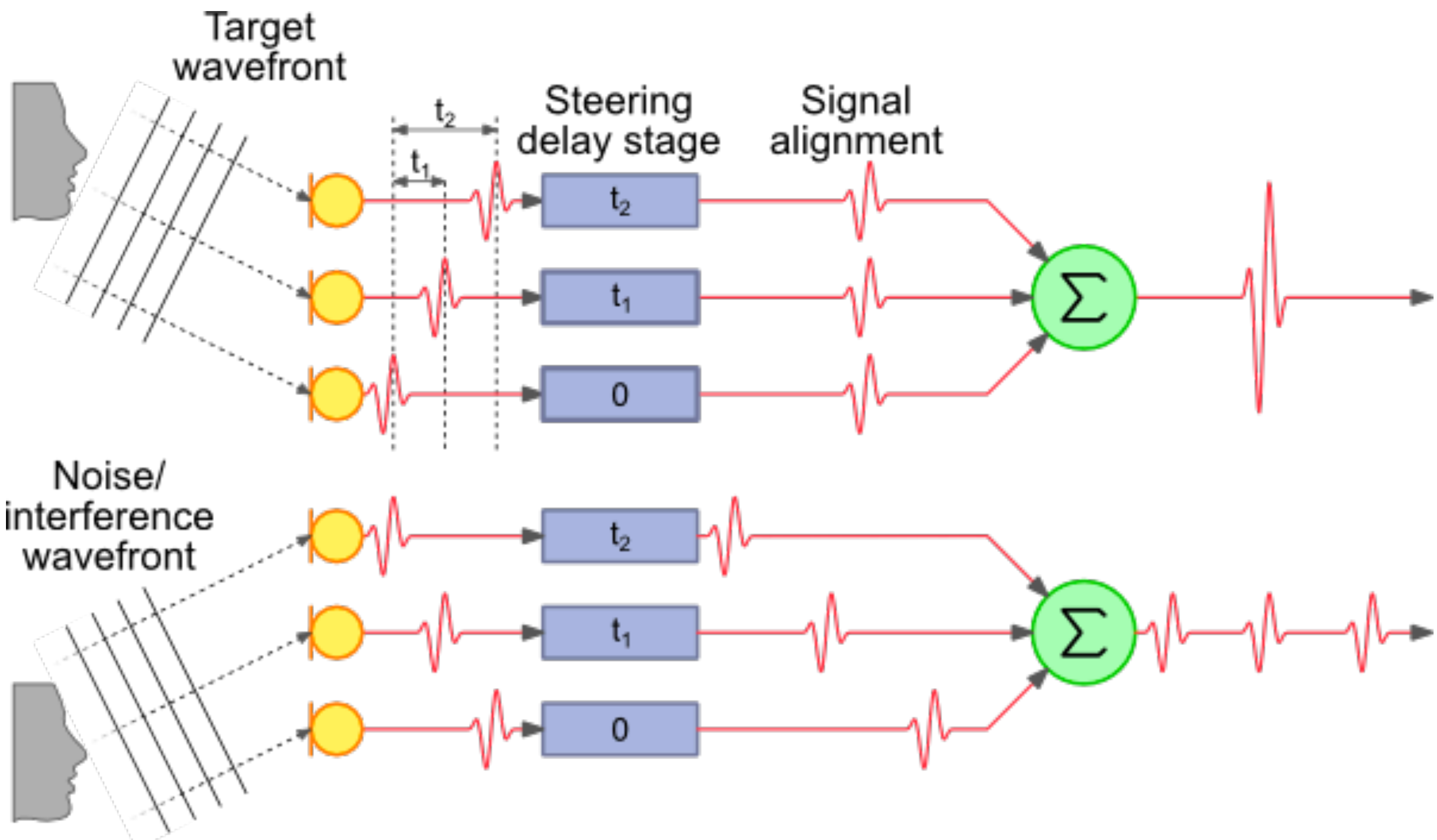
Microphone array processing



Beamforming



Beamforming



Spatial aliasing