## Tune-Up Tuesday #4 Deconvolution for October 9, 2025

Before we talk about deconvolution, let's define convolution. Then, we'll derive a deconvolution algorithm and apply it to two examples. You could have worked either example for the Tune-Up.

**Convolution**. For a finite impulse response filter with M+1 filter coefficients  $b_0, b_1, ..., b_M$ , the output signal y[n] for an input signal x[n] is computed according to

$$y[n] = b_0 x[n] + b_1 x[n-1] + \cdots + b_M x[n-M]$$

If we input the discrete-time impulse signal  $\delta[n]$ , which has value of 1 at n=0 and 0 otherwise, the output is called the *impulse response* (response is synonymous with output):

$$h[n] = b_0 \delta[n] + b_1 \delta[n-1] + \cdots + b_M \delta[n-M]$$

Hence,

$$h[k] = b_k$$
 for  $k = 0, 1, ..., M$ 

Otherwise, h[k] = 0. Because  $h[k] = b_k$  for k = 0, 1, ..., M,

$$y[n] = h[0]x[n] + h[1]x[n-1] + \dots + h[M+1]x[n-(M+1)] = \sum_{k=0}^{M} h[k] x[n-k]$$

This computation is known as convolution of h[n] and x[n]:

$$y[n] = h[n] * x[n]$$

Given input signal x[n] and impulse response h[n], we can compute output signal y[n]. If x[n] is also finite in length, then the length of y[n] will be the length of h[n] plus the length of x[n] minus 1.

**Deconvolution.** Whereas convolution computes the output signal y[n] from input signal x[n] and an impulse response h[n] of a FIR filter, deconvolution seeks to find impulse response h[n] given input signal x[n] and output signal y[n]. We can choose the input signal x[n], also known as a test signal, and observe the output signal.

**Practical scenario**. We would start the test signal and the observation at a particular point in time, which we'll say is at n = 0 without loss of generality. Further, we will assume that x[n] = 0 for n < 0; i.e., x[n] is a causal signal.

**Deconvolution Algorithm.** We'll work backwards in the time domain to compute the FIR filter coefficients. We derive a time-domain deconvolution algorithm by first evaluating the output at n = 0:

$$y[0] = h[0] x[0] + h[1] x[-1] + h[2] x[-2] + \dots + h[M] x[-M]$$

As mentioned above, we'll assume x[n] is a causal signal; i.e., x[n] = 0 for n < 0. Since we know x[n] and y[n], we have one equation and one unknown at n = 0:

$$y[0] = h[0] x[0]$$

and we can compute

$$h[0] = \frac{y[0]}{x[0]}$$

For this calculation to be valid, the first value of the test signal, x[0], cannot be zero.

Second output: 
$$y[1] = h[0] x[1] + h[1] x[0]$$
, and therefore,  $h[1] = \frac{y[1] - h[0] x[1]}{x[0]}$ .

Third output: 
$$y[2] = h[0] x[2] + h[1] x[1] + h[2] x[0]$$
 and  $h[2] = \frac{y[2] - h[0] x[2] - h[1] x[1]}{x[0]}$ 

In general, for the *N*th output,  $h[N] = \frac{y[N] - \sum_{i=0}^{N-1} h[i] x[N-i]}{x[0]}$ .

The MATLAB script <u>utdeconvolve.m</u> implements this algorithm.

**Example**. *Problem 4.3(b)*. In this problem, we're given

- causal input signal x[n] with non-zero values [ 1 2 3 4 5 ]
- causal output signal y[n] with non-zero values [11111-5]

We can compute the FIR filter coefficients using the above deconvolution algorithm:

$$h[0] = \frac{y[0]}{x[0]} = \frac{1}{1} = 1$$

$$h[1] = \frac{y[1] - h[0] x[1]}{x[0]} = \frac{1 - 1 \cdot 2}{1} = -1$$

$$h[2] = \frac{y[2] - h[0] x[2] - h[1] x[1]}{x[0]} = \frac{1 - 1 \cdot 3 - (-1) \cdot 2}{1} = 0$$

The values of h[n] for n > 2 are zero. The MATLAB script <u>utdeconvolve.m</u> will give the same answer for h[n]. We can validate the answer by convolving h[n] and x[n]. We can use the Matlab command conv to do this:

Alternately, we could use the filter command. Keeping in mind that the filter command produces as many output samples as there are input samples,

When convolving two finite-length signals x[n] and h[n], the result y[n] has finite length. The length of y[n] is the length of x[n] plus the number of filter coefficients minus 1. Since the length of y[n] is 6 and the length of x[n] is 5, there are 2 filter coefficients.