

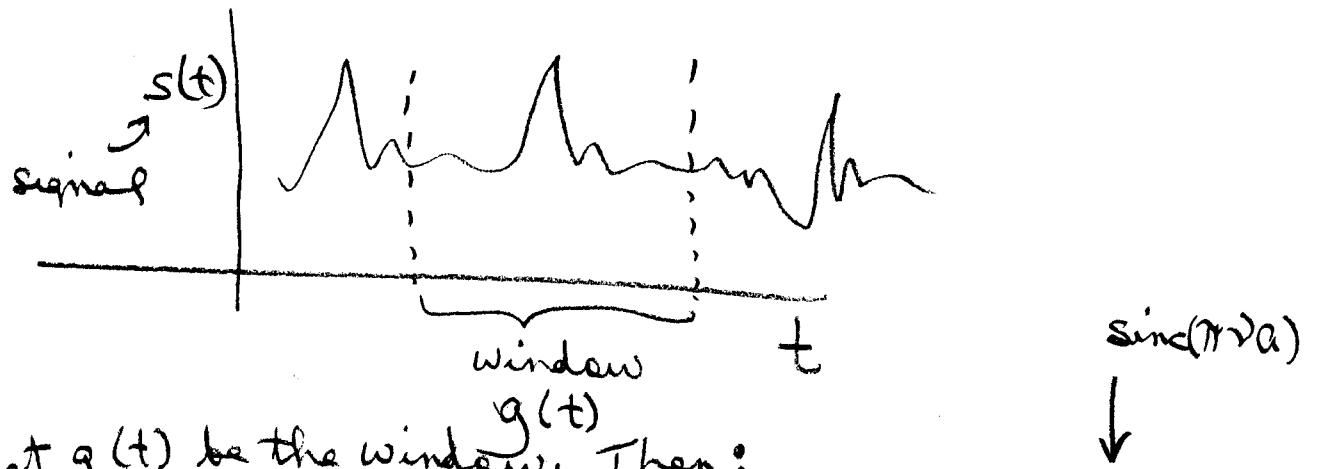
## Chapter 9 Windows & Discrete FT

(Following Fante 3.5)

### 9.1 Windows

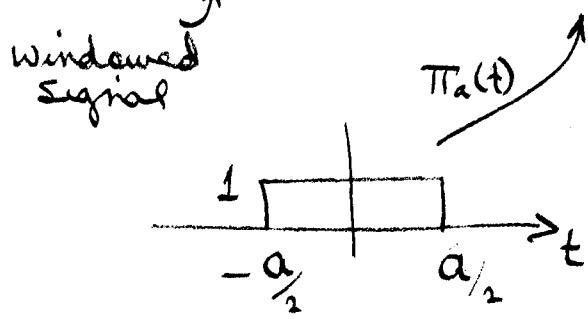
Whereas filters had to do with frequency space, windows refer to time space.

Windows in time occur unvariably and naturally: our measurements must start and stop somewhere.



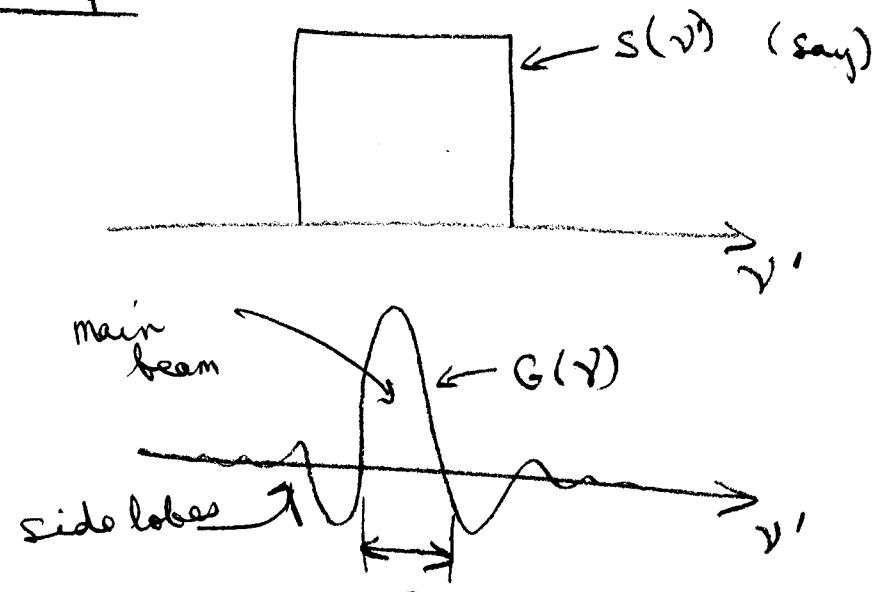
Let  $g(t)$  be the window. Then:

$$w(t) = s(t)g(t) \Leftrightarrow S(\nu) * G(\nu) \equiv W(\nu)$$

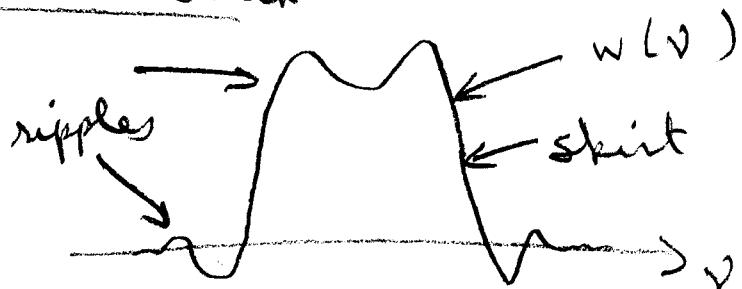


$$\begin{aligned} S(\nu) * G(\nu) &\equiv W(\nu) \\ &= \int_{-\infty}^{\infty} S(\nu') G(\nu - \nu') d\nu' \\ &= \int_{-\infty}^{\infty} S(\nu') \sin(\pi a(\nu - \nu')) d\nu' \end{aligned}$$

Example



after convolution  $\frac{2}{a}$



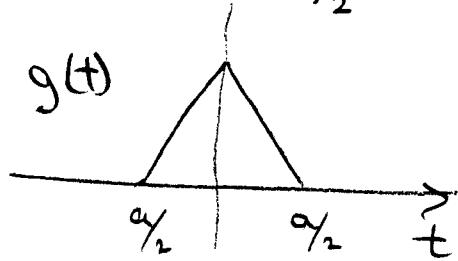
Windowing corrupts the signal. In effect, the Fourier Transform is built on the premise that  $s(t)$  is 0 at  $\pm\infty$ . Since the F.T. assumes limits of  $\pm\infty$ , the window effectively says that the signal drops to 0 at the window edges and stays 0 from then on. If the signal were finite at the window edge, then it experiences a false truncation. This introduces false oscillations in the frequency domain (ripples) and a false 'skirt'.

↑ caused by the side lobes

↑ caused by the main beam of  $\text{sinc}(\pi v(f-f'))$

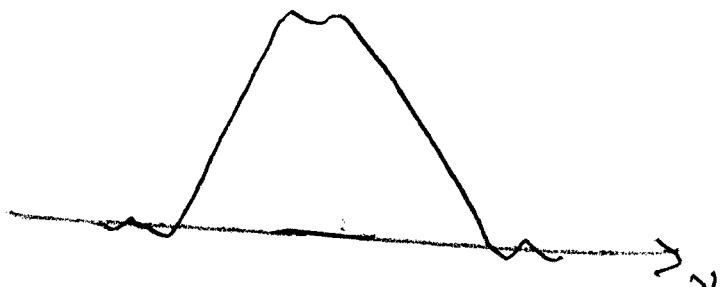
Let's try another window shape to see if we can do better. Let's try  $\Lambda_a(t)$ , the triangular window. This won't start and stop so abruptly so the ripples should be less but it will cause the loss of signal information near the window edges.

$$\text{Now, } \Lambda_{\alpha_2}(t) \Leftrightarrow \left(\frac{\alpha}{2}\right)^2 \sin^2(\pi \nu \alpha_2)$$



Thus :

$$w(\nu) \text{ is}$$



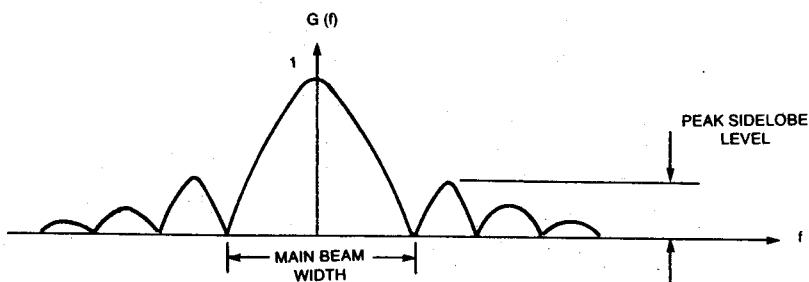
Note: the ripples are smaller but the skirt is bigger (more tapered) (by making  $g=0$  at edges)

In general, the ripples can be reduced, but at the expense of a broader skirt (due to loss of info near the edges).

A number of common windows are as follows:

from Fante pages 84-85

TABLE 3.4 Properties of Some Commonly Used Windowing Functions



Window $g(t)$	<i>Fourier Transform <math>G(f)</math></i>	Width of Main Beam (between Zeros)	Peak Sidelobe Amplitude (Relative to Main Beam Peak)
$\begin{aligned} g(t) &= 1, \quad  t  \leq \frac{T_o}{2} \\ &= 0, \quad  t  > \frac{T_o}{2} \end{aligned} \quad (\text{rectangular})$	$\frac{2}{T_o}$	0.22	
$\begin{aligned} g(t) &= 1 - \frac{2 t }{T_o}, \quad  t  \leq \frac{T_o}{2} \\ &= 0, \quad  t  > \frac{T_o}{2} \end{aligned} \quad (\text{Bartlett})$	$\frac{4}{T_o}$	0.056	
$\begin{aligned} g(t) &= \cos^2\left(\frac{\pi t}{T_o}\right), \quad  t  \leq \frac{T_o}{2} \\ &= 0, \quad  t  > \frac{T_o}{2} \end{aligned} \quad (\text{Hanning})$	$\frac{4}{T_o}$	0.028	
$\begin{aligned} g(t) &= 0.54 + 0.46 \cos \frac{2\pi t}{T_o}, \quad  t  \leq \frac{T_o}{2} \\ &= 0, \quad  t  > \frac{T_o}{2} \end{aligned} \quad (\text{Hamming})$	$\frac{4}{T_o}$	0.0089	
$\begin{aligned} g(t) &= 0.42 + 0.5 \cos \frac{2\pi t}{T_o} + 0.08 \cos \frac{4\pi t}{T_o}, \quad  t  \leq \frac{T_o}{2} \\ &= 0, \quad  t  > \frac{T_o}{2} \end{aligned} \quad (\text{Blackman})$	$\frac{6}{T_o}$	0.0014	
$g(t) = \frac{I_0[\pi\beta\sqrt{1-(2t/T_o)^2}]}{I_0(\pi\beta)}, \quad  t  \leq \frac{T_o}{2} \quad (\text{Taylor})$	$\frac{2(\beta^2+1)^{1/2}}{T_o}$	$0.22\left(\frac{\sinh \pi\beta}{\pi\beta}\right)^{-1}$	

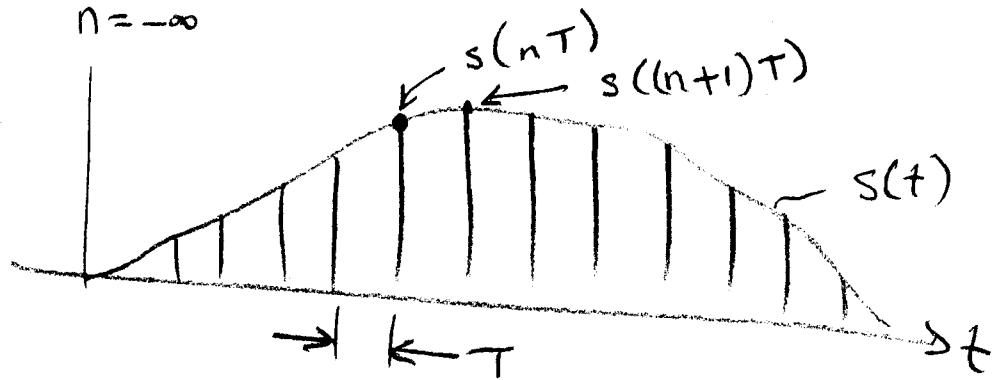
$I_0$  = modified Bessel function

## 9.2 Discrete Fourier Transforms (following Fante chapter 5)

There is a whole parallel development for the discrete (ie sampled) signal as for the continuous signal. We have already met the sampled signal in our discussion on aliasing in chapter 6.

Recall that we showed that

$$\hat{S}(v) = \sum_{n=-\infty}^{\infty} s(nT) e^{-2\pi i v n T}$$



We showed how the FT. of  $s(nT)$  is periodic and that  $\hat{S}(v) = \frac{1}{T} \sum_{n=-\infty}^{\infty} s(v + n/T)$  so that

we need to limit  $v$  or increase the sampling rate  $1/T$  if we are to capture  $s(v)$  faithfully. If  $B$  is the bandwidth of  $s(v)$ , then we must respect:  $\frac{1}{T} \geq B$  (Nyquist criteria)

But what if we want to reconstruct  $s(t)$  from  $\hat{s}(\nu)$ . We will have  $\hat{s}(\nu)$  in some discrete form in a computer. So in doing the inverse FT of a discrete signal,  $\hat{s}(\nu)$ , we meet the same situation as we found when we tried to do the F.T. of  $s(nT)$  - only this time the roles of  $\nu$  and  $t$  are reversed.

So we expect the same aliasing problem - the reconstructed  $s(t)$  will be replicated in an erroneous periodic fashion.

The criteria  $\frac{1}{T} \geq B$  translates to

$$\frac{1}{\Delta\nu} \geq T_0 \text{ ie } \Delta\nu \geq \frac{1}{T_0}$$

or  $\Delta\nu = \frac{1}{T_0}$  at the critical sampling rate.

So if we had sampled for 1 second, the frequency resolution is

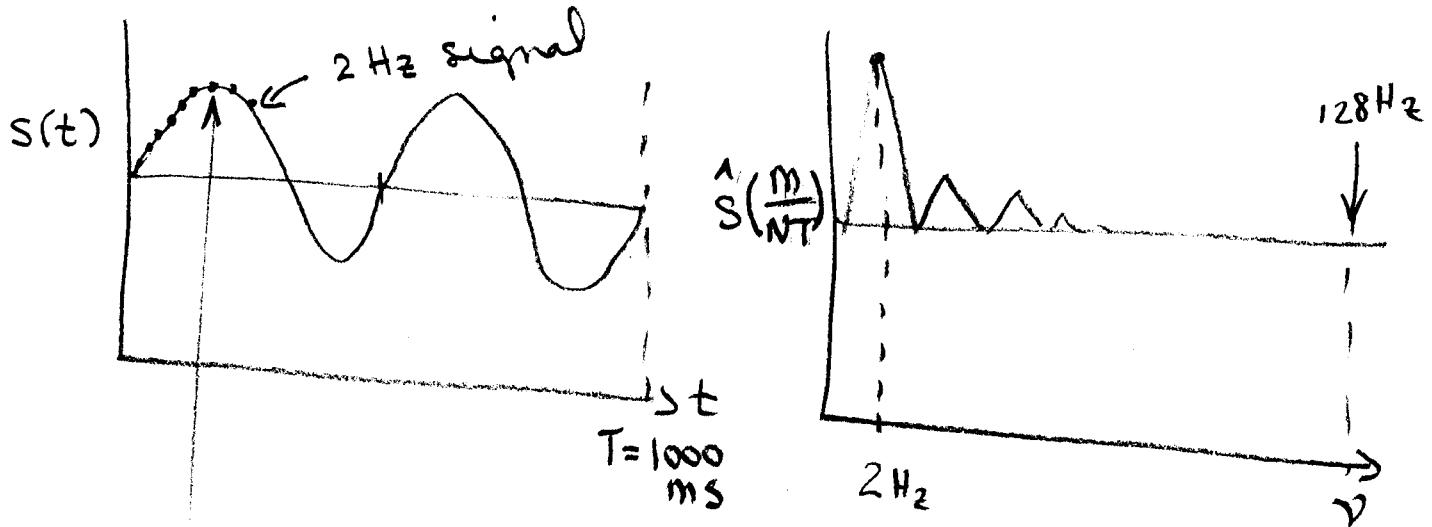
$$\Delta\nu = 1 \text{ Hz.}$$

Thus rather than working with  $\hat{s}(\nu)$  we deal with  $\hat{s}$  in steps of  $\Delta\nu$ , ie

$$\hat{s}(m\Delta\nu) \text{ or } \hat{s}\left(\frac{m}{T_0}\right) = \hat{s}\left(\frac{m}{NT}\right)$$

Note:  $T$  = sampling rate  
 $\Rightarrow \Delta\nu$   
 $B$  = bandwidth,  
 $\Rightarrow$  time window,  $T_0$

So you will see in commercial code the following displays:



Samples, say 256 samples  
for the 1000 ms window

$$\Rightarrow T = \frac{1}{256} \text{ s sampling rate}$$

$\therefore$  highest frequency  
(Nyquist limit)

$$\text{is } \frac{B}{2} = \frac{1}{2T} = \frac{1}{2 \times \frac{1}{256}} = \frac{256}{2} = 128 \text{ Hz}$$

9-2d

One final point, you will also meet 'zero padding' in commercial code. Zero padding simply adds zeros at the end of the time signal.

Let's double the signal length from  $N$  samples to  $2N$  samples. Then we have

$$\hat{S}\left(\frac{m}{2NT}\right) = \sum_{n=0}^{2N-1} s(nT) e^{-2\pi i \frac{mn}{2NT}}$$

$\nwarrow$  this is a windowed case

$$= \sum_{n=0}^{N-1} s(nT) e^{-2\pi i \left(\frac{m}{2}\right)\left(\frac{n}{N}\right)}$$

Compare this to the original:

$$\hat{S}\left(\frac{m}{NT}\right) = \sum_{n=0}^{N-1} s(nT) e^{-2\pi i \frac{mn}{N}}$$

The only difference is  $m \rightarrow m/2$ , ie we interpolate the spectrum.

So zero padding is used to interpolate a signal to get a smoother appearance - but be warned: It does not increase the resolution since that is limited by  $\Delta\nu = \frac{1}{T_0}$ .

9.3 Recap

Window =  $g(t)$ , Signal =  $s(t)$

$$\omega = s \cdot t \implies s(v) * G(v) = w(v)$$

$$\int_{-\infty}^{\infty} s(v') G(v-v') dv'$$

$$\text{if } S(v) = \pi \Rightarrow w(v) = \begin{cases} \text{skirt} \\ \text{ripple} \end{cases}$$

Try to choose  $g(t)$  to reduce ripple without creating too much skirt.

DFT

$$\hat{s}\left(\frac{m}{N\tau}\right) = \sum_{n=-\infty}^{\infty} s(n\tau) e^{-2\pi i \frac{mn}{N}}$$

$$\frac{1}{\tau} > B + \frac{1}{\Delta v} \geq T_0 \quad (\text{frequency resolution limit})$$

↑ aliasing limit

Use zero padding to get smoother appearance.

—The End—