### **DISCRETE FOURIER TRANSFORM**



- We should find a way to calculate (at least approximately) the Fourier transform of any signal on a computer (or a DSP)
- This implies the following:
  - The spectrum should be calculated on the basis of the samples of the input signal
  - Only a *finite number* of samples should be used for calculation
  - The results should be obtained as a discrete array of values
  - The calculation should be performed as efficiently as possible (in terms of speed and memory)
  - The influence of the finite word length should be minimized

### Limiting the duration of the signal (windowing)



### Sampling of the signal



 Calculation of the spectrum on the basis of signal samples is equivalent to using DTFT instead of classical FT

$$\overline{X}(f) = \int_{-\infty}^{\infty} \overline{x}(t) e^{-j2\pi ft} dt \quad \longrightarrow \quad \overline{X}(\xi) = \sum_{n=-\infty}^{\infty} \overline{x}(n) e^{-j2\pi\xi n}$$

### Sampling of the signal

• The expression for DTFT is obtained from the expression for the spectrum of a sampled continuous-time signal:

$$\hat{X}(f) = \sum_{n=-\infty}^{\infty} x(nT) e^{-j2\pi f nT} = \frac{1}{T} \sum_{m=-\infty}^{\infty} X(f - mf_s)$$

- If the condition of the sampling theorem is not met, *aliasing* occurs (copies of the original spectrum overlap)
- The high-frequency region of the spectrum is usually the one more affected by aliasing
- It becomes impossible to extract the original spectrum even by using an ideal low-pass filter

### **Spectrum discretization**



• Spectrum is calculated only in a finite number of points  $\xi_k = k/N$ , k = 0, 1, ..., N-1.  $X(k) = \sum_{n=0}^{N-1} x_p(n) e^{-j2\pi \frac{k}{N}n} \qquad x_p(n) = \sum_{m=-\infty}^{\infty} \overline{x}(n-mN)$ 



$$\begin{aligned} x(n) &= x(n)w(n) \\ \overline{X}(\xi) &= \sum_{n=-\infty}^{\infty} x(n)w(n)e^{-j2\pi\xi n} = \sum_{n=0}^{N-1} x(n)w(n)e^{-j2\pi\xi n} \approx X(\xi) \\ \overline{X}(\xi) &= X(\xi) \circledast W(\xi) = \int_{-1/2}^{1/2} X(\lambda)W(\xi - \lambda)d\lambda \end{aligned}$$

- Convolution distorts the original spectrum
- This influence will be analyzed for the discrete-time case, which does not make a big difference

### **Rectangular window function**



Zeros of  $W(\omega)$  are located at frequencies  $\omega = \frac{2\pi}{N}k$ ,  $k = \pm 1, \pm 2,...$ 

$$\frac{W(\omega)}{W(0)}\Big|_{\omega=\frac{3\pi}{N}} = \left|\frac{\sin (\pi/2)}{\sin (\pi/2)}\right| : N \approx \frac{1}{N \cdot (3\pi/2)} = \frac{2}{3\pi} \approx 0.21 \qquad 20 \log(0.21) \approx -13 dB$$

## Frequency resolution

- The width of the main arcade of the window function defines the *frequency resolution*
- The minimum distance between two components which will still be distinguishable in the spectrum is:

$$\Delta \omega \ge \frac{2\pi}{N} \qquad \Delta f \ge \frac{f_s}{N} = \frac{1}{NT} = f_2 - f_1 \quad [\text{Hz}]$$

## Frequency resolution

• Minimum length of the rectangular window function for the desired frequency resolution  $\Delta f(\Delta \omega)$  is:

$$N \geq \frac{f_s}{\Delta f} = \frac{2\pi}{\Delta \omega}$$

- How to improve frequency resolution (reduce  $\Delta f = f_s / N$ )?
  - Decrease f<sub>s</sub> (increases the effects of sampling)
  - Increase N (requires more calculations, and sometimes it is not even possible due to limited availability of samples)
- Improved frequency resolution decreases temporal resolution

# Spectral leakage (DTFT)

- Consequence of the existence of side lobes in the spectrum of the window function
- The spectrum of the windows signal may contain components even at frequencies at which the original spectrum was zero
  - Main arcades of weaker spectral component may be completely hidden



- The effect may be reduced by using better window functions
  - There are window functions with sidelobes suppressed by nearly 100 dB

# Spectral leakage (DFT)

- It remains to be seen whether the DFT samples of the DTFT spectrum will faithfully represent the amplitudes of peaks in the DTFT spectrum
  - Convolution of the spectrum of the window function with a component in the original spectrum should appear at the location of a DFT sample
  - DFT samples are located at frequencies  $\omega_k = 2k\pi/N$
  - The components in the original spectrum should also appear at  $\omega_k = 2k\pi/N$  in order to be faithfully represented by the DFT spectrum



## Bartlett (triangular) prozorska funkcija



- Side lobes are attenuated by 26 dB
- Main lobe is approximately twice as wide, which deteriorates frequency resolution approximately twice
  - This can be compensated by taking twice as many samples

$$\Delta \omega \geq \frac{4\pi}{N} \qquad \Delta f \geq 2\frac{f_s}{N} \qquad N \geq 2\frac{f_s}{\Delta f}$$

## Some other window functions

#### Hann window function

$$w(n) = \begin{cases} 0,5-0,5\cos\frac{2\pi n}{N-1}, & 0 \le n \le N-1 \\ 0, & \text{elsewhere} \end{cases}$$

- Side lobe attenuation: 31 dB
- Frequency resolution as with Bartlett window

#### Hamming window function

$$w(n) = \begin{cases} 0,54 - 0,46\cos\frac{2\pi n}{N-1}, & 0 \le n \le N-1 \\ 0, & \text{elsewhere} \end{cases}$$



- Side lobe attenuation: 43 dB
- Frequency resolution as with Bartlett window

## Spectra of window functions



## Frequency resolution (example 1)

- The signal containing 4 sinusoids at frequencies 1, 1.5, 2.5 and 2.75 kHz, is sampled with  $f_s = 10$  kHz. What is the minimum number of samples of this signal to be taken for the 4 spectral peaks to remain visible:
  - a) if rectangular window is used;
  - b) if Hamming window is used?

a) 
$$N \ge \frac{f_s}{\Delta f} = \frac{10}{0.25} = 40$$
 samples  
b)  $N \ge 2\frac{f_s}{\Delta f} = \frac{20}{0.25} = 80$  samples

## Frequency resolution (example 2)

 A 10-ms segment of a continuous-time signal is sampled with f<sub>s</sub> = 10 kHz. The signal contains sinusoids at f<sub>1</sub> = 1 kHz and f<sub>2</sub> = 2 kHz as well as at a frequency f<sub>3</sub> which lies between f<sub>1</sub> and f<sub>2</sub>. How close can f<sub>3</sub> get to f<sub>1</sub> or f<sub>2</sub>, for individual spectral peaks to remain visible, if a rectangular window is used?

$$N = f_s \cdot \Delta t = 10 \text{ kHz} \cdot 10 \text{ ms} = 100 \text{ samples}$$
$$\Delta f = \frac{f_s}{N} = 100 \text{ Hz}$$
$$f_1 + \Delta f = 1.1 \text{ kHz} \le f_3 \le 1.9 \text{ kHz} = f_2 - \Delta f$$

## How to obtain a discrete spectrum?

- Discrete nature of a spectrum generally comes from periodicity in the signal
  - Periodic continous-time signals can have infinitely many harmonics in the spectrum
  - Periodic discrete-time signals can have only a finite number of them
    - k-th component in the spectrum corresponds to the frequency  $2k\pi/N$
    - After N components they begin repeating



### Fourier expansion of a periodic signal

$$x_{p}(n) = \frac{1}{N} \sum_{k=0}^{N-1} X(k) e^{j \frac{2\pi}{N} nk}$$

What are the values of coefficients X(k)?

$$x_{p}(n) = \frac{1}{N} \sum_{k=0}^{N-1} X(k) W_{N}^{-nk} \qquad W_{N} = e^{-j\frac{2\pi}{N}}$$

$$\sum_{n=0}^{N-1} x_p(n) W_N^{nr} = \frac{1}{N} \sum_{n=0}^{N-1} \sum_{k=0}^{N-1} X(k) W_N^{-(k-r)n}$$
$$= \sum_{k=0}^{N-1} X(k) \left[ \frac{1}{N} \sum_{n=0}^{N-1} W_N^{-(k-r)n} \right] = X(r)$$

$$X(k) = \sum_{n=0}^{N-1} x_p(n) W_N^{nk}$$

## Discrete Fourier transform (DFT)

 Mapping of a periodic discrete signal x(n) of period N into a sequence of complex numbers:

$$X(k) = \sum_{n=0}^{N-1} x(n) e^{-j\frac{2\pi}{N}nk}$$

- The obtained sequence is also periodic with period N
- Inverse DFT is given by:

$$x(n) = \frac{1}{N} \sum_{k=0}^{N-1} X(k) e^{j \frac{2\pi}{N} nk}$$

## Relationship between DFT and FTD

 DFT actually represents a discrete version of DTFT, since:

$$X(k) = \overline{X}(\omega)\Big|_{\omega=k\frac{2\pi}{N}}$$

where  $\overline{x}(n)$  is the signal which contains only the initial period of the signal x(n)



### Properties of DFT

### Periodicity

$$X(k) = X(k+N)$$

### Linearity

$$DFT{ax(n) + by(n)} = aX(k) + bY(k)$$

### **Time shifting**

$$\mathsf{DFT}\{x(n-m)\} = e^{-j\frac{2\pi}{N}km}X(k)$$

**Modulation** 

DFT{
$$e^{j\frac{2\pi}{N}n!}x(n)$$
} = X(k-1)

### DFT of a real signal

If the discrete-time signal x(n) is real,

 $X(N-k) = X(-k) = X^*(k)$  (Hermitian symmetry)

This amounts to:  $\operatorname{Re}\{X(-k)\} = \operatorname{Re}\{X(k)\}$  $\operatorname{Im}\{X(-k)\} = -\operatorname{Im}\{X(k)\}$ 

which is equivalent to:

$$|X(-k)| = |X(k)|$$
  
arg{X(-k)} = -arg{X(k)}

### **Transform of convolution**

$$\mathsf{DFT}\{x(n) \circledast y(n)\} = X(k)Y(k)$$

### **Transform of product**

$$\mathsf{DFT}\{x(n)y(n)\} = \frac{1}{N}X(k) \circledast Y(k)$$

Parseval's theorem

$$\sum_{n=0}^{N-1} |x(n)|^2 = \frac{1}{N} \sum_{k=0}^{N-1} |X(k)|^2$$