

The output signal y[n] is obtained by taking every M^{th} sample of the input signal. If M=3 we should just take every third sample of x[n] to form the desired signal y[m].

Obviously, it only makes sense to reduce the sampling rate if the information constant of the signal we wish to preserve is band limited to $\frac{f_s}{6}$; half the desired sampling rate since the spectral components above this frequency will be aliased into frequencies below $\frac{f_s}{6}$ according to the sampling rule.

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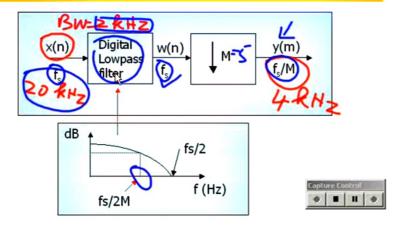
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The diagram below shows a block representations of a times M decimator.

The signal x(n) is first passed through a lowpass filter that attenuates the band from fs/2M to fs/2 to prevent aliasing.



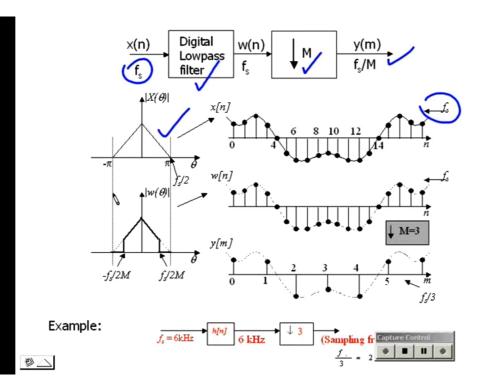
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Should we use IIP or FIR for the lowpass filtering required?

Using an IIR filter in this case has an obvious shortcoming. We cannot take advantage of the fact that we only have to compute every Nth output, since previous outputs are required to compute the Mth output. Thus no saving is realised.

On the other hand, using an FIR filter, in this case implies that we can do our computations at the rate of fs/M. Thus, using an FIR filter in the decimation process will lead to a significantly lower computation rate.

Another advantage of using an FIR filter is the fact that we can easily design linear phase filters and septime control is desirable in many applications.

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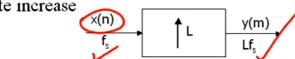
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Interpolation

■ The process of interpolation involves a sampling rate increase

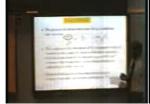


- The sequence x(n) was derived by sampling x(t) at a sampling rate f_s and we want to obtain a sequence y(n) that approximates as closely as possible the sequence that would have been obtained had we sampled x(t) at the rate Lf_s.
- Interpolation involves inserting between any samples x(n) and x(n-1) and additional L-1 samples.



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Interpolation Examples

$$x[n]=\{1, 2, 4, 3, -5, 6, -7, 2, 4, 3\}$$

$$\uparrow 2$$
 $y[m]=\{1,0,2,0,4,0,3,0,-5,0,6,0,-7,0,2,0,4,0,3,0\}$

$$x[n]=\{1, 2, 4, 3, 5, 6, -7, 2, 4, 3\}$$

 $y[m] = \{1,0,0,2,0,0,4,0,0,3,0,0,-5,0,0,6,0,0,-7,0,0,2,0,0,4,0,0,3,0,0\}$

Capture Control

■ II ●

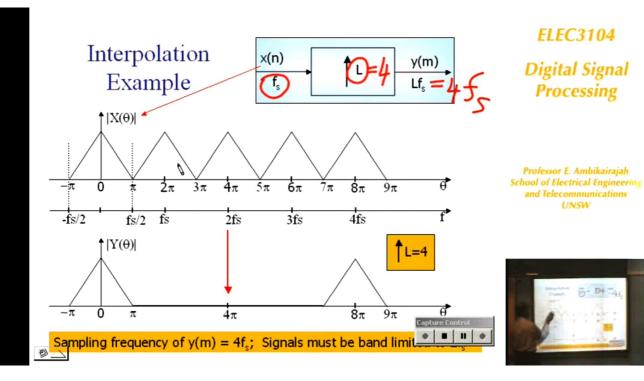
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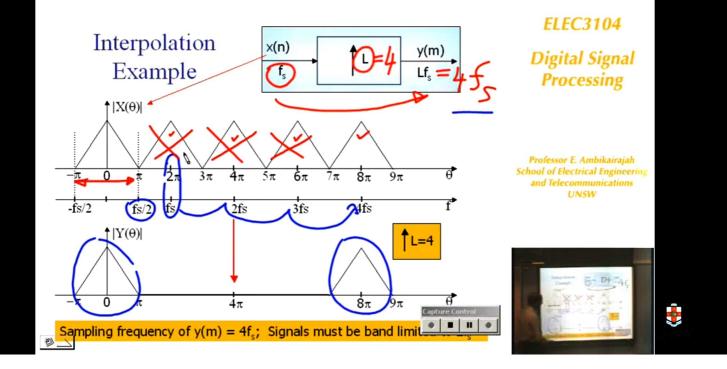
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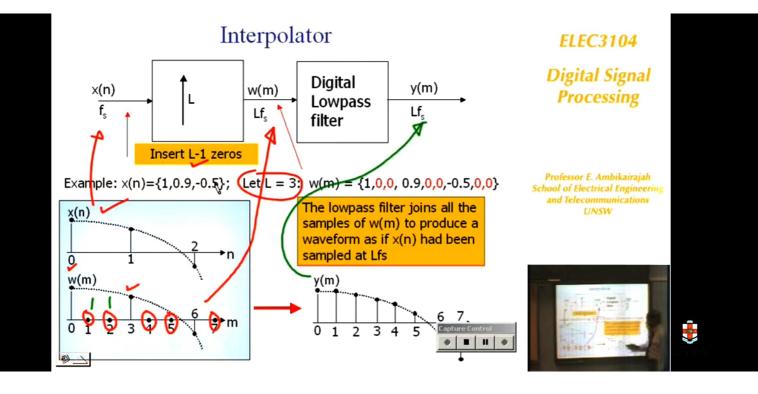


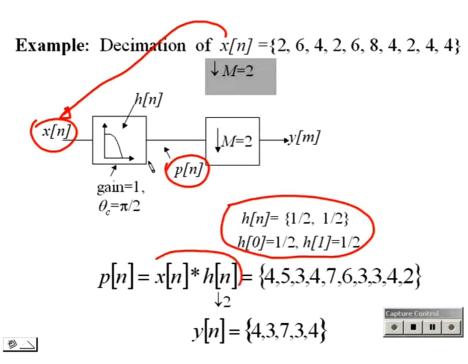








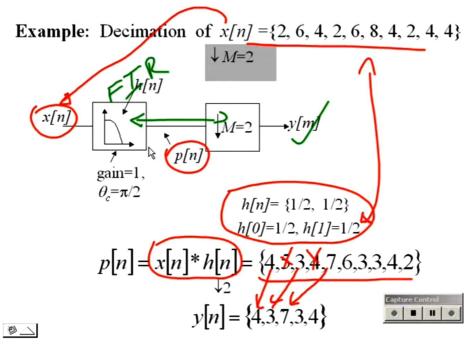




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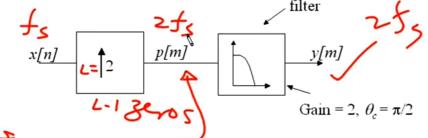


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Example: Linear interpolation of $x[n] = \{1,3,5,3,7\}$



 $x[n] = \{1,3,5,3,7\}$ $p[n] = \{1,0,3,0,5,0,3,0,7,0\} \text{ (insert zeros)}$ $h[n] = \left\{\frac{1}{2},1,\frac{1}{2}\right\}$

$$y[n] = p[n] * h[n] = \{1,2,3,4,5,4,3,5,7,3.5\}$$



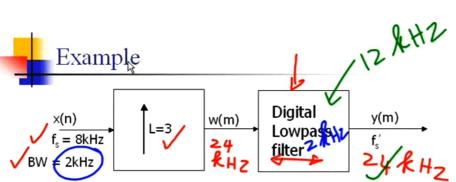
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• What should be the cut-off frequency of the digital lowpass filter?

• What should be the value of f_s'?



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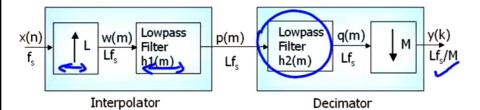




Figure below shows that the sampling frequency change is achieved by first interpolating the data by L and then decimating by M.

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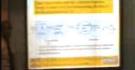
The two Digital Lowpas Filters, h1(m) and h2(m) can be combined into a single filter since they are in cascade and have a common sampling frequency





the resulting operation is a decimation process by a Control

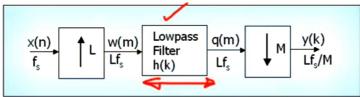




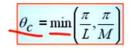




Summary: Sampling Rate Conversion by Non-Integer Factors



The lowpass filter that we require is the one that has a cut-off frequency:





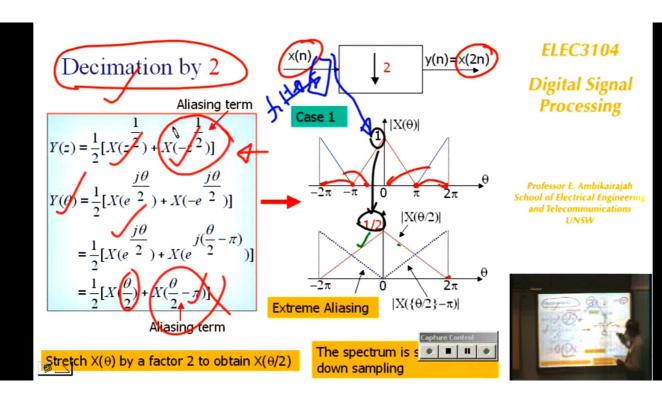


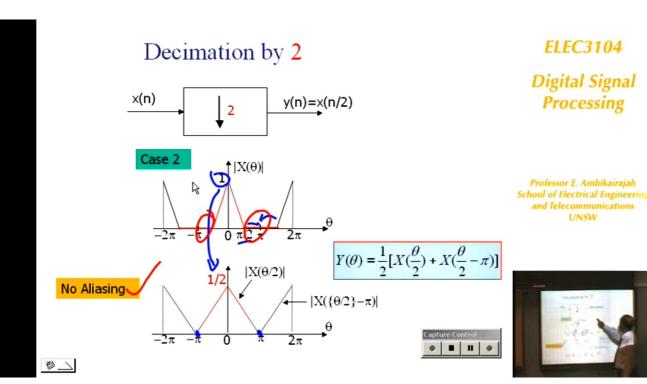
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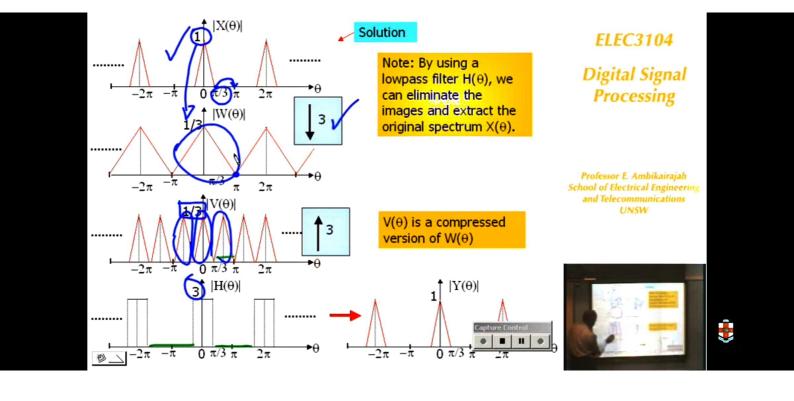


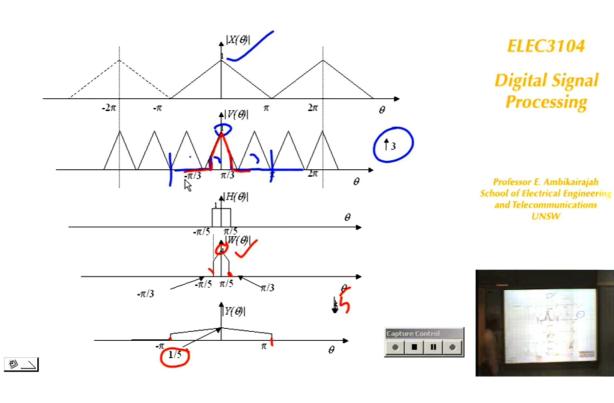










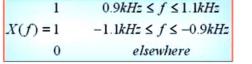


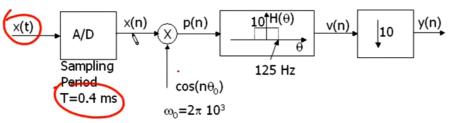
Example: x(t) is the input signal for the system shown below. The analogue signal x(t) has the spectrum X(f) given by:

$$1 0.9kHz \le f \le 1.1kHz$$

$$X(f) = 1 -1.1kHz \le f \le -0.9kHz$$

$$0 elsewhere$$





 $H(\theta)$ is an ideal lowpass filter (gain=10) with cut-off frequency f_c = 125 Hz. Sketch, one above another, $|X(\theta)|,\,|P(\theta)|,|V(\theta)|,\,|Y(\theta)|$ against θ .

Note: $P(\theta) = \frac{1}{2} [X(\theta + \theta_0) + X(\theta - \theta_0)]$

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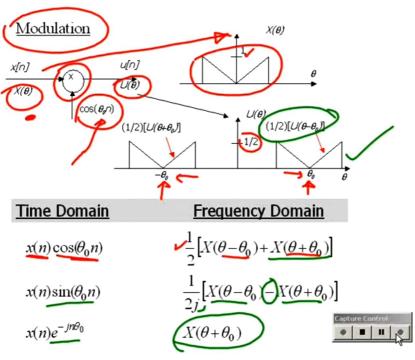
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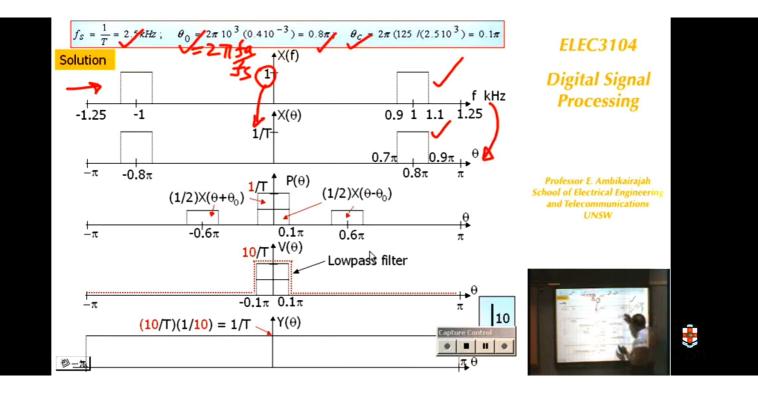


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Example:

x(n) is the input signal for the system shown below.

If $x(n) = 0.5 \delta(n+1) + \delta(n) + 0.5\delta(n-1)$, Show that

$$S(\theta) = 1/2$$

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$$0$$

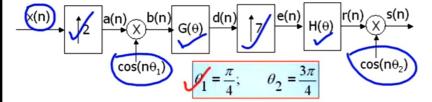
$$10\pi/4 \le \theta \le 11\pi/4$$

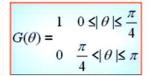
$$-11\pi/4 \le \theta \le -10\pi/4$$

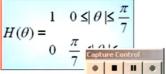
$$elsewhere$$

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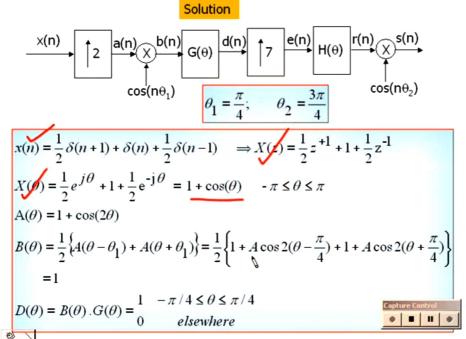












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