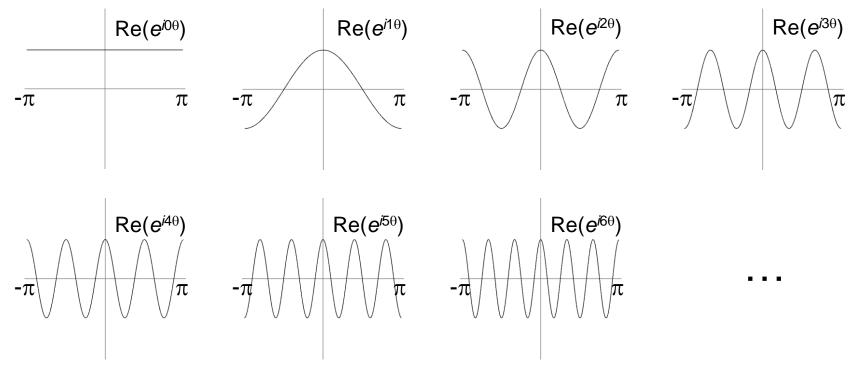
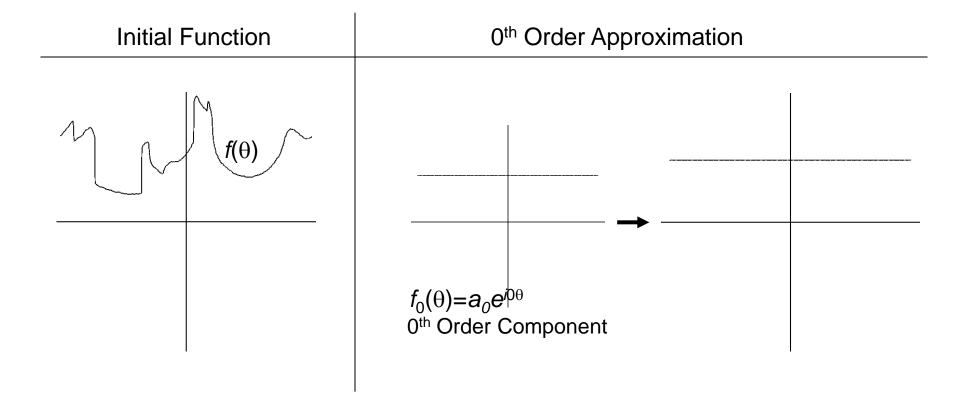
# Physically Based Rendering (600.657)

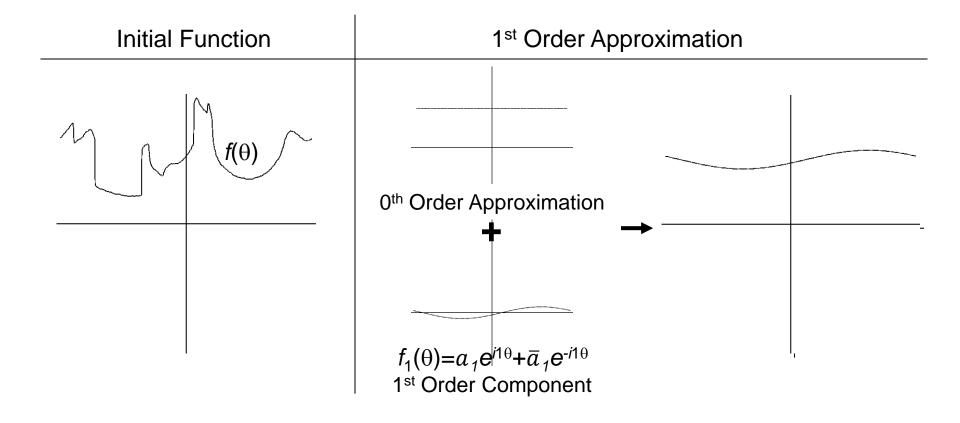
Sampling

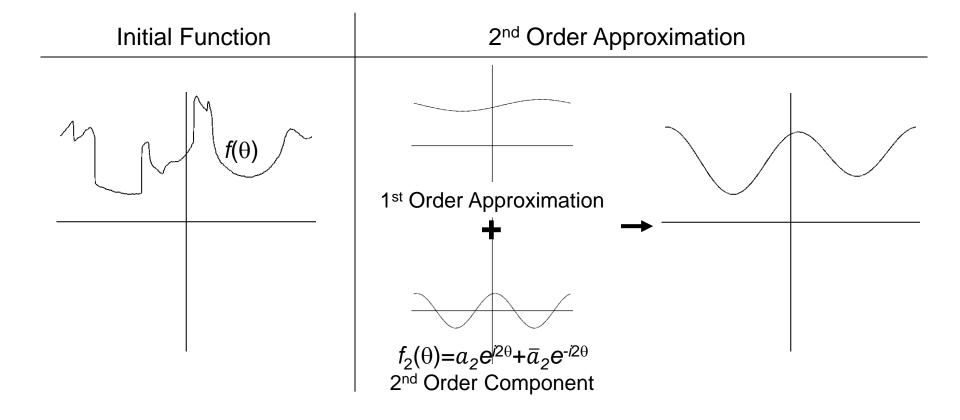
 Fourier analysis provides a way for expressing (or approximating) any signal as a sum of complex exponentials.

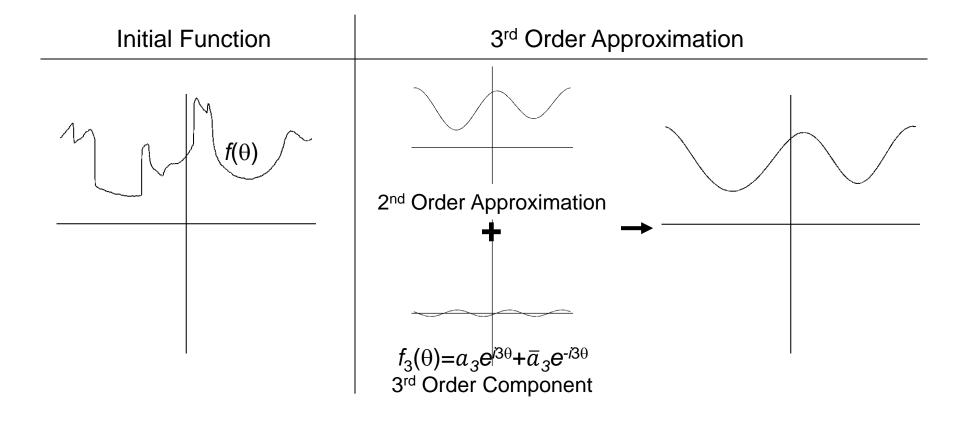


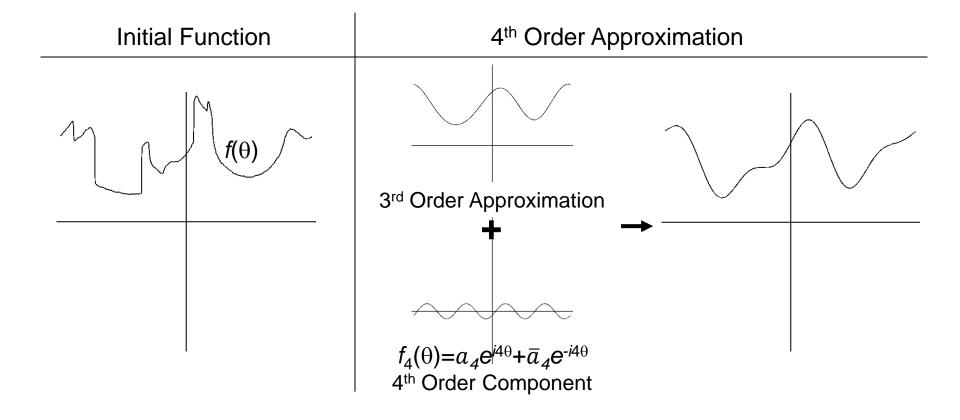
The Building Blocks for the Fourier Decomposition

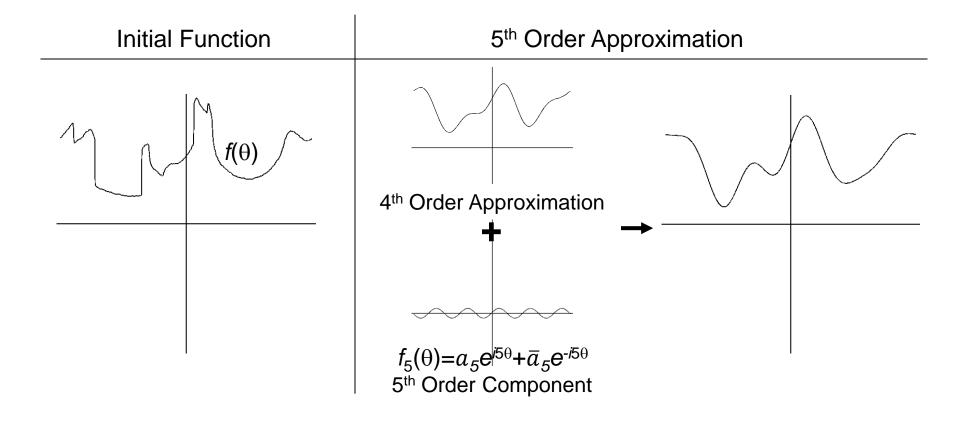


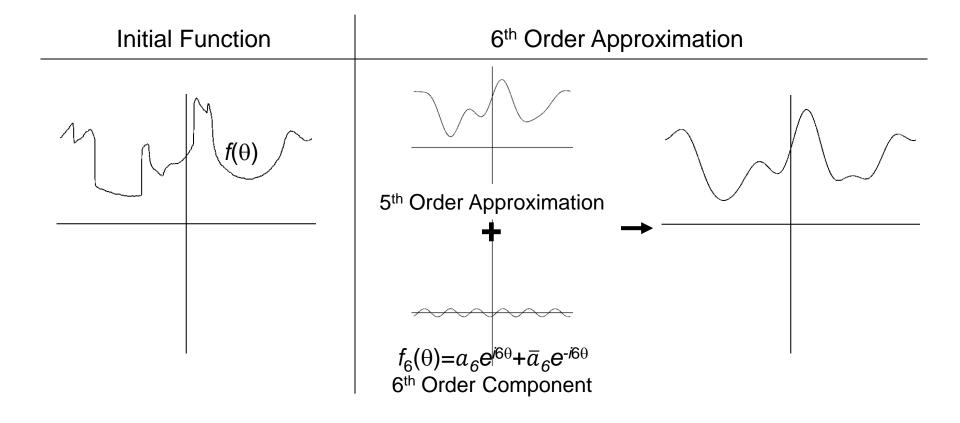


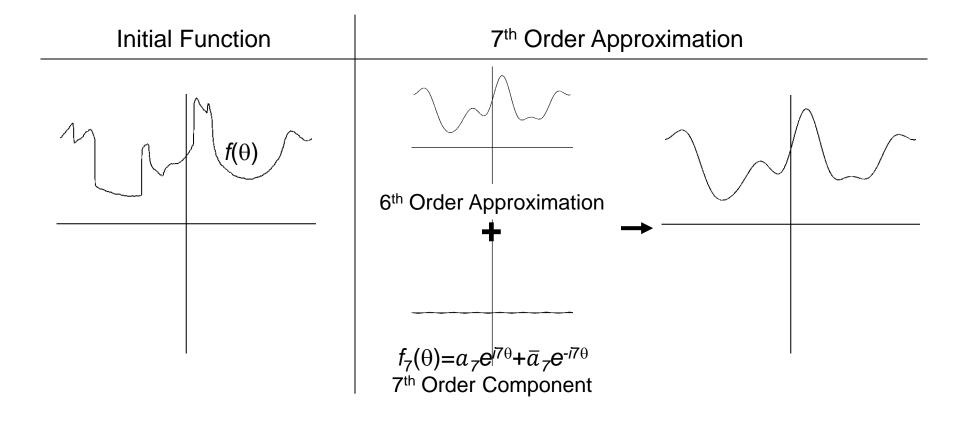


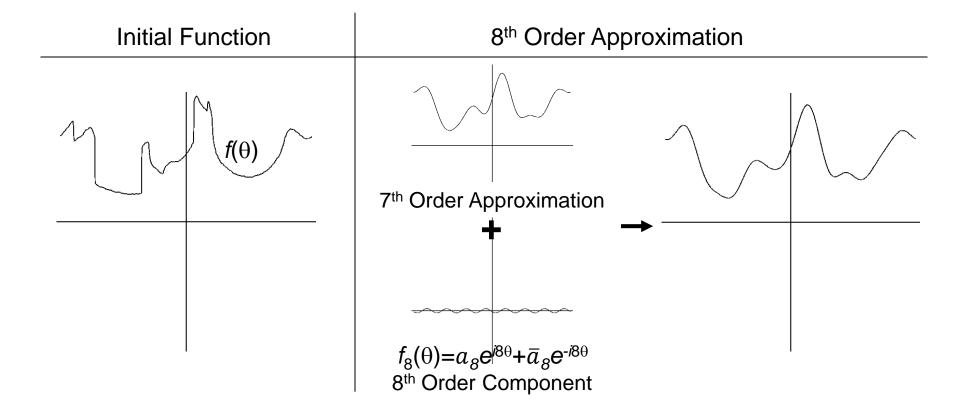


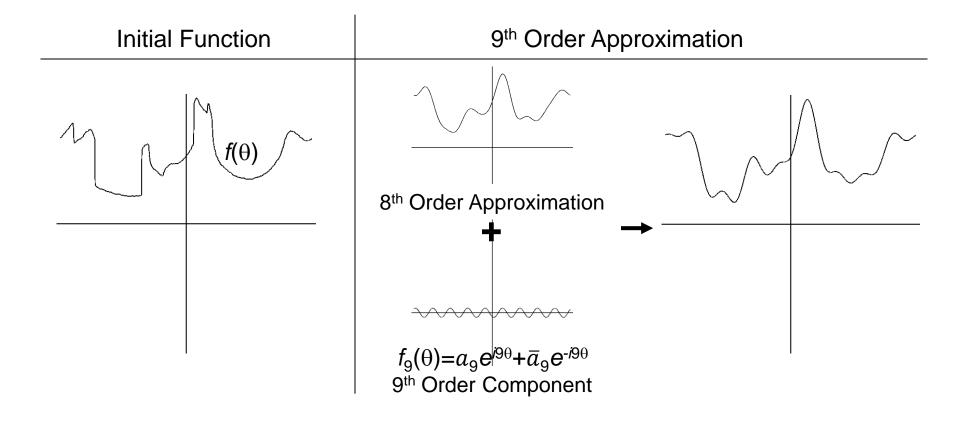


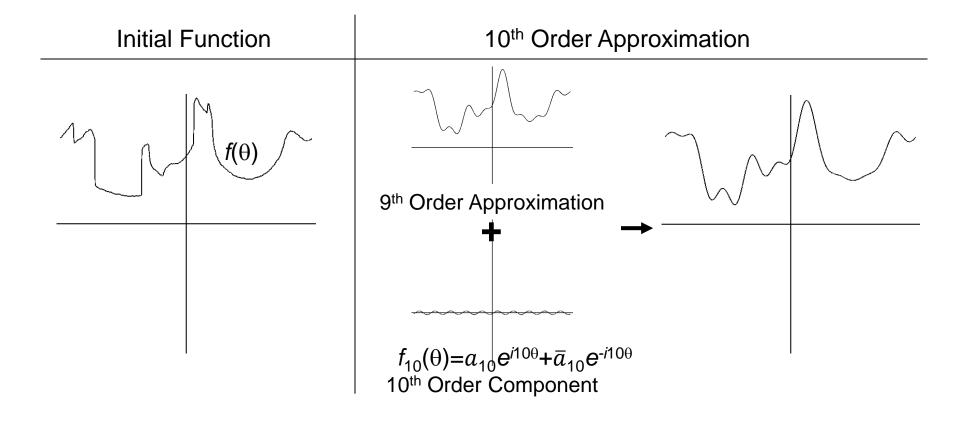


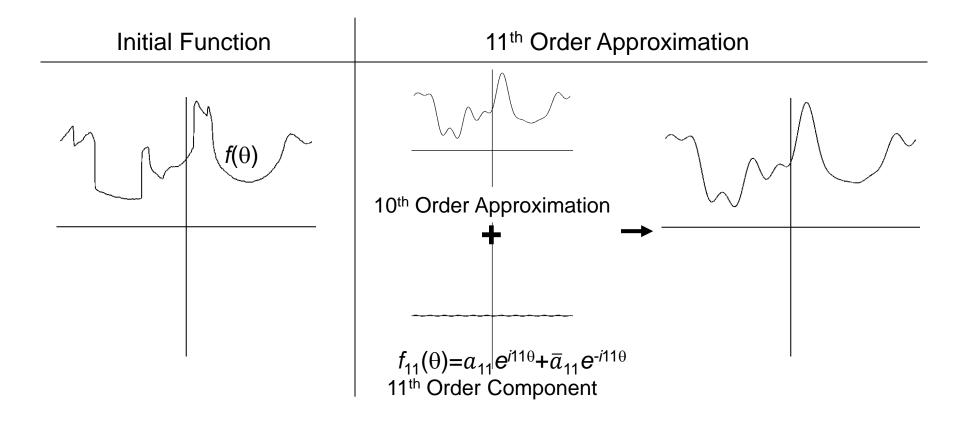


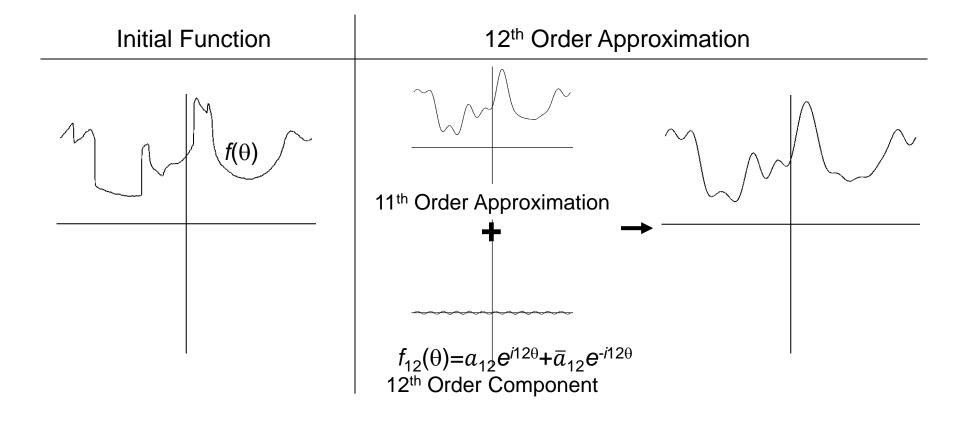


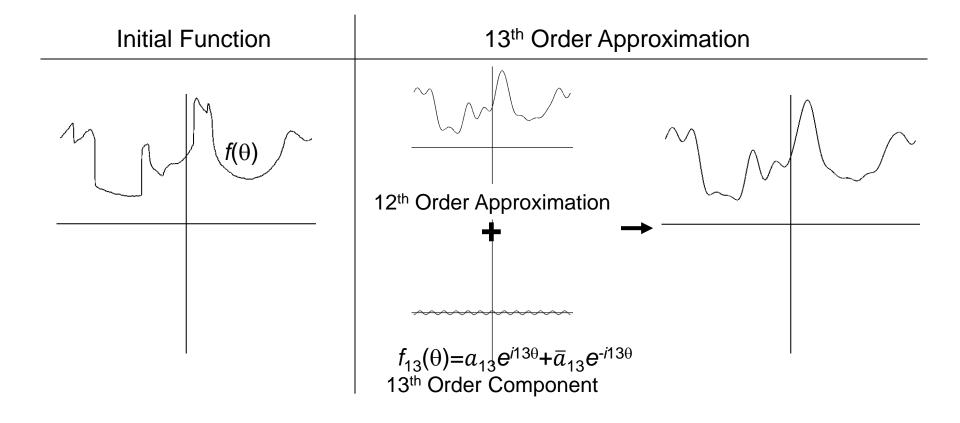


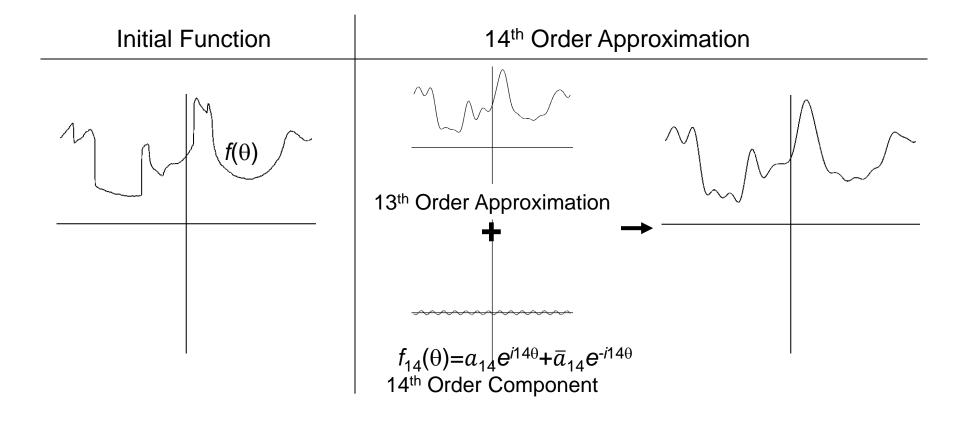


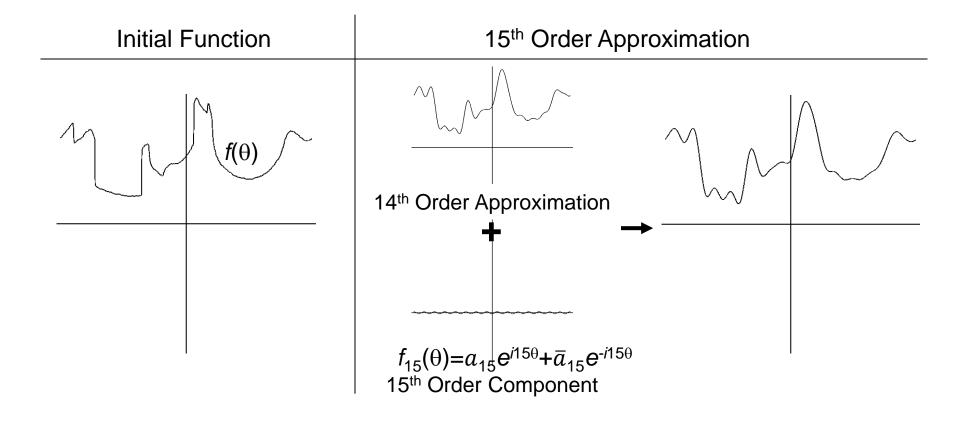


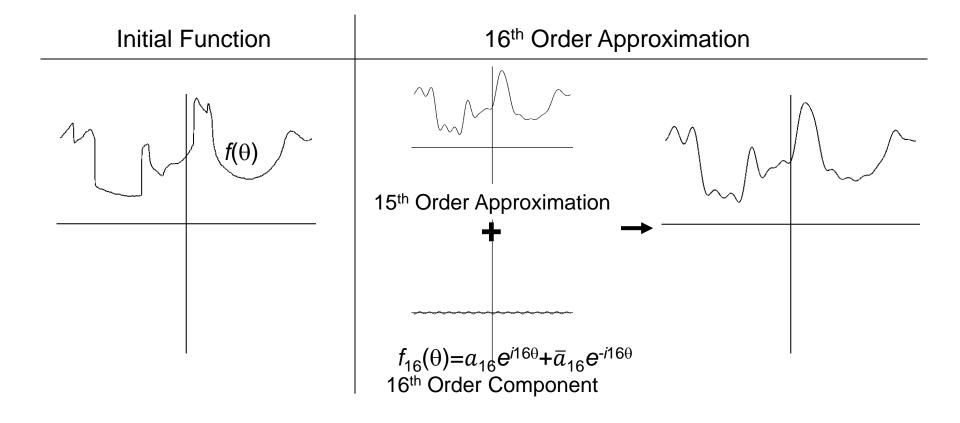






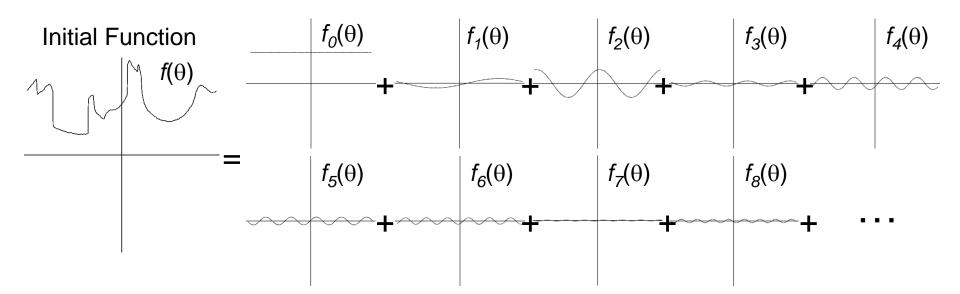






 Combining all of the frequency components together, we get the initial function.

$$f(\theta) = \sum_{k=-\infty}^{\infty} f_k(\theta) = \sum_{k=-\infty}^{\infty} a_k \frac{e^{ik\theta}}{\sqrt{2\pi}}$$



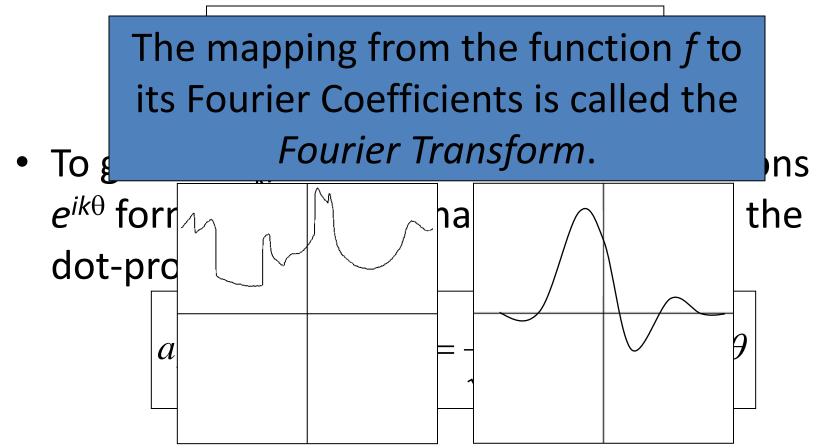
 Combining all of the frequency components together, we get the initial function.

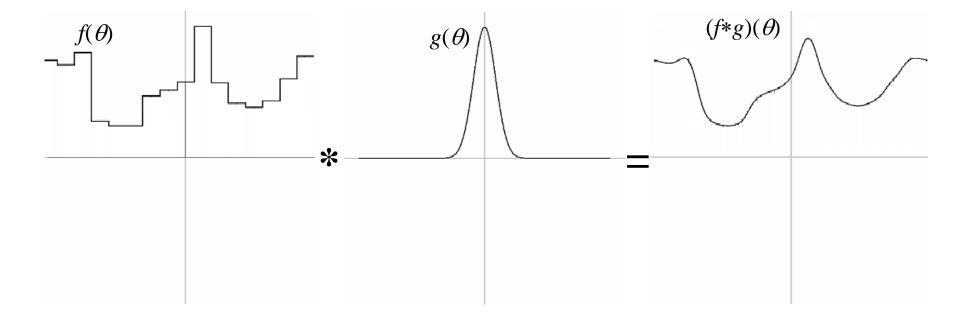
$$f(\theta) = \sum_{k=-\infty}^{\infty} f_k(\theta) = \sum_{k=-\infty}^{\infty} a_k \frac{e^{ik\theta}}{\sqrt{2\pi}}$$

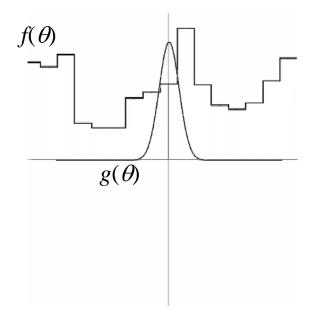
• To get the  $a_k$ , use the fact that the functions  $e^{ik\theta}$  form an orthonormal basis and take the dot-product:

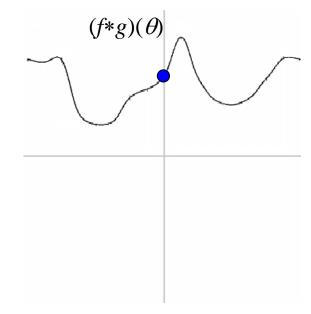
$$\left| a_k = \left\langle f(\theta), \frac{e^{ik\theta}}{\sqrt{2\pi}} \right\rangle = \frac{1}{\sqrt{2\pi}} \int_{-\pi}^{\pi} f(\theta) e^{-ik\theta} d\theta \right|$$

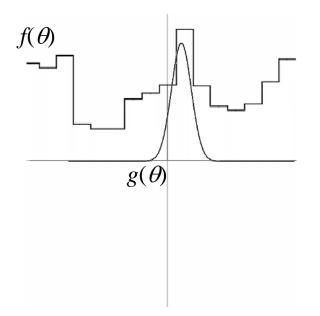
 Combining all of the frequency components together, we get the initial function.

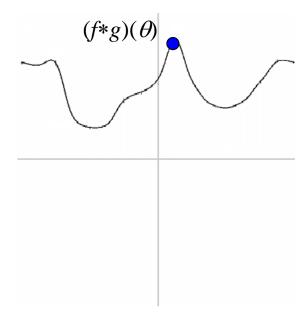


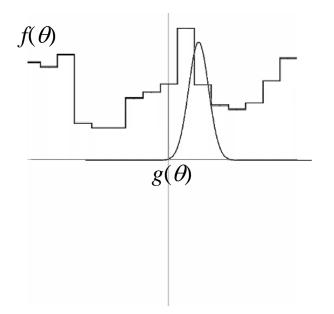


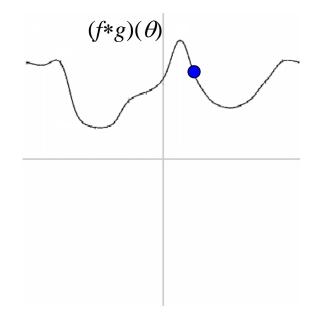


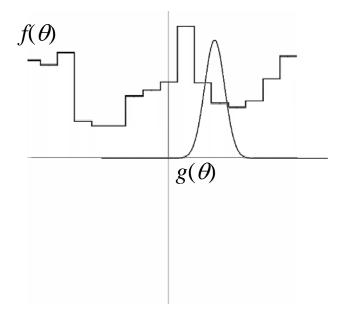


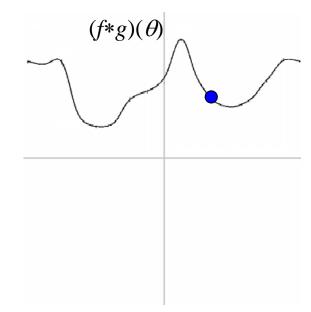


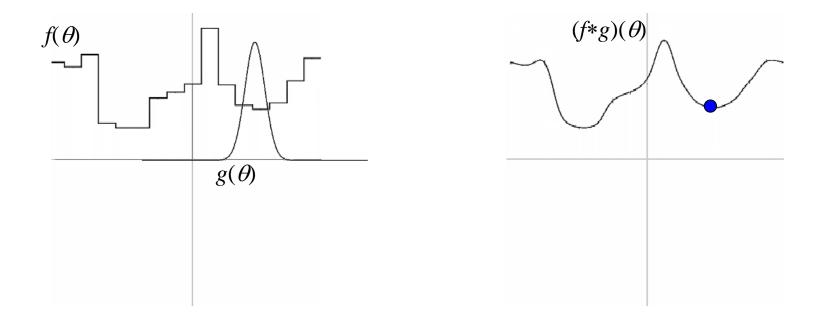


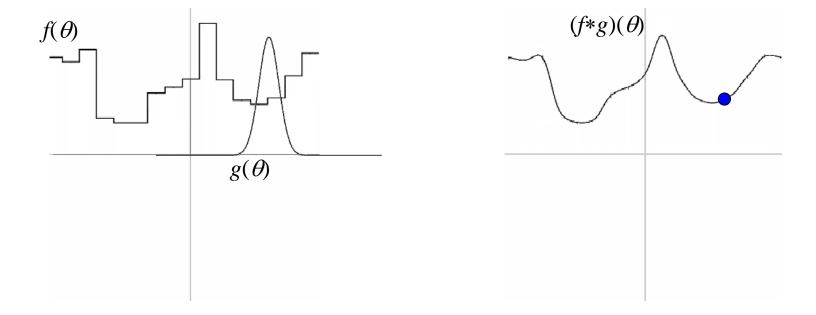


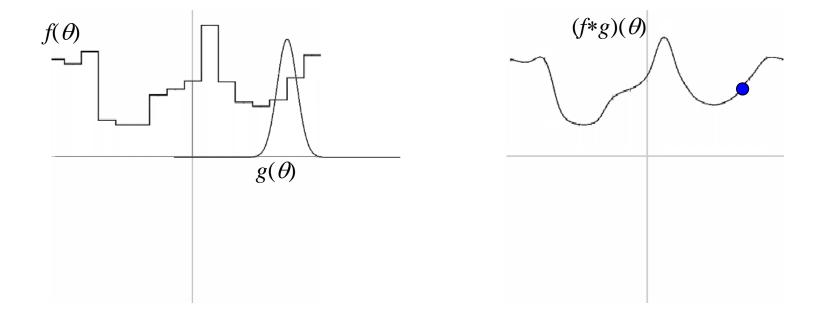




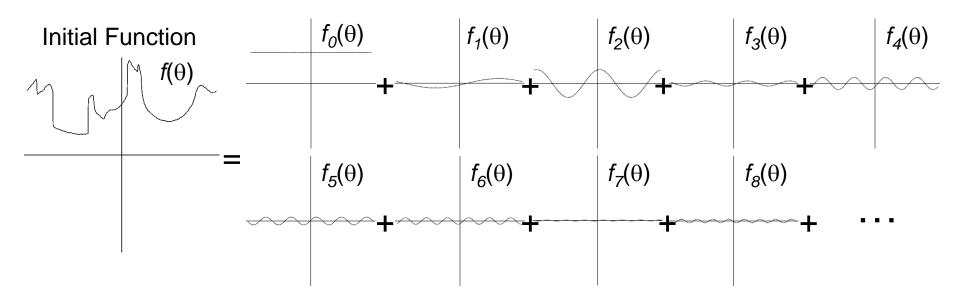








Q: What so special about the complex exponentials?



A: Translating a complex exponential is the same as multiplying it:

$$e^{ik(\theta-\theta_0)}=e^{-ik\theta_0}\cdot e^{ik\theta}$$

A: Translating a complex exponential is the same as multiplying it:

$$e^{ik(\theta-\theta_0)} = e^{-ik\theta_0} \cdot e^{ik\theta}$$

⇒Convolution in the spatial domain is multiplication in the frequency domain.

$$f(\theta) = \sum_{k=-\infty}^{\infty} a_k \frac{e^{ik\theta}}{\sqrt{2\pi}} \quad \text{and} \quad g(\theta) = \sum_{k=-\infty}^{\infty} b_k \frac{e^{ik\theta}}{\sqrt{2\pi}}$$
$$(f * g)(\theta) = \sum_{k=-\infty}^{\infty} a_k b_k \frac{e^{ik\theta}}{\sqrt{2\pi}}$$

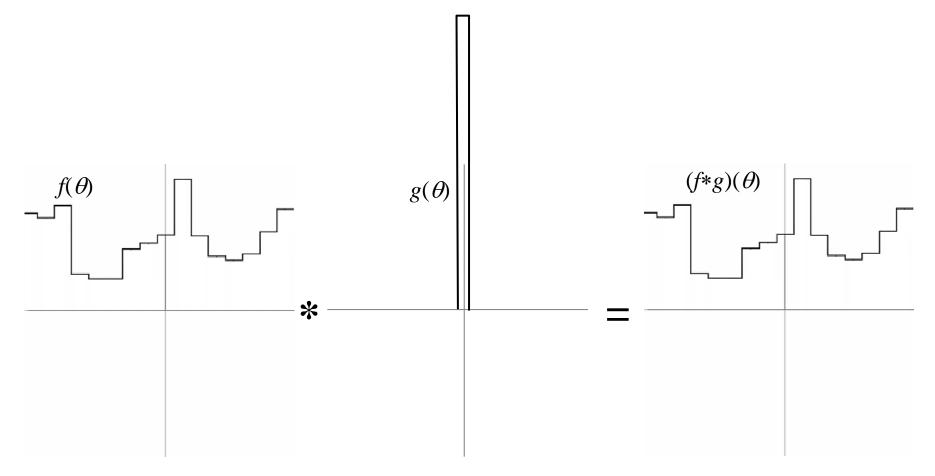
A: Translating a complex exponential is the same as multiplying it:

$$e^{ik(\theta-\theta_0)} = e^{-ik\theta_0} \cdot e^{ik\theta}$$

- ⇒Convolution in the spatial domain is multiplication in the frequency domain.
- ⇒Because the Fourier Transform is (almost) its own inverse, multiplication in the spatial domain is convolution in the frequency.

$$(f \cdot g)(\theta) = \sum_{k=-\infty}^{\infty} (a * b)_k \frac{e^{ik\theta}}{\sqrt{2\pi}}$$

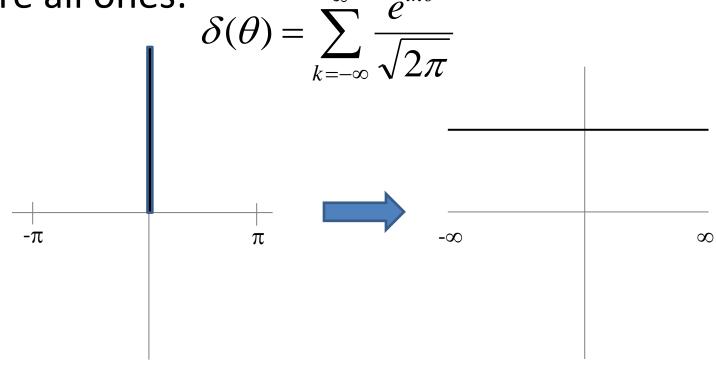
If g is a delta function (infinitely narrow, unit area), convolving with g preserves the signal.



If g is a delta function (infinitely narrow, unit area), convolving with g preserves the signal.

⇒The Fourier coefficients of the delta function

are all ones:



#### Convolution

If g is a delta function (infinitely narrow, unit area), convolving with g preserves the signal.

⇒The Fourier coefficients of the delta function are all ones:  $\delta(\theta) = \sum_{k=-\infty}^{\infty} \frac{e^{ik\theta}}{\sqrt{2\pi}}$ 

$$\delta(\theta) = \sum_{k=-\infty}^{\infty} \frac{e^{ik\theta}}{\sqrt{2\pi}}$$

⇒ The Fourier coefficients of a translated delta function are:

Signal series:
$$\delta(\theta - \theta_0) = \sum_{k = -\infty}^{\infty} e^{-ik\theta_0} \frac{e^{ik\theta}}{\sqrt{2\pi}}$$

⇒The Fourier coefficients of the sum of two evenly spaced delta function are:

$$\delta(\theta + \pi) + \delta(\theta) = \sum_{k = -\infty}^{\infty} \frac{e^{ik\theta}}{\sqrt{2\pi}} + \sum_{k = -\infty}^{\infty} e^{ik\pi} \frac{e^{ik\theta}}{\sqrt{2\pi}}$$

⇒The Fourier coefficients of the sum of two evenly spaced delta function are:

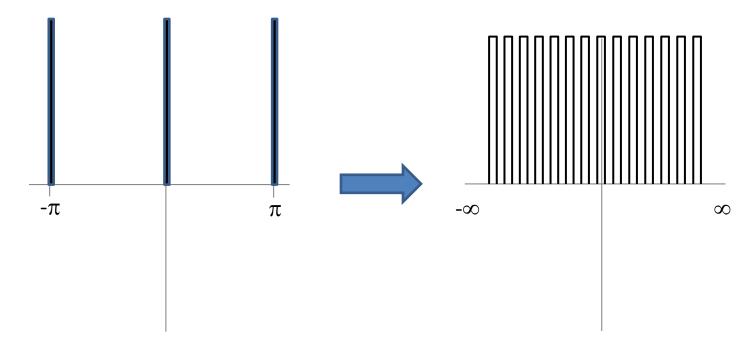
$$\delta(\theta + \pi) + \delta(\theta) = \sum_{k = -\infty}^{\infty} \frac{e^{ik\theta}}{\sqrt{2\pi}} + \sum_{k = -\infty}^{\infty} e^{ik\pi} \frac{e^{ik\theta}}{\sqrt{2\pi}}$$
$$= \sum_{k = -\infty}^{\infty} \left(1 + e^{ik\pi}\right) \frac{e^{ik\theta}}{\sqrt{2\pi}}$$

⇒The Fourier coefficients of the sum of two evenly spaced delta function are:

$$\delta(\theta + \pi) + \delta(\theta) = \sum_{k = -\infty}^{\infty} \frac{e^{ik\theta}}{\sqrt{2\pi}} + \sum_{k = -\infty}^{\infty} e^{ik\pi} \frac{e^{ik\theta}}{\sqrt{2\pi}}$$
$$= \sum_{k = -\infty}^{\infty} \left(1 + e^{ik\pi}\right) \frac{e^{ik\theta}}{\sqrt{2\pi}}$$
$$= \sum_{k = -\infty}^{\infty} \begin{cases} 2 & \text{if } k \text{ is even} \\ 0 & \text{if } k \text{ is odd} \end{cases} \frac{e^{ik\theta}}{\sqrt{2\pi}}$$

⇒The Fourier coefficients of the sum of two evenly spaced delta function are:

$$\delta(\theta + \pi) + \delta(\theta) = \sum_{k = -\infty}^{\infty} \begin{cases} 2 & \text{if } k \text{ is even} \\ 0 & \text{if } k \text{ is even} \end{cases} \frac{e^{ik\theta}}{\sqrt{2\pi}}$$



⇒The Fourier coefficients of the sum of N evenly spaced delta function are:

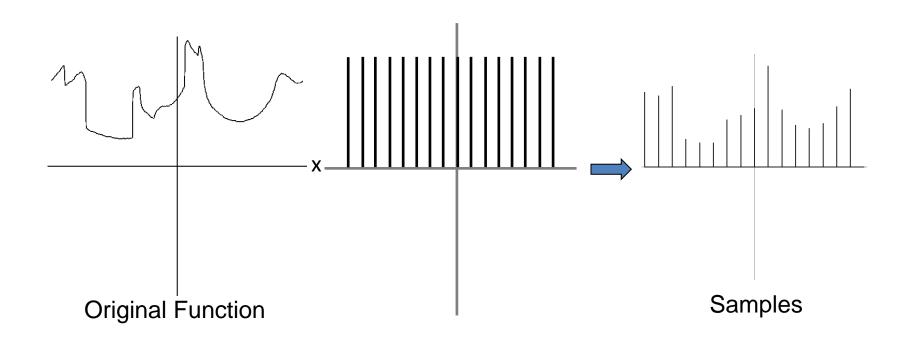
$$\sum_{j=0}^{N-1} \delta \left( \theta + \pi - \frac{2\pi j}{N} \right) = \sum_{k=-\infty}^{\infty} e^{ik\pi} \left( \sum_{j=0}^{N-1} e^{-ik\frac{2\pi j}{N}} \right) \frac{e^{ik\theta}}{\sqrt{2\pi}}$$

$$= (-1)^N \sum_{k=-\infty}^{\infty} \begin{cases} N & \text{if } N \text{ divides } k \\ 0 & \text{otherwise} \end{cases} \frac{e^{ik\theta}}{\sqrt{2\pi}}$$

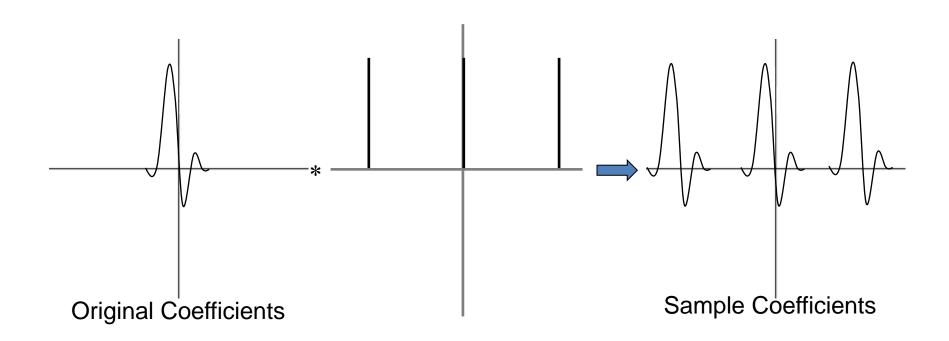
⇒The Fourier coefficients of the sum of N evenly spaced delta function are:

$$\sum_{j=0}^{N-1} \delta\left(\theta + \pi - \frac{2\pi j}{N}\right) = (-1)^N \sum_{k=-\infty}^{\infty} \begin{cases} N & \text{if } N \text{ divides } k \\ 0 & \text{otherwise} \end{cases} \frac{e^{ik\theta}}{\sqrt{2\pi}}$$

We can express the sampling of a signal as multiplication of the signal by an impulse train.



We can express the sampling of a signal as multiplication of the signal by an impulse train. But multiplication  $\Rightarrow$  convolution

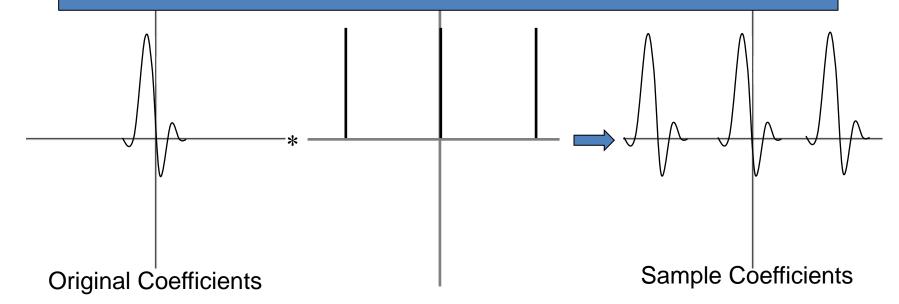




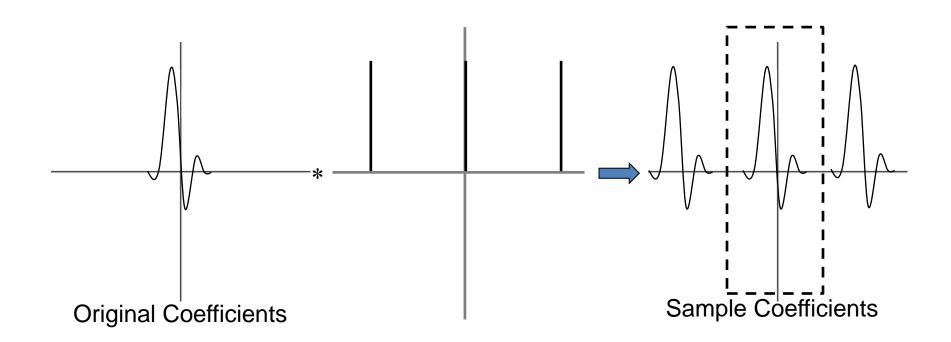
The sample coefficients are disjoint copies of the signal if:

w m

- 1. The signal is band-limited (Fourier coefficients are zero beyond some point)
- Pt 2. The sampling between impulses in the frequency domain is sufficiently far apart (sampling is fine enough).

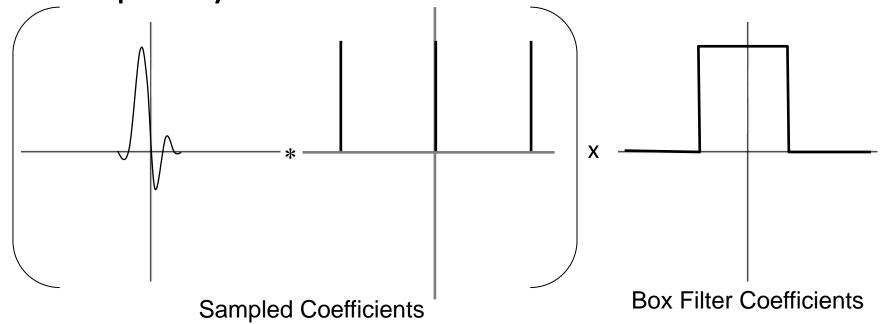


If the sampling conditions are satisfied, we can reconstruct by convolving with a filter that pulls out the center of the spectrum.



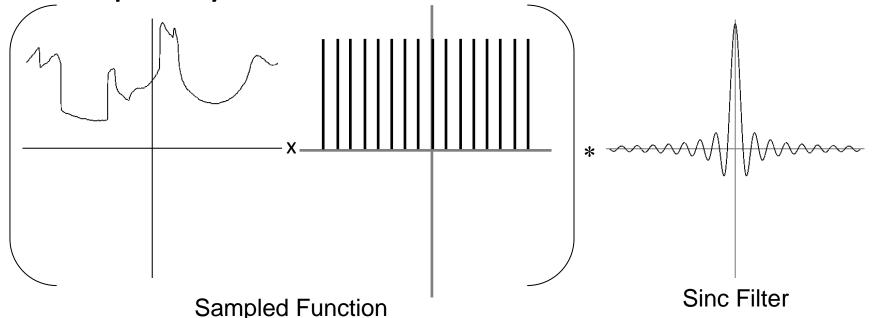
If the sampling conditions are satisfied, we can reconstruct by convolving with a filter that pulls out the center of the spectrum.

⇒ Want to multiply by a box filter in the frequency domain.



If the sampling conditions are satisfied, we can reconstruct by convolving with a filter that pulls out the center of the spectrum.

⇒ Want to multiply by a box filter in the frequency domain ⇔ Convolve with a sinc.

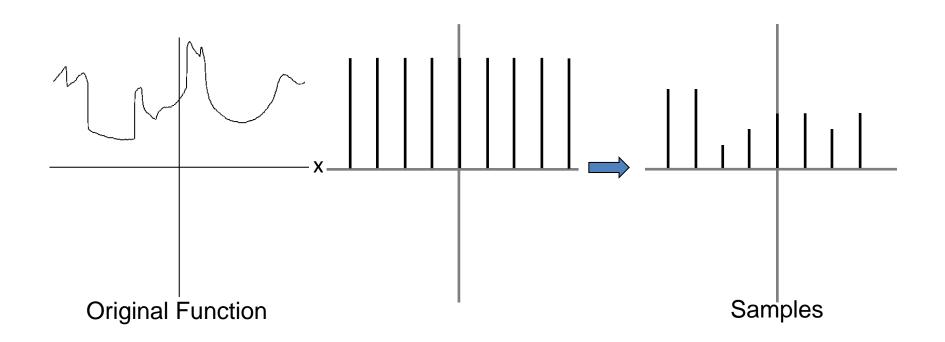


We are assuming that the input signal is bandlimited and the sampling is fine enough.

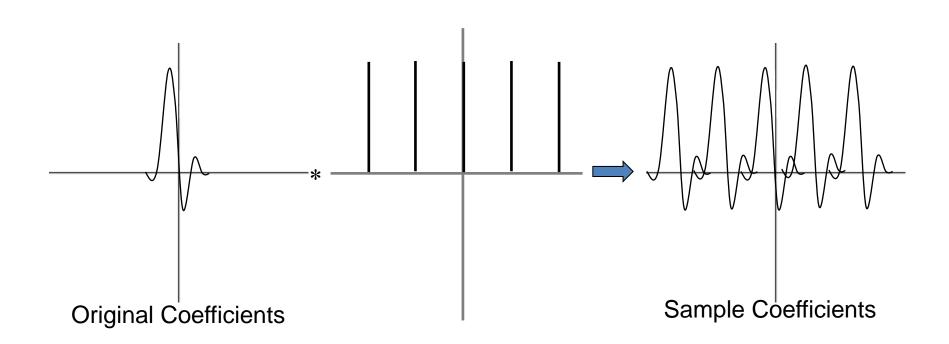
In practice, this assumption is false:

- The signal is not band-limited (occluding contours, sharp shadow boundaries, etc.)
- We are limited in the extent to which we can sample.

We are assuming that the input signal is bandlimited and the sampling is fine enough. In practice, this assumption is false.



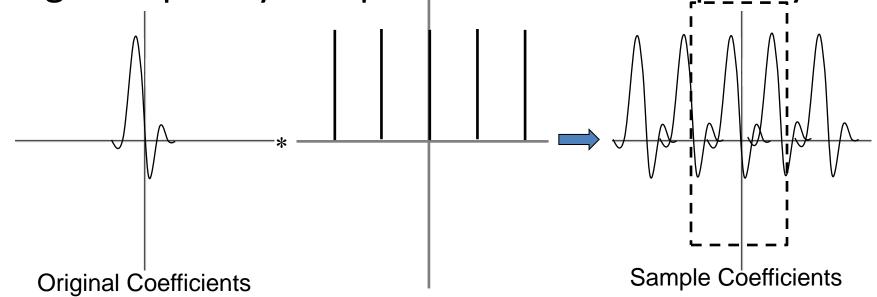
We are assuming that the input signal is bandlimited and the sampling is fine enough. In practice, this assumption is false.



We are assuming that the input signal is bandlimited and the sampling is fine enough.

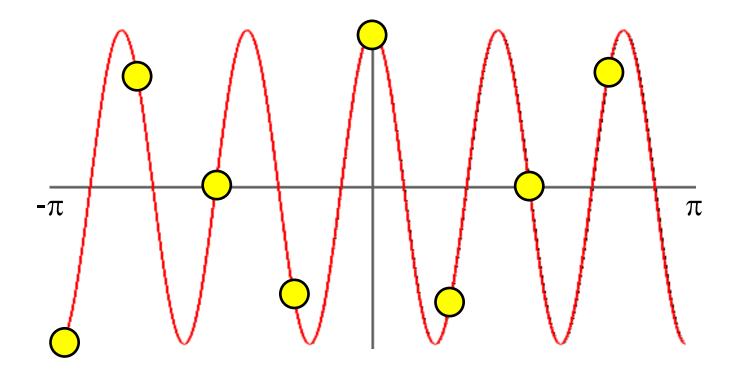
In practice, this assumption is false.

Multiplying with a box function, we pick up high-frequency components as low-frequency



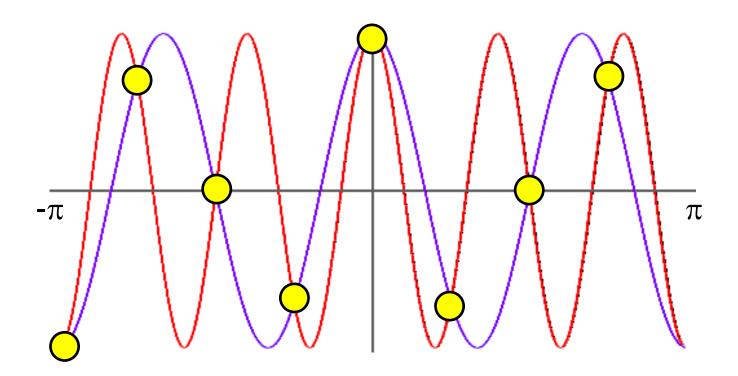
# Aliasing

 When a high-frequency signal is sampled with insufficiently many samples, it will be perceived as a lower-frequency signal. This masking of higher frequencies as lower ones is referred to as <u>aliasing</u>.



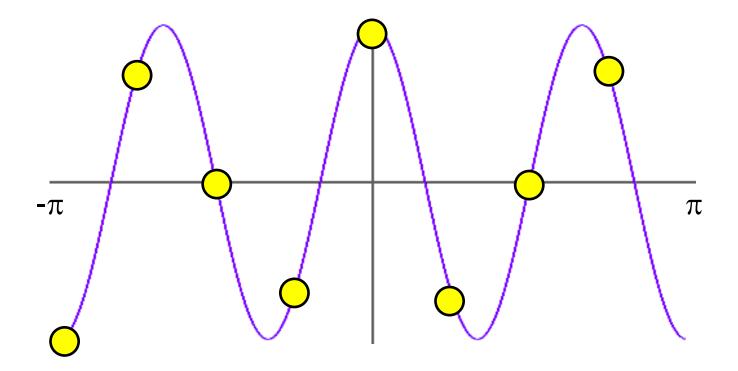
# Aliasing

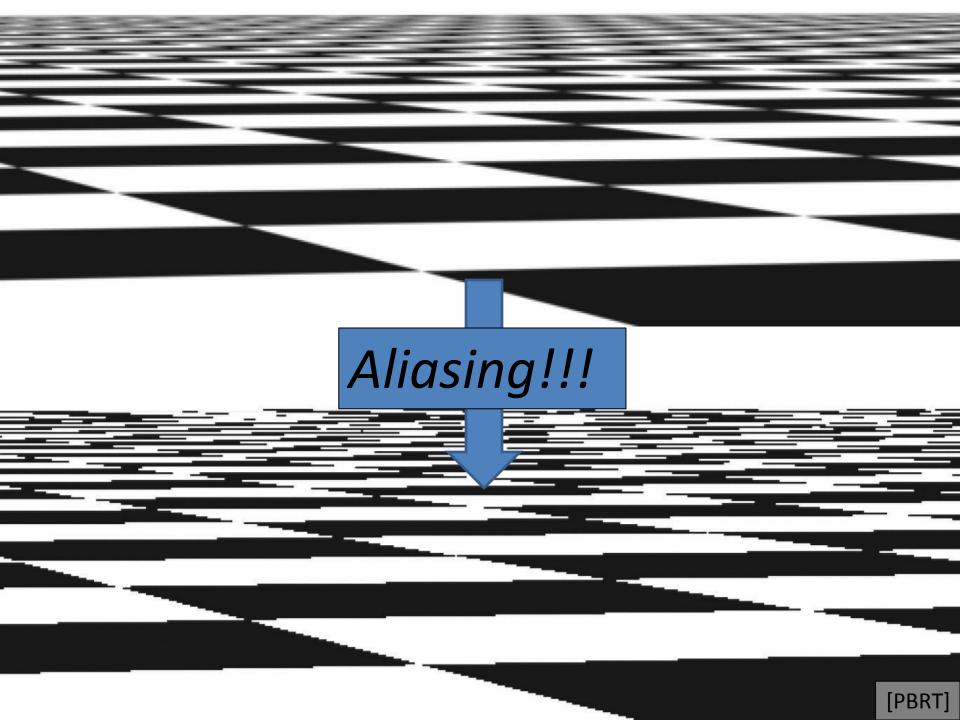
 When a high-frequency signal is sampled with insufficiently many samples, it will be perceived as a lower-frequency signal. This masking of higher frequencies as lower ones is referred to as <u>aliasing</u>.



# Aliasing

 When a high-frequency signal is sampled with insufficiently many samples, it will be perceived as a lower-frequency signal. This masking of higher frequencies as lower ones is referred to as <u>aliasing</u>.

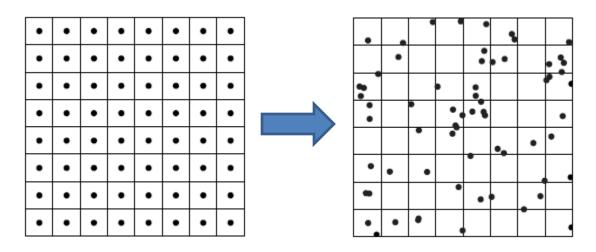




Since we can't increase the sampling rate beyond the necessary (Nyquist) frequency, our samples are bound to contain high-frequency info.

Since we can't increase the sampling rate beyond the necessary (Nyquist) frequency, our samples are bound to contain high-frequency info.

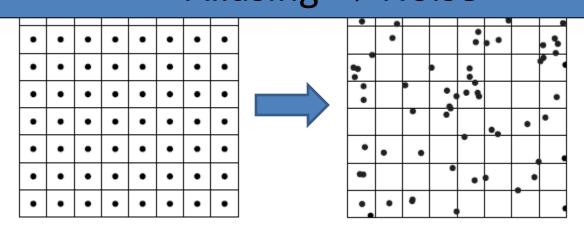
Try to remove the effects of aliasing by randomizing the sampling positions.

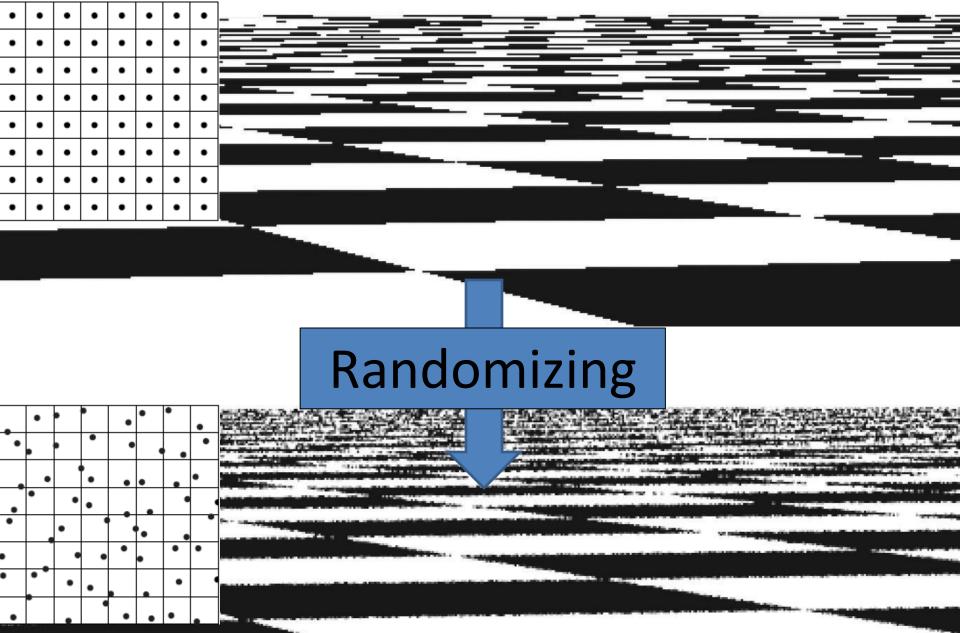


Since we can't increase the sampling rate beyond the necessary (Nyquist) frequency, our samples are bound to contain high-frequency info.

We still sample the high frequency, but we now de-correlate the phase alignment.

Aliasing  $\Rightarrow$  Noise

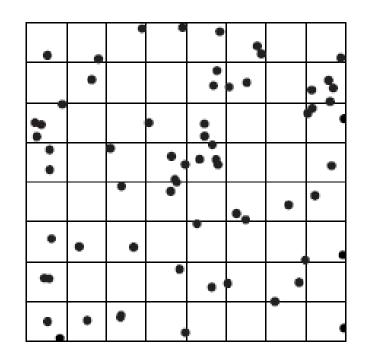


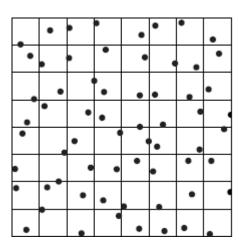


# Random Sampling

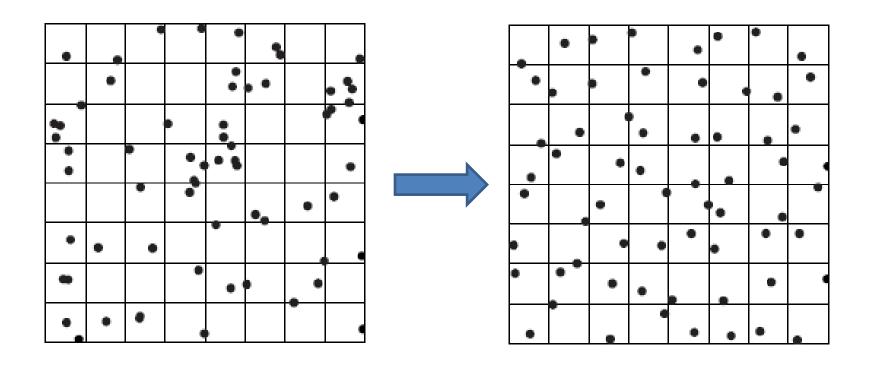
#### **Challenge:**

If we randomly sample the plane, we are likely to get some regions that are over-sampled and others that are under-sampled.

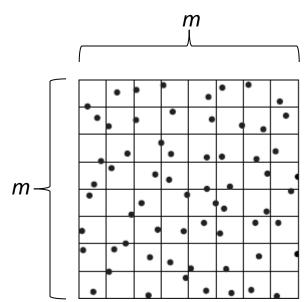




Decompose domain into regular cells and choose a sample randomly from within each cell.

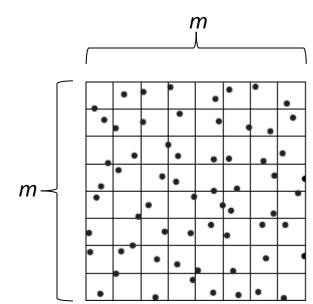


For a d-dimensional space, partitioning each dimension into m domains gives  $m^d$  samples.



#### **Limitation:**

We would like *m* to be large to have stratified samples, but for large *d* we hit an impractically large number of samples.

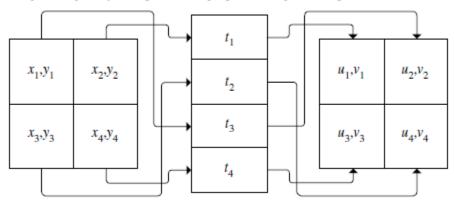


#### **Limitation**:

We would like *m* to be large to have stratified samples, but for large *d* we hit an impractically large number of samples.

#### Approach:

Separately generate stratified samples for each dimension and then combine.



#### **Limitation:**

For dimensions d>1, the number of samples has to be factorizable into d (roughly equal) factors.

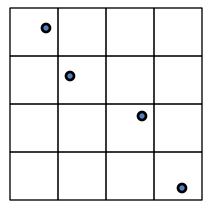
To generate m (not  $m^2$ ) samples in d dimensions:

1. Partition each dimension into *m* parts and place a random sample in each diagonal cell.

•			
	•		
		0	
			•

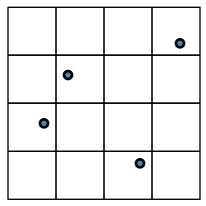
To generate m (not  $m^2$ ) samples in d dimensions:

 Partition each dimension into m parts and place a random sample in each diagonal cell. Each column/row has one sample.



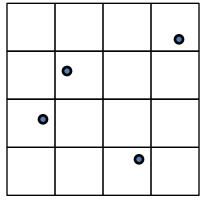
To generate m (not  $m^2$ ) samples in d dimensions:

- Partition each dimension into m parts and place a random sample in each diagonal cell. Each column/row has one sample.
- 2. Randomly permute along each dimension.



To generate m (not  $m^2$ ) samples in d dimensions:

- Partition each dimension into m parts and place a random sample in each diagonal cell. Each column/row has one sample.
- 2. Randomly permute along each dimension. Each column/row still has one sample.



# Other Sampling Methods

- 1. Low-discrepancy sampling
- 2. Best-candidate sampling
- 3. Adaptive Sampling

#### 1. Low-discrepancy sampling

 Find the sampling that minimizes the difference between the expect number of samples in a region and the actual number of samples.

#### 1. Low-discrepancy sampling

 Find the sampling that minimizes the difference between the expect number of samples in a region and the actual number of samples.
 For an integer written out in base b as:

$$n = \sum_{k=0}^{\infty} a_k b^k \quad \text{with} \quad 0 \le a_k < b$$

define the *radical inverse* to be the floating point value in the range[0,1) with:

$$\Phi_b(n) = \sum_{k=0}^{\infty} a_k b^{-1-k}$$

- 1. Low-discrepancy sampling
- 2. Best-candidate sampling
  - Generate samples that are guaranteed not to get too close.

- 1. Low-discrepancy sampling
- 2. Best-candidate sampling
  - Generate samples that are guaranteed not to get too close.
    - Pre-compute and then tile.

- 1. Low-discrepancy sampling
- 2. Best-candidate sampling
- 3. Adaptive sampling
  - Evaluate the results from the samples and determine if more samples are required.

- 1. Low-discrepancy sampling
- 2. Best-candidate sampling
- 3. Adaptive sampling
  - Evaluate the results from the samples and determine if more samples are required.
    - Samples come from different geometry
    - Sample color variation is large.

#### Reconstruction

Note that in using random sampling, we are no longer sampling at the pixel-resolution.

Once we have the samples, we would like to reconstruct a function sampled at a fixed resolution (width x height).

#### Reconstruction

Note that in using random sampling, we are no longer sampling at the pixel-resolution so we have to:

- 1. Reconstruct a function from the samples.
- 2. Sample the function at the resolution of the image.

#### Reconstruction

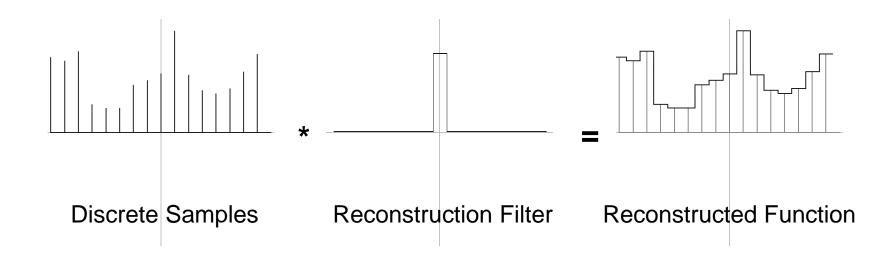
Note that in using random sampling, we are no longer sampling at the pixel-resolution so we have to:

- 1. Reconstruct a function from the samples.
- 2. Sample the function at the resolution of the image.

To avoid aliasing, we need to reconstruct with an appropriate (smoothed) filter.

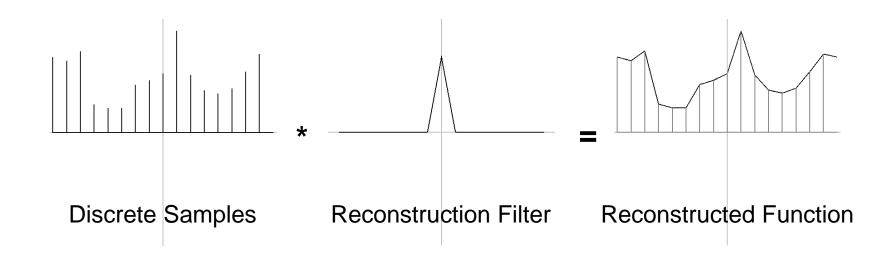
#### **Box-Filter:**

- ✓ Can be implemented efficiently because the filter is non-zero in a very small region.
- ➤ Introduces high frequency content that will cause aliasing when sampled into an image.



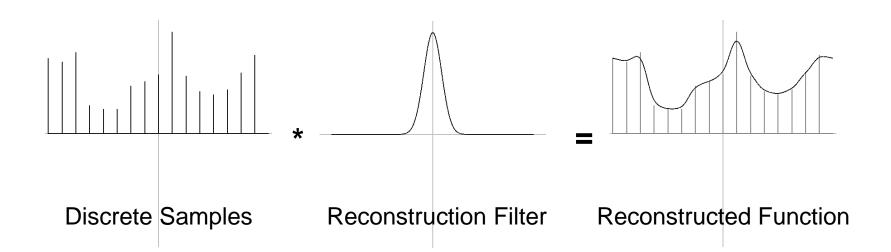
#### **Hat-Filter**:

- ✓ Can be implemented efficiently because the filter is non-zero in a very small region.
- \* Partially addresses the aliasing problem, but still introduces high frequency content that will cause aliasing when sampled into an image.



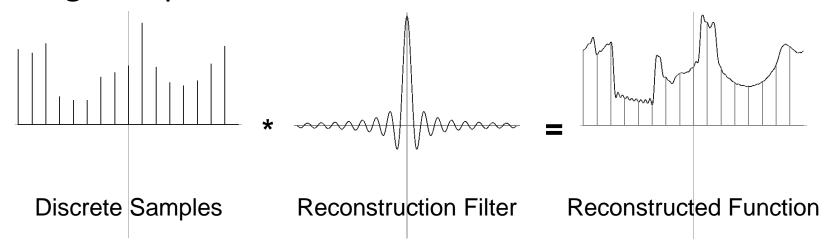
#### (Clamped) Gaussian-Filter:

- ➤ Is slow to implement because the filter is non-zero in a large region.
- ✓ Addresses the aliasing problem by killing off most of the high frequencies.



#### Sinc-Filter:

- ➤ Is slow to implement because the filter is non-zero in a large region.
- \* Assigns negative weights.
- \* Ringing at discontinuities.
- ✓ Addresses the aliasing problem by killing off the high frequencies.



#### Lanczos-Filter:

Modulates the first *k* lobes of the sinc filter with a cosine function.