Multirate Signal Processing: Signal Interpolation

How do we change the sample frequency of a signal:

Method 1: Use the sampling theorem (Lecture No. 3)

Define F_s^1 as the original sample frequency, and F_s^2 as the new sample frequency. Recall our interpolation function, where $B = \frac{F_s^1}{2}$:

$$g(t) = \frac{\sin(2\pi Bt)}{2\pi Bt}.$$

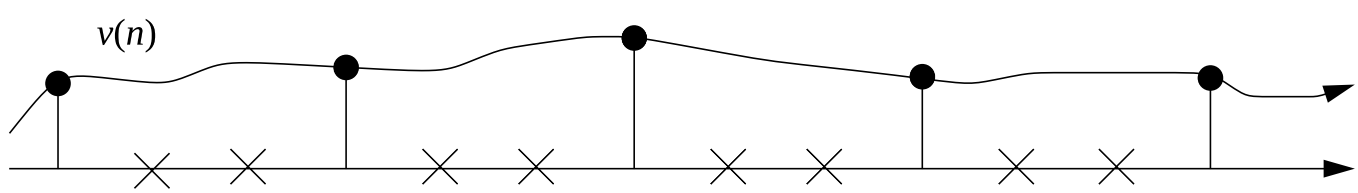
$x(\frac{m}{F_s^2})$ may be expressed as:

$$x(\frac{m}{F_s^2}) = \sum_{n=-\infty}^{\infty} x(\frac{n}{F_s^1}) g(\frac{m}{F_s^2} - \frac{n}{F_s^1}).$$

What are the disadvantages of this method?

Method 2:

Consider the signal $x(n)$. What is the spectrum of $v(n) = x(Ln)$?



Recall the frequency-scaling property:

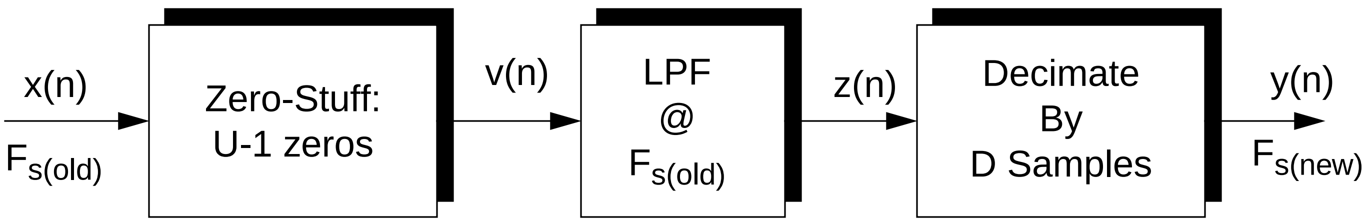
$$V(\omega) = \sum_{m=-\infty}^{\infty} v(m)e^{-j\omega m}$$

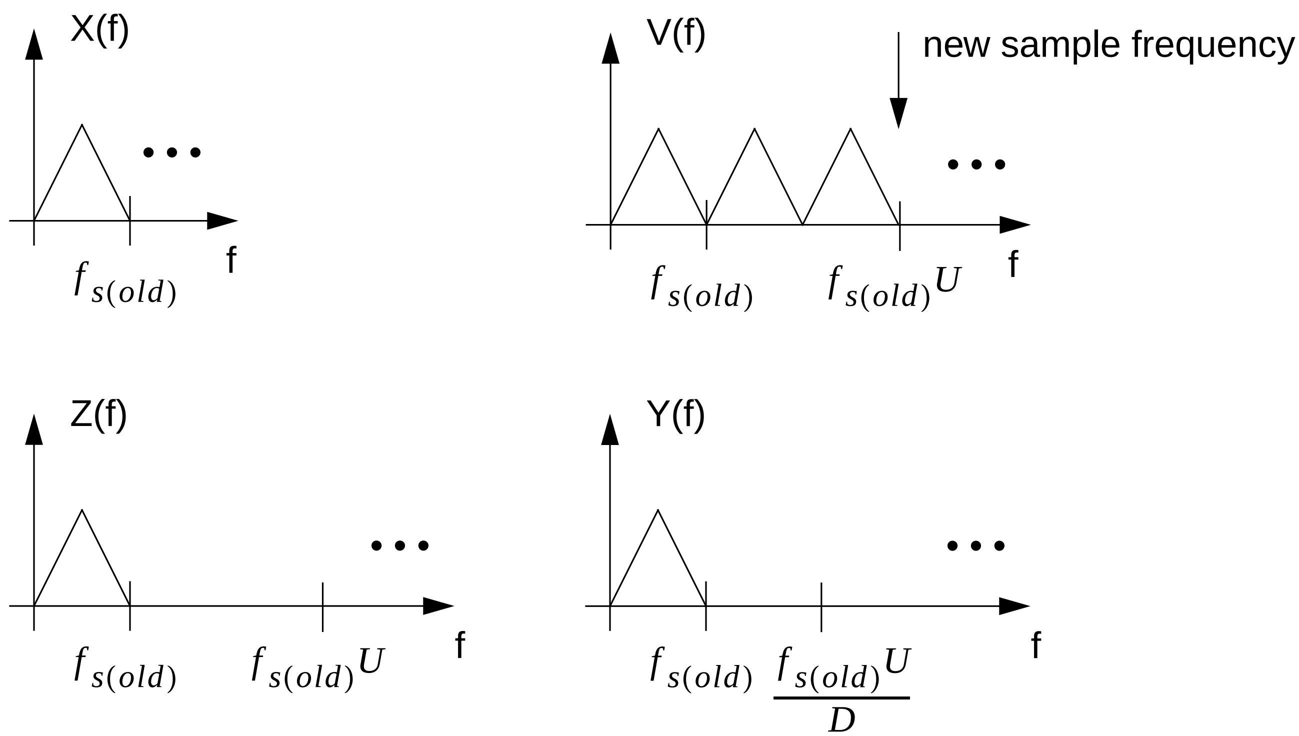
$$= \sum_{n=-\infty}^{\infty} v(n)e^{-j\omega nL}$$

$$= X(\omega L)$$

Signal Interpolation/Decimation By A Ratio Of Integers

$$F_{s(new)} = F_{s(old)}\left(\frac{U}{D}\right)$$





Note that the LPF is run at the decimation rate of D!

Questions:
- Under what conditions will this introduce no distortion?
- How do we implement this efficiently?
- How should we convert from 8 kHz to 6.4 kHz?
- What about the infamous 44.1 kHz CD sample frequency?