The Digital Downconverter

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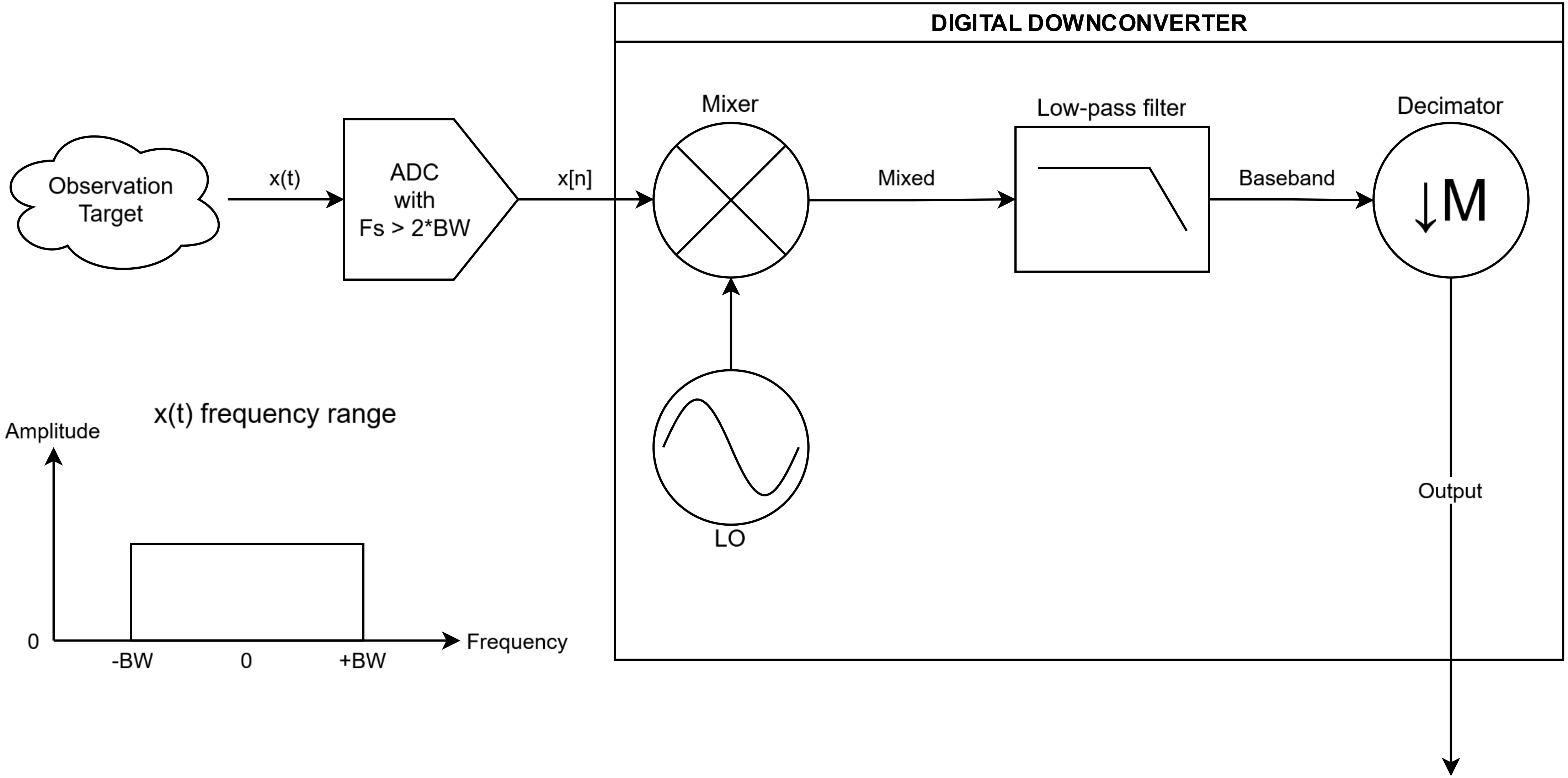
Description

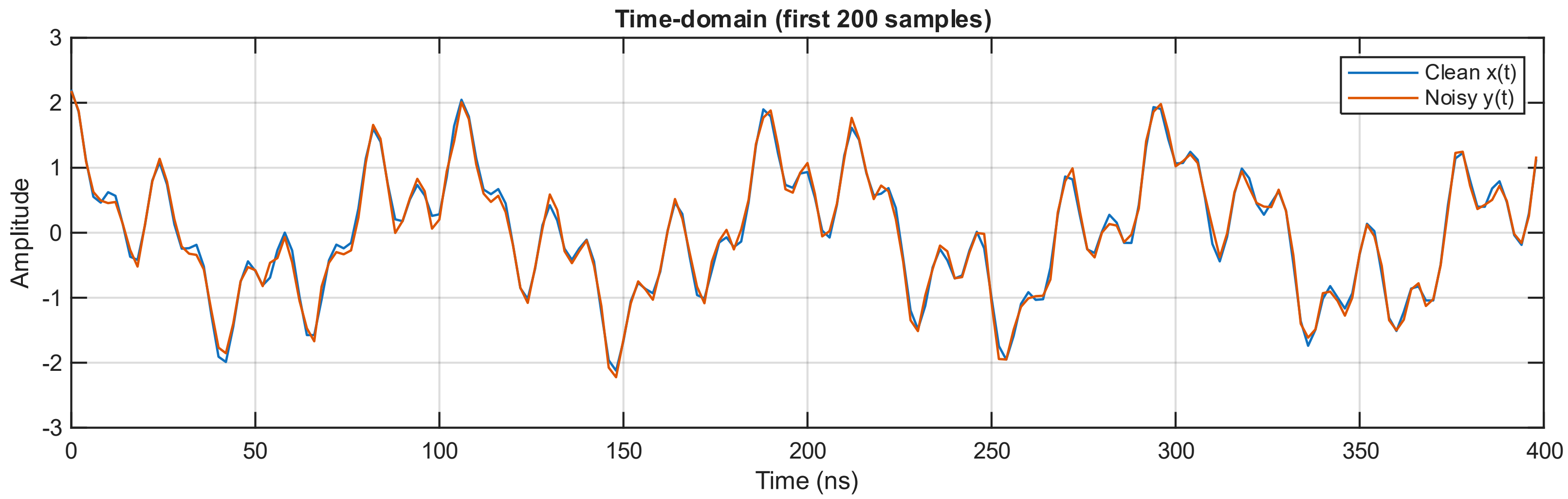
- Used to lower the frequency of a high-frequency signal
- Lossless process
- Components
  - Local oscillator (LO)
  - Mixer
  - Low-pass filter (LPF)
  - Decimator / Downsampler

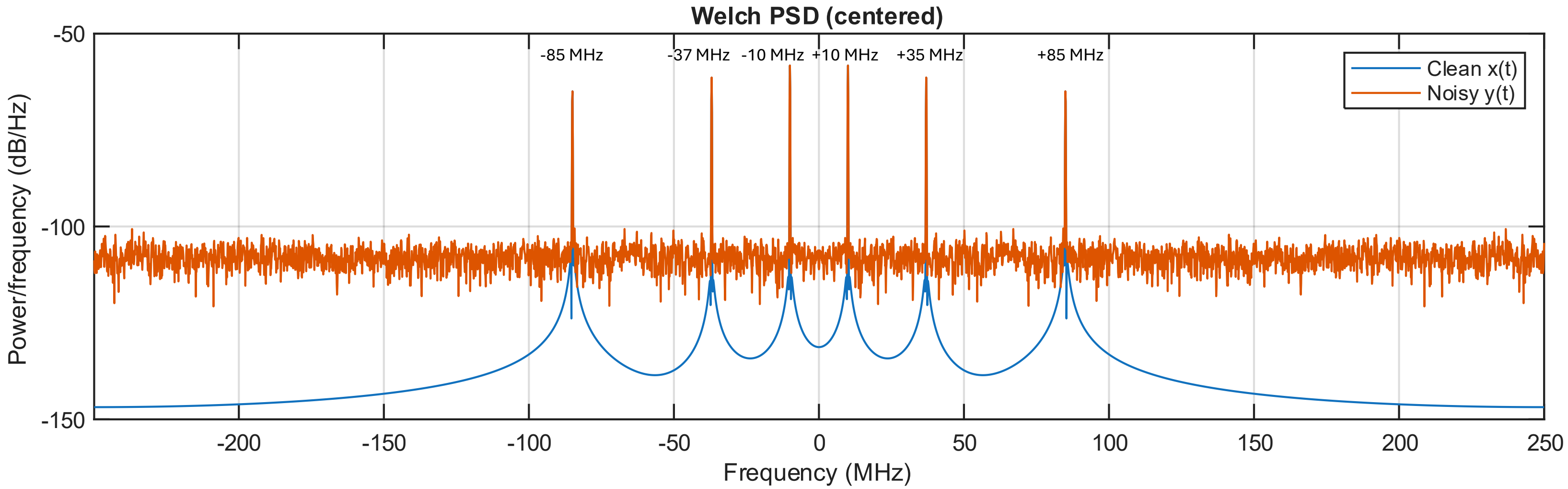
The Digital Downconverter



Incoming Frequencies

- Tones at 10, 37, 85 MHz with additive white gaussian noise on y(t)
- BW = 85 MHz, F_s = 500 MHz
- Real signal: complex symmetry





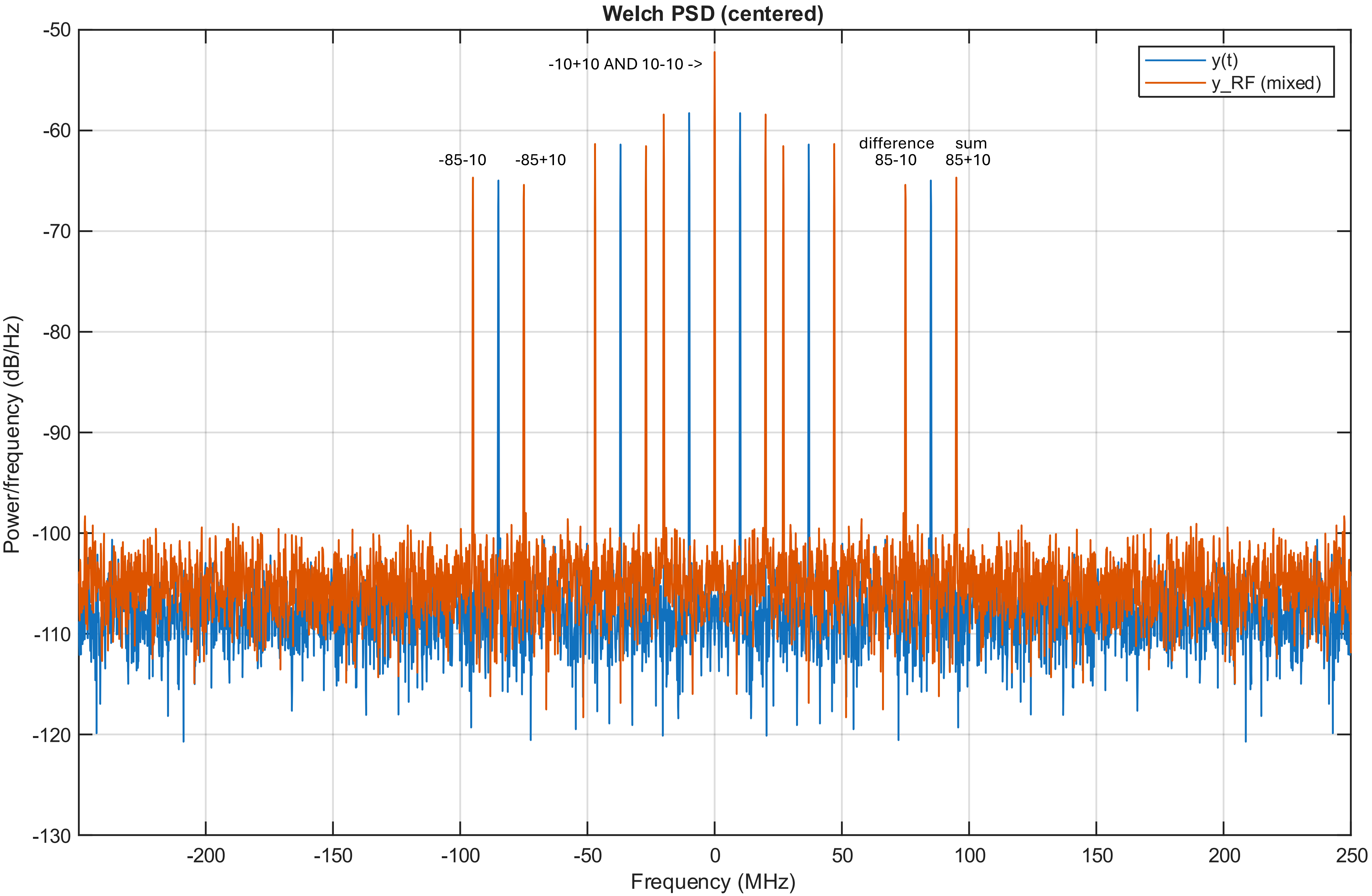
Mixing Stage

- Mixing is multiplication of x[n] and the LO of a constant frequency
    - LO frequency is chosen to be 10 MHz
- Multiplication of two periodic signals follows this trigonometric identity:

$$\cos a \cos b = \frac{\cos(a-b) + \cos(a+b)}{2}$$

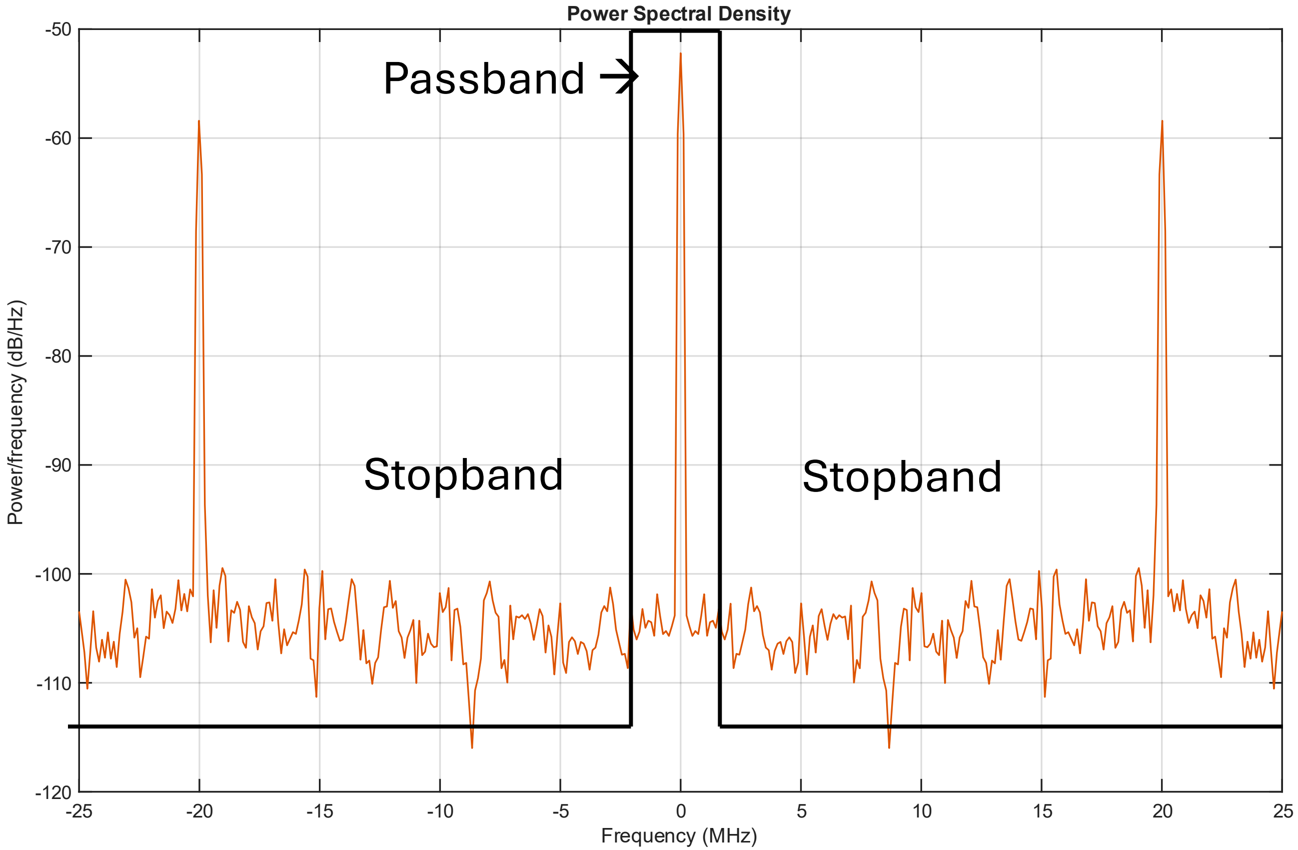
- The goal of downconversion is to frequency shift the signal of interest near 0 Hz.
    - This is why the LO frequency is equal to the frequency of the signal of interest (10 MHz)
- Keep the lower frequency cos(a-b) component (**difference**)
- Discard the higher frequency cos(a+b) component (**sum**)

- Original frequencies are blue peaks
- Mixed signal frequencies are orange peaks
- LO frequency = 10 MHz
- **Orange peaks are +/- 10 MHz apart from blue peaks**
- At DC: the sum image of -10 MHz and the difference image of +10 MHz combine, essentially doubling the amplitude



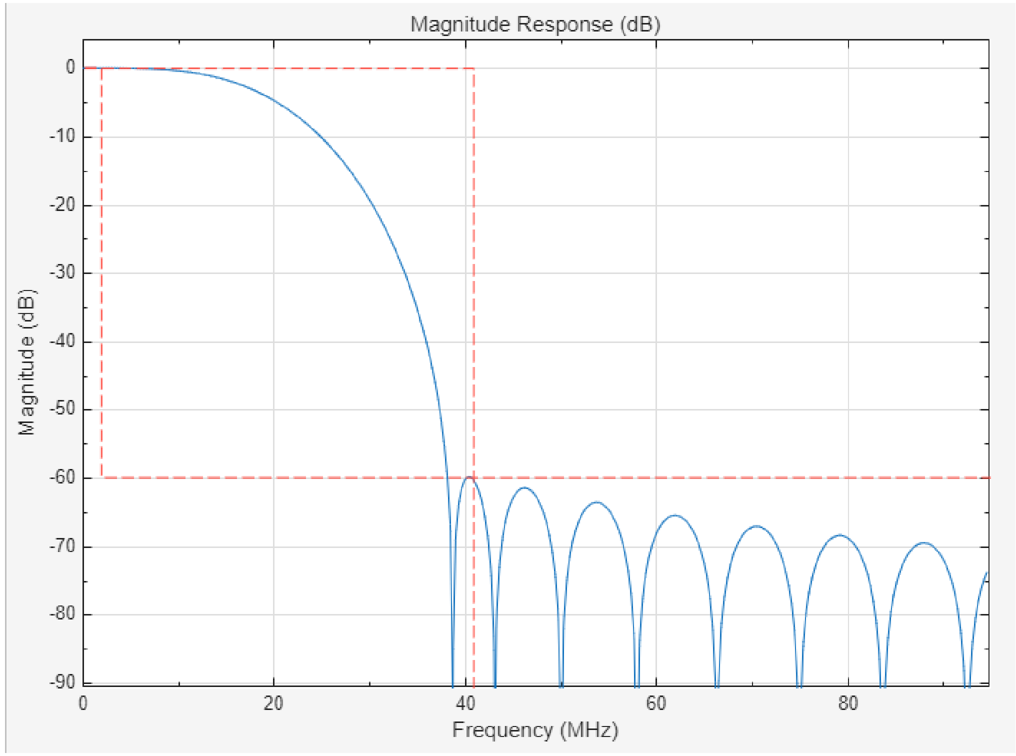
Low-Pass Filtering

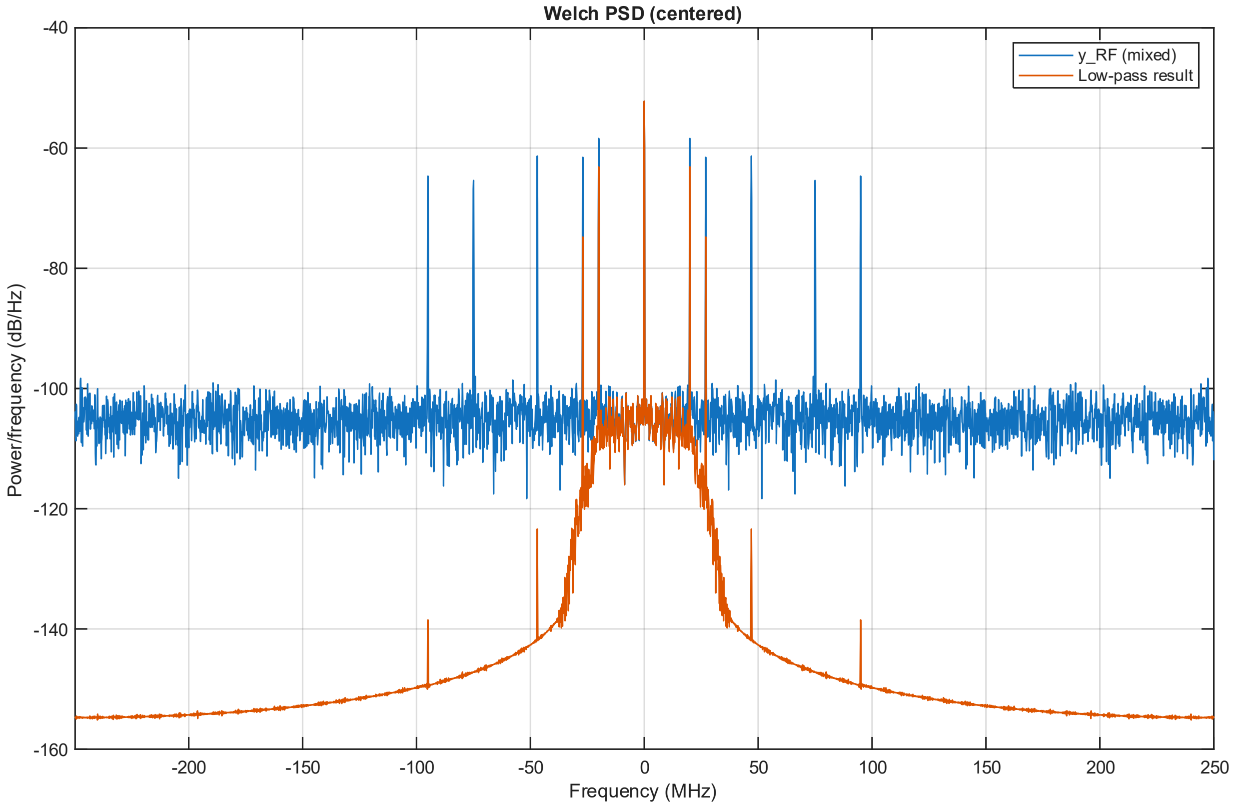
- Mixing shifted the signal of interested to be centered at DC (0 Hz)
- An ideal low pass filter blocks all signals with frequencies greater than the cutoff frequency
- Cutoff frequency of 2 MHz will pass the signal centered at DC and block other signals



Low-Pass Filtering (cont.)

- Non-ideal filter allows +/- 20 and +/- 27 MHZ tones to pass
- -60 dB attenuation at 41 MHz



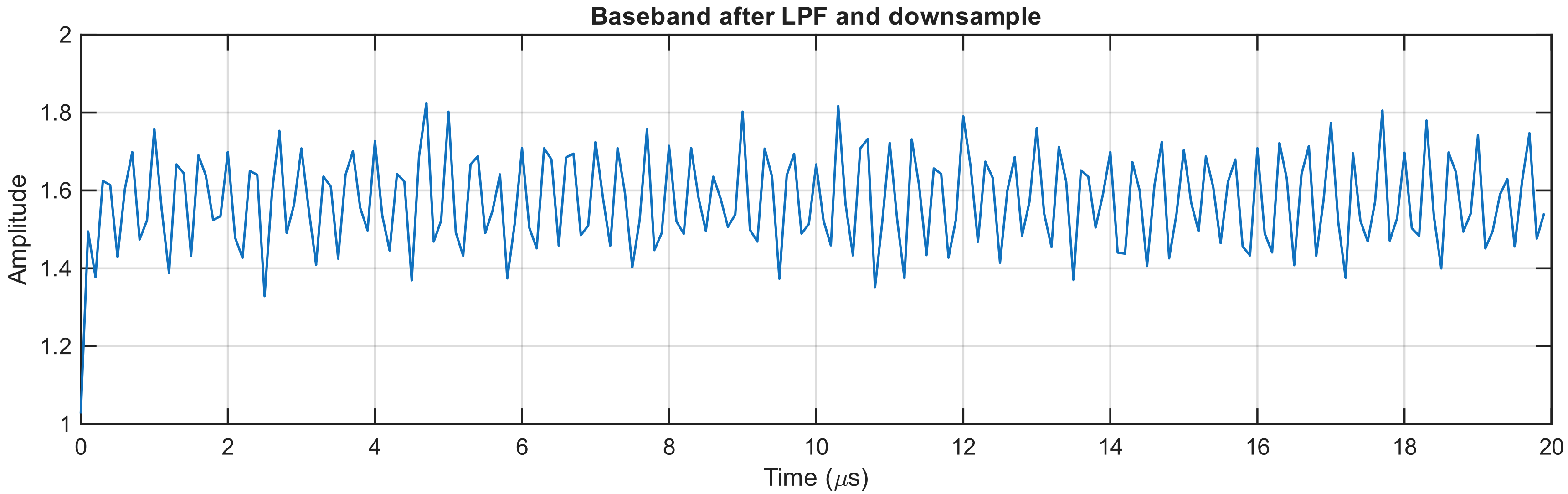


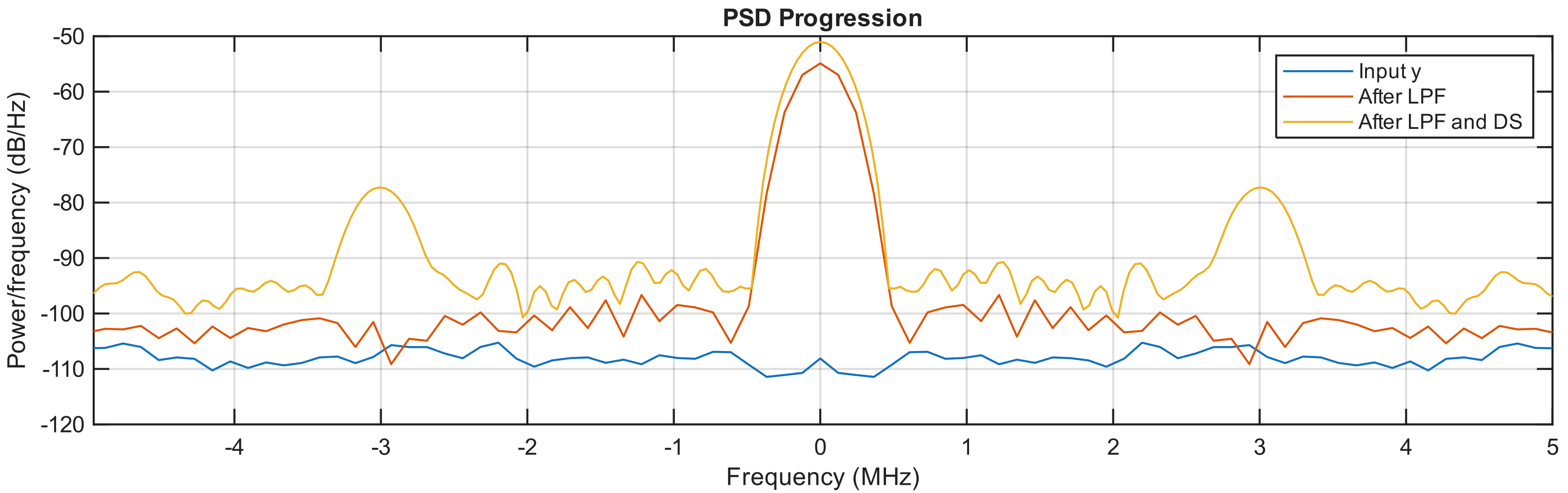
Downsampling

- Goal: resample with a lower sample frequency now that our signal bandwidth is much lower
  - Decimation factor: M = 50, keep every 50th sample
  - $F_{s,new} = F_{s,old}/M = 500\ MHz/50 = 10\ MHz$
- No aliasing will occur if Nyquist-Shannon sampling theorem is followed: $F_s > 2*BW$
  - BW of low-passed signal is ideally 2 MHz, but realistically 27 MHz
- Due to non-ideal low-pass filter, **we expect to see aliasing after downsampling**
  - For no aliasing: $M < F_{s,old}/(2*BW)$ **M < 9.26** for BW of 27 MHz

Downsampling (cont.)

- Time-series plot is lower frequency (x axis unit change)
- Downsampled result shows signals at +/- 3 MHz
    - This is an alias of the +/- 27 MHz signal, shifted by the new sample frequency of 10 MHz
    - 27 MHz - 3*(10 MHz) = -3 MHz
- Downsampled signal is slightly stronger at DC
    - This is an alias of the +/- 20 MHz signal
    - 20 MHz – 2*(10 MHz) = 0 MHz
- **Aliasing of other signals is interfering with signal of interest**





Lessons learned

- Mixing two signals is equivalent to **multiplication**
- Mixing produces two images, the **sum** and **difference**
- The local oscillator frequency can be chosen to move a **signal of interest** to DC
  - The LO frequency equals the signal of interest frequency
  - In this case, a sum and difference image of both conjugates of the signal of interest will **combine at DC** = 0 Hz
- A low-pass filter is used to block all signals except the signal of interest near DC
- Downsampling **reduces the sample frequency** of the signal without aliasing, provided the LPF reduces the bandwidth enough
  - Reducing the sample frequency gets rid of unnecessary samples, making it practical to store and analyze the signal
- The result is 1 narrow-bandwidth signal **chosen** out of a wide range of frequencies