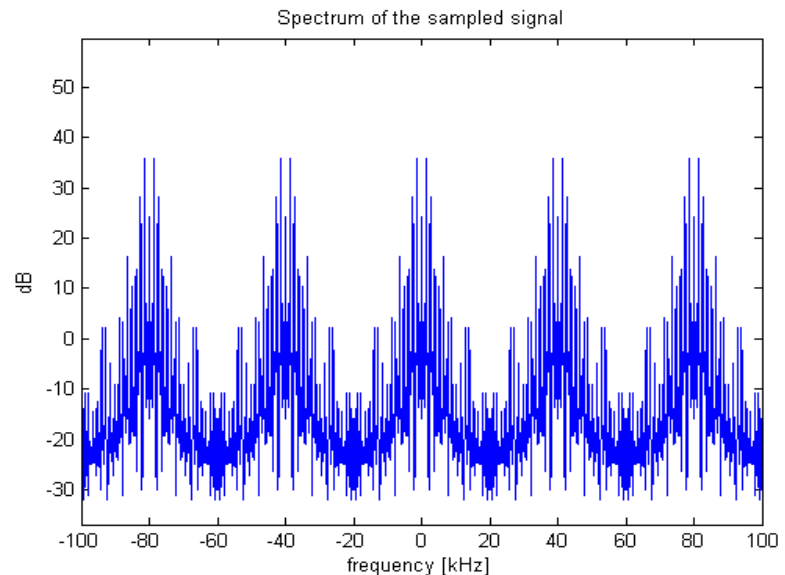
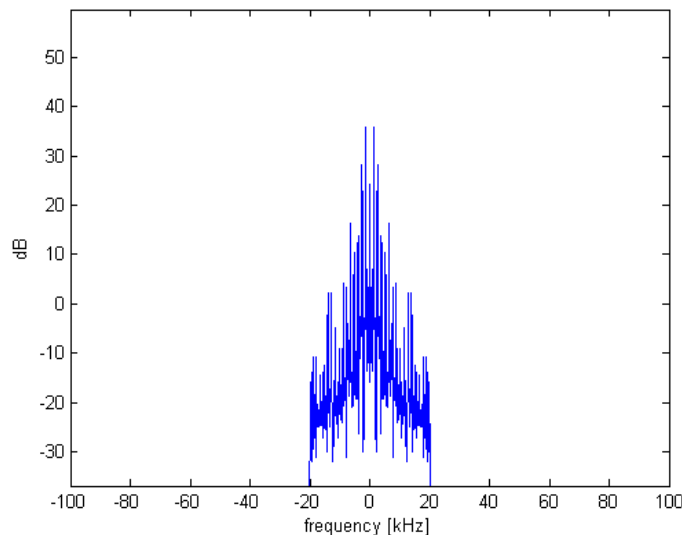


# Sampling theorem

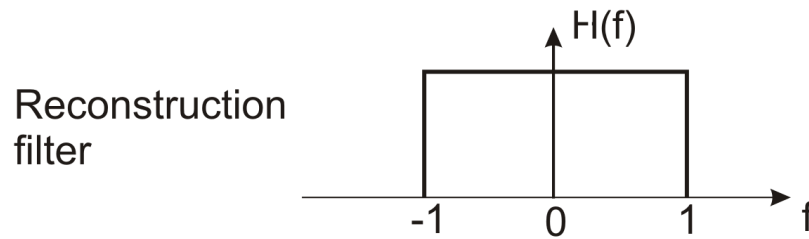
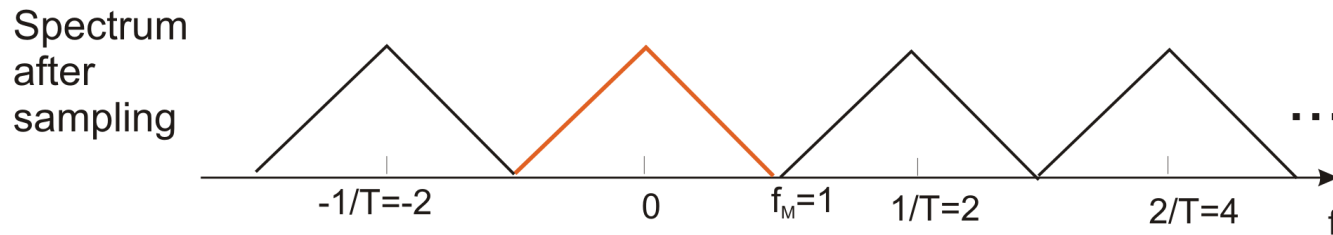
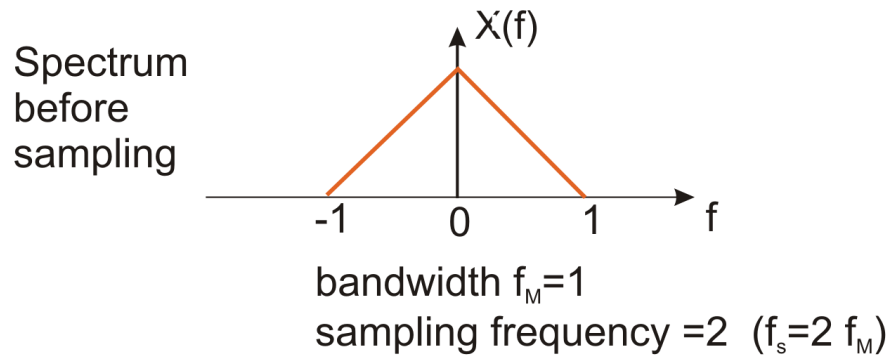
Given: spectrum (Fourier transform) of audio signal (on the left).

Find: minimum sampling frequency, enabling perfect reconstruction of audio signal

Solution: Bandwidth  $B=20$  kHz, so sampling frequency  $f_s = 1/T = 2B = 40$  kHz  
spectra copies don't overlap, so perfect reconstruction is possible

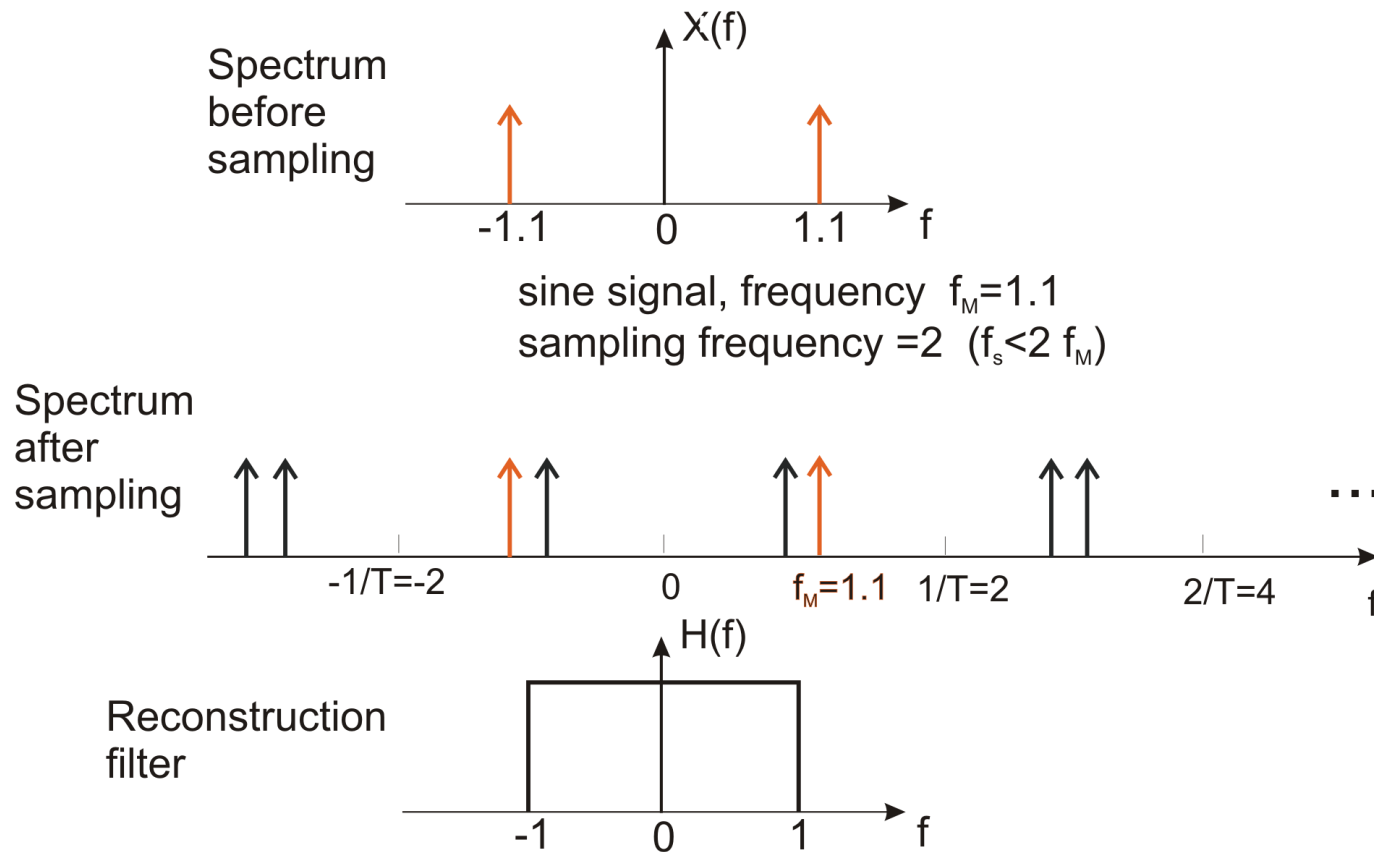


# Sampling



It is **possible** to reconstruct signal from its samples

# Sampling

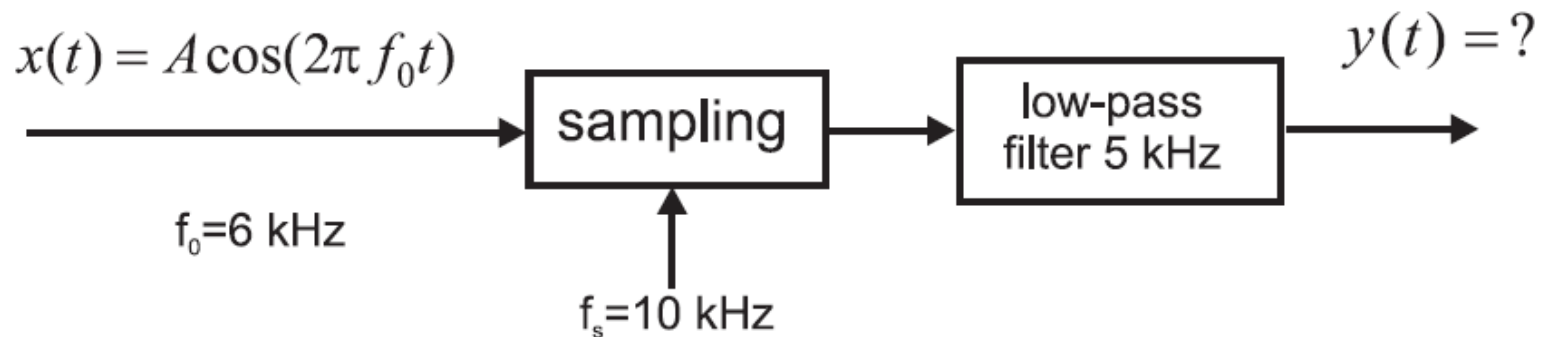


It is **not possible** to reconstruct signal from its samples

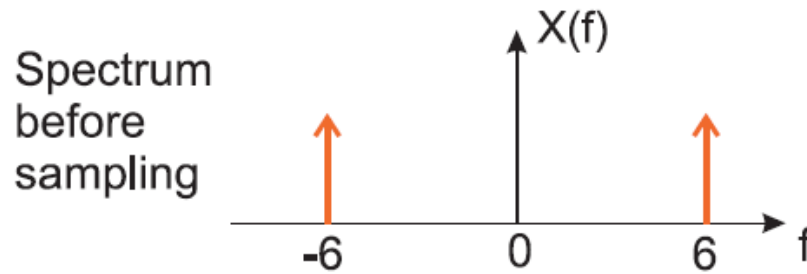
At the output of the lowpass filter we obtain cosine of frequency 0.9 kHz

# Sampling

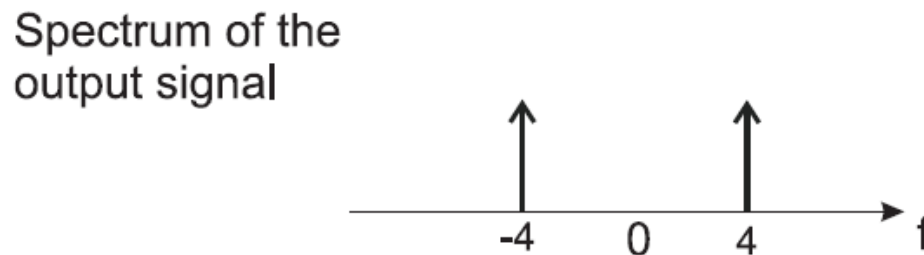
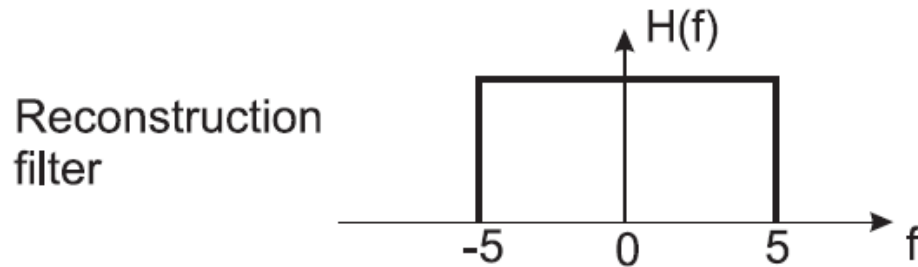
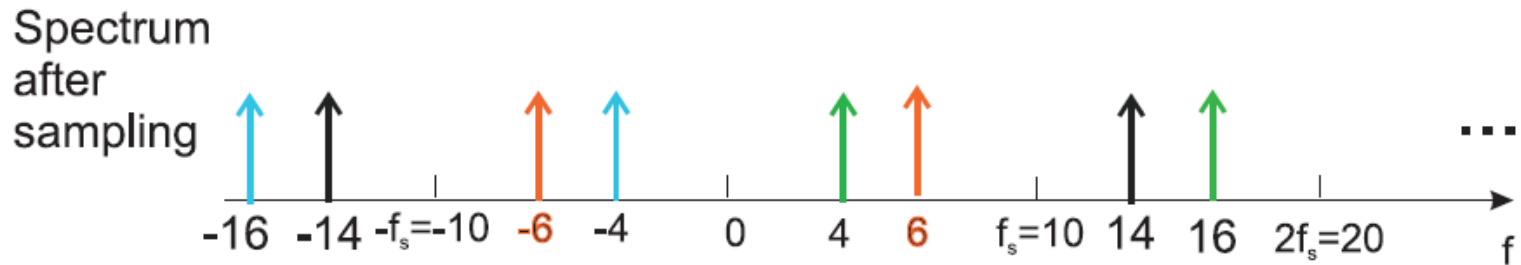
Cosine signal of frequency  $f_0 = 6$  kHz is sampled at a sampling frequency  $f_s = 1/T = 10$  kHz. Draw the spectrum of the sampled signal. Is it possible to obtain an exact copy of the continuous signal from the sampled signal? What is the signal at the output of the lowpass filter?



Solution in the next slide:



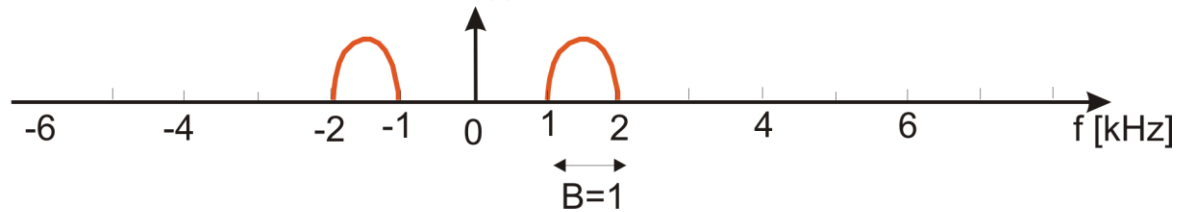
sine signal, frequency  $f_M = 6$  kHz  
 sampling frequency  $f_s = 10$  kHz ( $f_s < 2 f_M$ )



At the output of the lowpass filter we obtain cosine of frequency 4 kHz

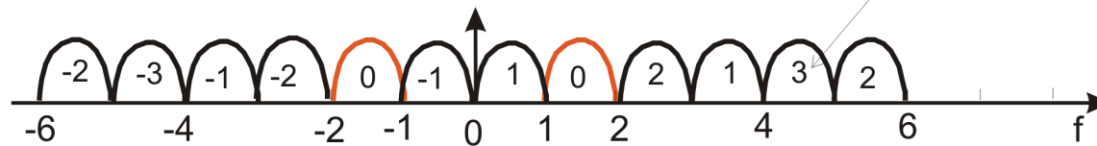
# Integer sampling

spectrum of input signal  $X(f)$

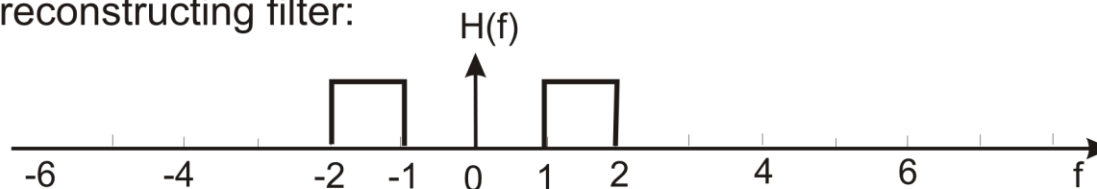


sampling frequency  $1/T=2B=2$

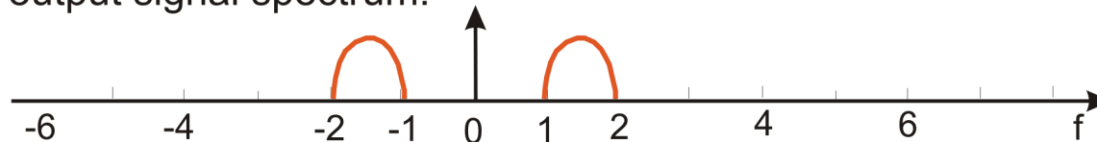
spectrum of samples  $\frac{1}{T} \sum_n X(f - \frac{n}{T}) = 2 \sum_n X(f - 2n)$



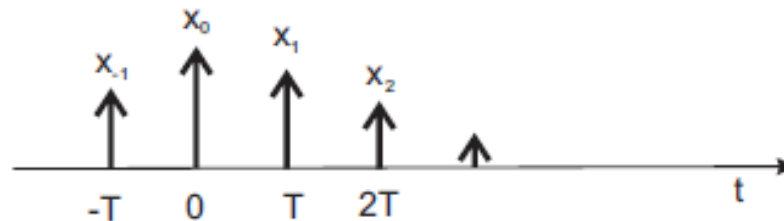
reconstructing filter:



output signal spectrum:  $H(f)$



# Discrete Time Fourier Transform



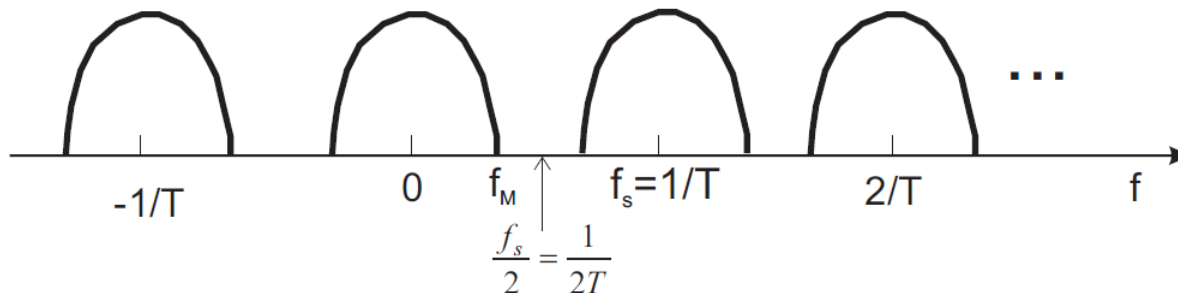
$T$  – sampling period  
 $1/T$  – sampling frequency

$$x_s(t) = \sum_n x_n \delta(t - nT)$$

Spectrum

$$X_s(f) = F[x_s(t)] = \sum_n x_n e^{-j2\pi f n T}$$

$X_s(f)$  a periodic function (period  $f_s = 1/T$ ), called the Discrete Time Fourier Transform

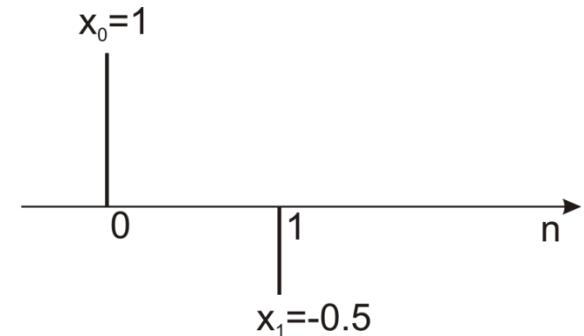


# Discrete Time Fourier Transform (DTFT) an example

$$X_s(f) = \sum_n x_n e^{-j2\pi f n T}$$

Discrete time signal  $\{x_n\}$  has only 2 nonzero values:

$$x_0 = 1, x_1 = -0.5$$



DTFT equals: 
$$X_s(f) = x_0 e^{-j2\pi f \cdot 0 \cdot T} + x_1 e^{-j2\pi f \cdot 1 \cdot T} = 1 - 0.5 e^{-j2\pi f T}$$

We may rewrite it: 
$$X_s(f) = 1 - 0.5 \cos(2\pi f T) + 0.5 j \sin(2\pi f T) = |X_s(f)| e^{j \arg(X_s(f))}$$

Amplitude spectrum 
$$\begin{aligned} |X_s(f)| &= \sqrt{(1 - 0.5 \cos(2\pi f T))^2 + (0.5 \sin(2\pi f T))^2} = \\ &= \sqrt{1 - \cos(2\pi f T) + 0.25} = \sqrt{\frac{5}{4} - \cos(2\pi f T)} \end{aligned}$$

where  $T = 1/f_s$  - sampling period,  $f_s$  - sampling frequency



# Discrete Fourier Transform (DFT)

DFT = sampled DTFT at frequencies  $f_k = \frac{k}{TL}$ ,  $k = 0, 1, \dots, L-1$   
 DTFT is calculated for L signal samples



$$X_k = \sum_{n=0}^{L-1} x_n e^{-j2\pi f_k nT} = \sum_{n=0}^{L-1} x_n e^{-j2\pi \frac{k}{TL} nT} = \sum_{n=0}^{L-1} x_n e^{-j2\pi \frac{kn}{L}}$$

Let's substitute  $W_L = e^{-j\frac{2\pi}{L}}$

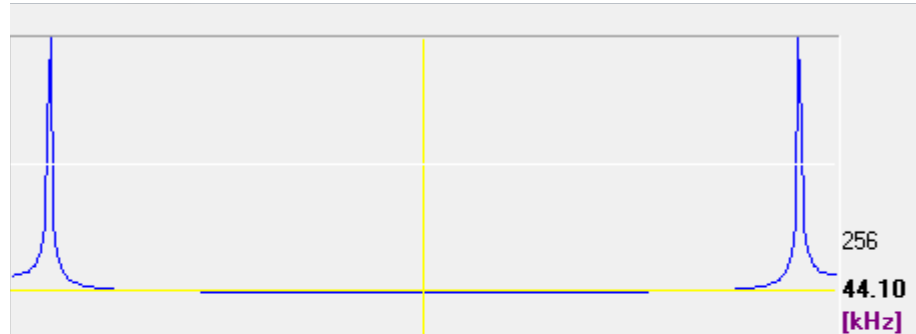
$$\text{DFT: } X_k = \sum_{n=0}^{L-1} x_n W_L^{kn}, \quad k = 0, 1, \dots, L-1$$

**Example:** sine signal of frequency  $f_0 = 2$  kHz is sampled at sampling frequency  $f_s = 1/T = 10$  kHz. Then DFT is calculated for  $L=100$  samples (DFT spectrum  $X_0, X_1, \dots, X_{L-1}$  is obtained).

Which DFT values have the greatest amplitude?

**Solution:** DFT delivers L values at frequencies  $f_k = \frac{k}{TL}$ ,  $k = 0, 1, \dots, L-1$ ,  $1/(TL) = 100$  Hz, so  $f_k = k \cdot 100$  Hz. There will be two DFT values of maximum amplitudes, at frequencies 2 kHz and  $10-2=8$  kHz (mirror image). Thus DFT coefficients  $X_{20}$  and  $X_{80}$  will have maximum values.

# DFT – frequency analysis



$|DFT|$  of  $\sin(2\pi f_0 nT)$ , sampling frequency  $f_s = 1/T = 44100$  Hz,  $L=256$  samples

DFT coefficients:  $X_0, X_1, \dots, X_{255}$ . Maximum value has coefficient  $X_{12}$

Task: Estimate frequency  $f_0$

Solution: DFT coefficient  $X_k$  corresponds to frequency  $f_k = k / (TL) = k f_s / L$   
 $f_{12} = 12 \times 44100 / 256 = 2065$  Hz.

Note that our result is not exact. DFT yields 256 samples of spectrum,  $f_{k+1} - f_k = 1/(TL) = 44100 / 256 = 172$  Hz. Frequency of observed sine signal may be between  $2065 - 172 / 2$  Hz and  $2065 + 172 / 2$  Hz, that is between 1979 and 2151 Hz. (in fact it was 2 kHz)

# Z transform

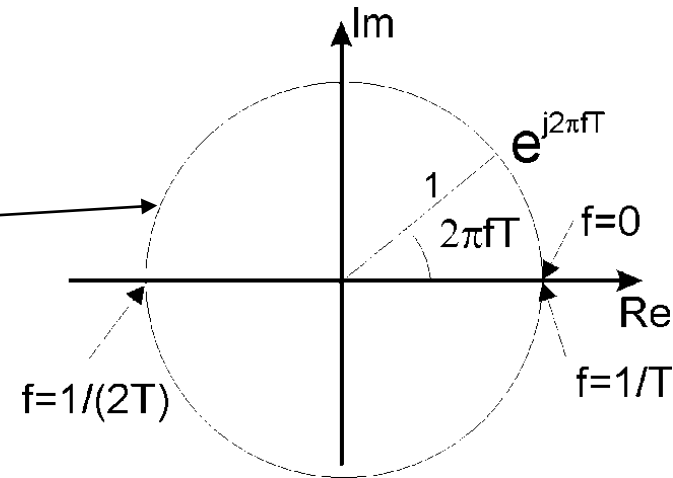
Discrete signal  $x_s(t) = \sum_n x_n \delta(t - nT)$

DTFT  $X_s(f) = \sum_n x_n F[\delta(t - nT)] = \sum_n x_n e^{-j2\pi f n T}$

Substitution  $z = e^{j2\pi f T}$

Z Transform  $X(z) = Z[\{x_n\}] = \sum_n x_n z^{-n}$

Generalization: any value of „z” for which the sum converges



**Properties:** linearity:  $Z[a\{x_n\} + b\{y_n\}] = a Z[\{x_n\}] + b Z[\{y_n\}] = a X(z) + b Y(z)$ ,

shift :  $Z[\{x_{n-k}\}] = z^{-k} X(z)$

convolution  $y_n = \sum_{k=-\infty}^{\infty} x_k h_{n-k} \Rightarrow y_n = x_n * h_n \quad Y(z) = X(z) H(z)$

# Calculation of Z transform

For finite number of samples we use straight method:  $X(z) = \sum_{n=M}^N x_n z^{-n}$

for example  $x_1 = 2, x_2 = -1$ ,  $X(z) = \sum_{n=1}^2 x_n z^{-n} = 2z^{-1} - z^{-2}$

For infinite number of samples we use series theory, for example

$$y_n = a^n 1_n \xrightarrow{|a| < 1} Y(z) = \sum_{n=0}^{\infty} a^n z^{-n} = \sum_{n=0}^{\infty} \left(\frac{a}{z}\right)^n = \frac{1}{1 - \frac{a}{z}} = \frac{z}{z - a}$$

We may use the properties of Z transform, like shift property or convolution property

# Calculation of Inverse Z transform

Inverse Z transform: calculation of time series (samples) knowing  $X(z)$ :  $x_n = Z^{-1}[X(z)]$

If  $X(z)$  is a polynomial  $X(z) = \sum_{n=M}^N x_n z^{-n}$  we use straight method:

for example  $X(z) = 2z - 1 + z^{-2} \rightarrow x_{-1} = 2, x_0 = -1, x_2 = 1$

Often we use known Z transforms, like these:  $Y(z) = \frac{z}{z-a} \rightarrow y_n = a^n 1_n$

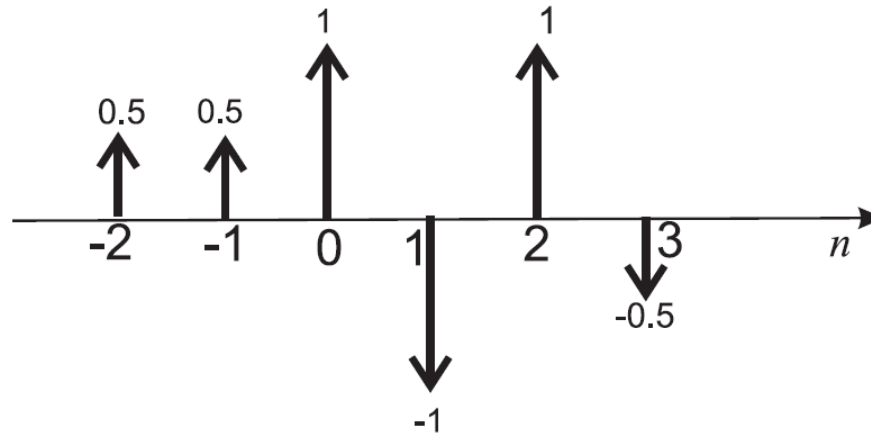
$$Y(z) = \frac{za}{(z-a)^2} \rightarrow y_n = n a^n 1_n$$

We also use properties of Z transform, for example:

$$Z^{-1}\left[\frac{b}{z-a}\right] = b Z^{-1}\left[\frac{1}{z-a}\right] = b Z^{-1}\left[\frac{1}{z} \frac{z}{z-a}\right] = b 1_{n-1} a^{n-1}$$

# Calculation of Straight and Inverse Z transform - examples

For a series of samples  $x_n$  (see figure) calculate Z transform.



Solution

$$X(z) = \sum_{n=-\infty}^{\infty} x_n z^{-n} = 0.5z^2 + 0.5z + 1 - z^{-1} + z^{-2} - 0.5z^{-3}$$

## Calculation of Straight and Inverse Z transform - examples

For a given Z transform:

$$X(z) = \frac{5z}{z + 0.8} + \frac{3}{z - 0.5}$$

calculate the inverse Z transform, that is a series of samples  $x_n$ .

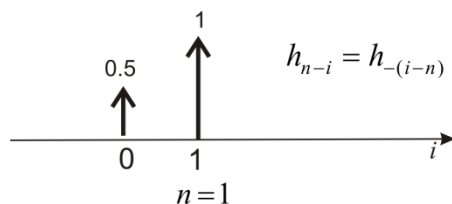
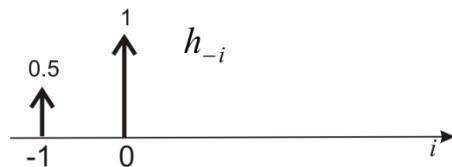
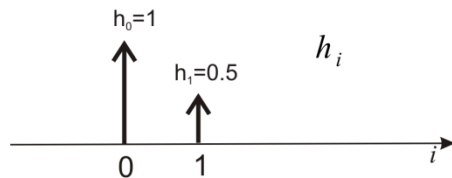
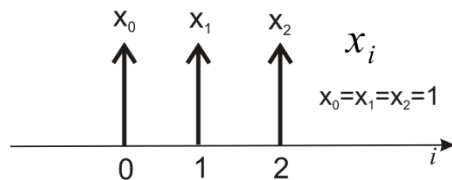
$$X(z) = 5 \frac{z}{z + 0.8} + 3z^{-1} \frac{z}{z - 0.5}$$

$$x_n = 5 \cdot 1_n \cdot (-0.8)^n + 3 \cdot 1_{n-1} \cdot (0.5)^{n-1}$$

# Calculation of convolution

## Calculation in time domain

$$y_n = \sum_{i=-\infty}^{\infty} x_i h_{n-i} \quad h_{n-i} = h_{-(i-n)}$$



$$\begin{aligned} n=0, y_0 &= 1 \times 1 = 1 \\ n=1, y_1 &= 1 \times 0.5 + 1 \times 1 = 1.5 \\ n=2, y_2 &= 1 \times 0.5 + 1 \times 1 = 1.5 \\ n=3, y_3 &= 1 \times 0.5 = 0.5 \end{aligned}$$

## Calculation in transform domain

$$Y(z) = X(z) \cdot H(z)$$

$$X(z) = 1 + z^{-1} + z^{-2}$$

$$H(z) = 1 + \frac{1}{2} z^{-1}$$

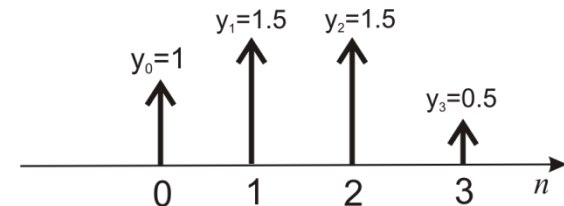
$$Y(z) = X(z)H(z) =$$

$$= 1 + \frac{1}{2} z^{-1} + z^{-1} + \frac{1}{2} z^{-2} + z^{-2} + \frac{1}{2} z^{-3} =$$

$$= 1 + \frac{3}{2} z^{-1} + \frac{3}{2} z^{-2} + \frac{1}{2} z^{-3} =$$

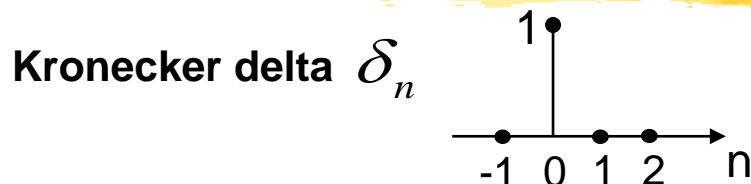
$$= y_0 + y_1 z^{-1} + y_2 z^{-2} + y_3 z^{-3}$$

Both methods give the same result





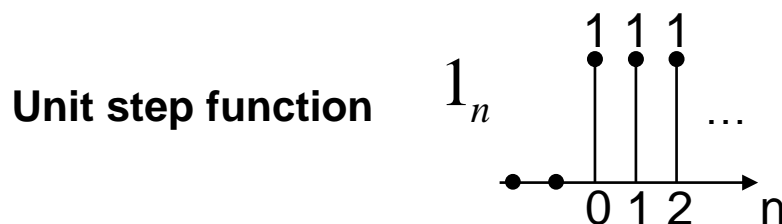
# Some signals and their Z transforms



$$Z[\delta_n] = \sum_n \delta_n z^{-n} = 1$$

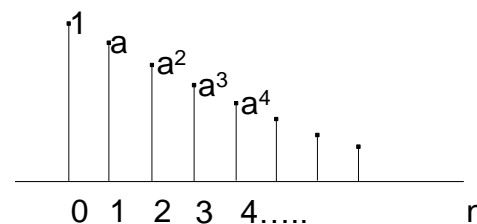
$$Z[\delta_{n-m}] = \sum_n \delta_{n-m} z^{-n} = z^{-m}$$

**Convolution with Kronecker delta**  $y_n = x_n * \delta_{n-m} = \sum_{k=-\infty}^{\infty} x_k \delta_{n-m-k} = x_{n-m}$



$$Z[1_n] = \sum_{n=0}^{\infty} z^{-n} = \sum_{n=0}^{\infty} \left(\frac{1}{z}\right)^n = \frac{1}{1 - \frac{1}{z}} = \frac{z}{z-1}$$

**Exponential function**  $x_n = 1_n a^n$ ,  $X(z) = \frac{z}{z-a}$



$a$  - pole of the rational function  $X(z)$

Important observation: If  $|a| < 1$  (pole within unit circle), then  $x_n \rightarrow 0$   
 If  $|a| > 1$  (pole outside unit circle), then  $x_n \rightarrow \text{infinity}$