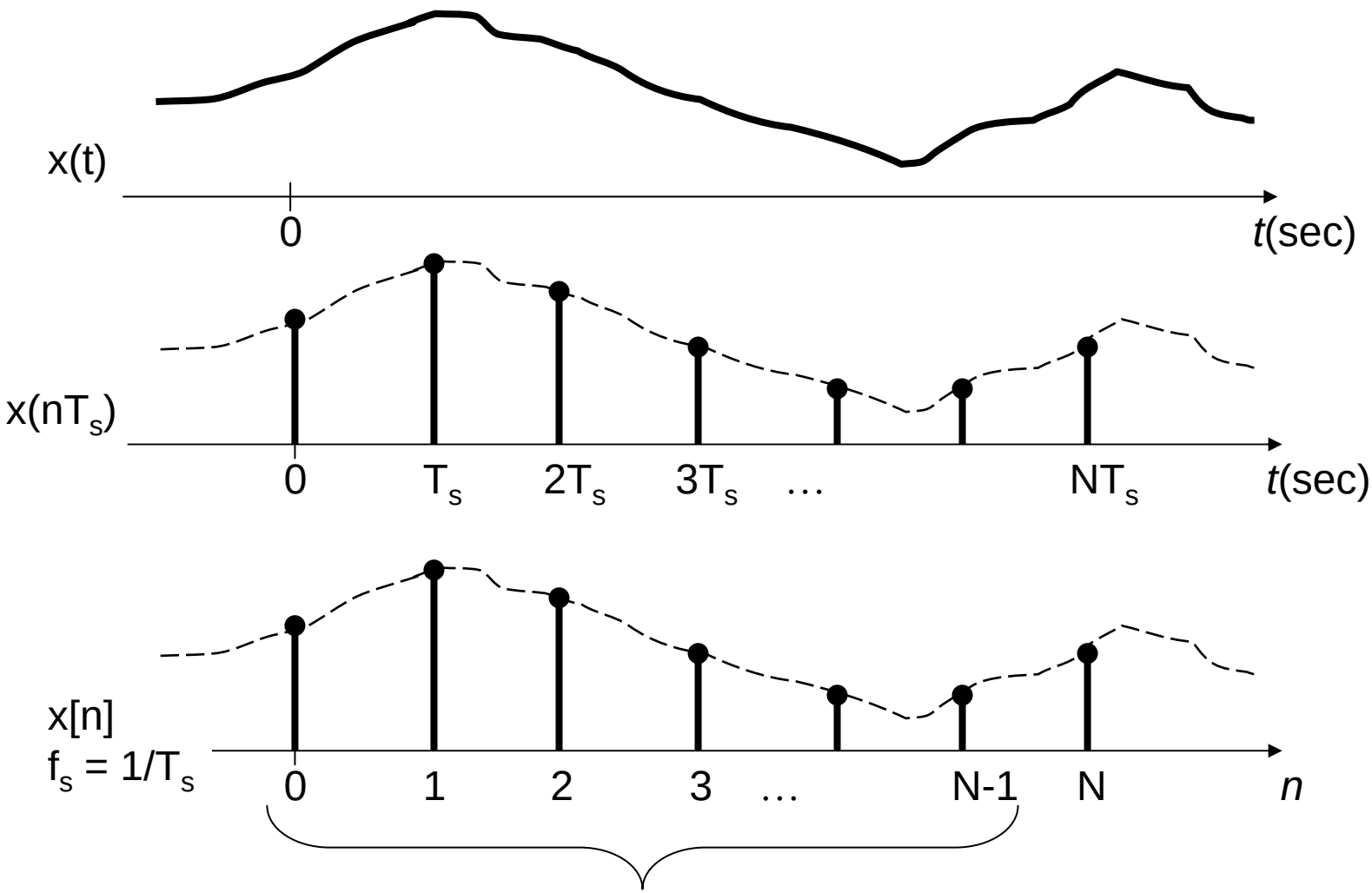
-To approximate the frequency content w/the DFT (FFT), first sample the analog signal:



Choose N samples of x(t), starting w/n=0 --
Compute DFT(FFT) using these samples

-The DFT(FFT) breaks the signal x[n] into its frequency components (in terms of discrete complex exponentials):

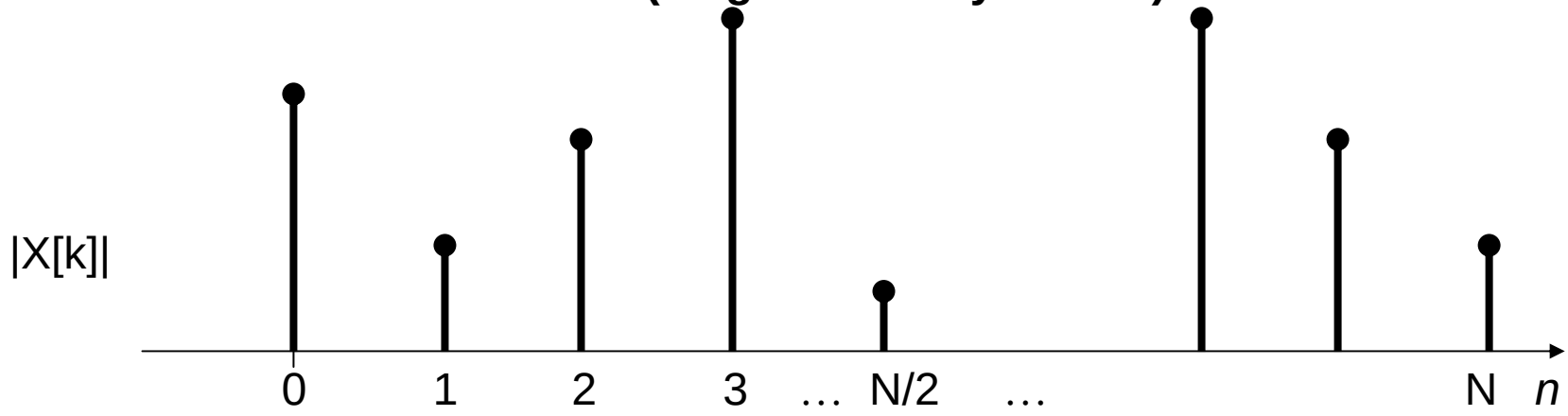
$$x[n] = \frac{1}{N}\sum_{k=0}^{N-1} X[k] e^{j2\pi\left(\frac{nk}{N}\right)}$$

-The amplitude of each frequency component is computed by:

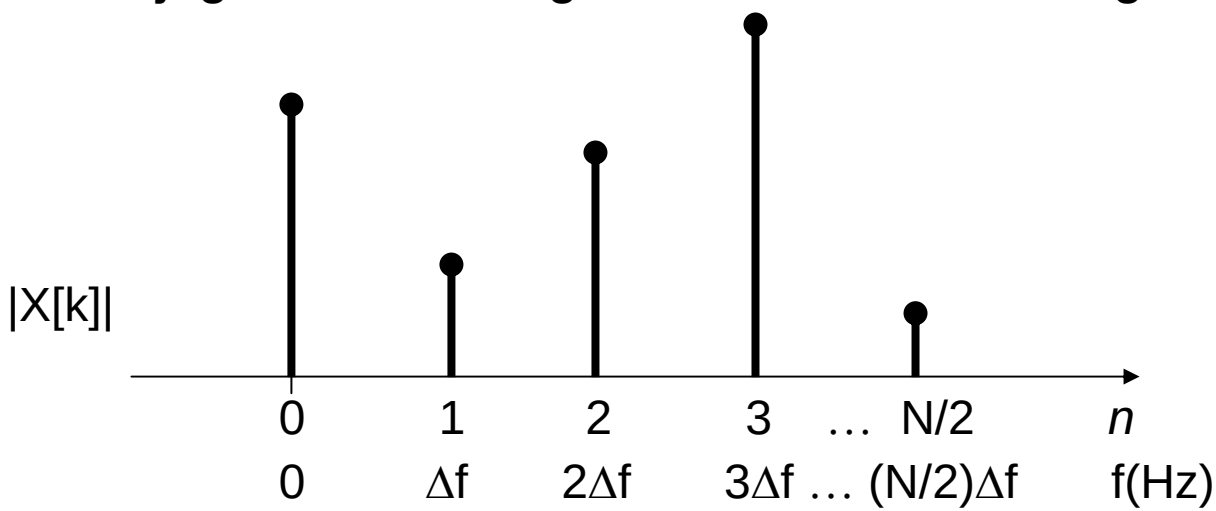
$$X[k] = \sum_{n=0}^{N-1} x[n] e^{-j2\pi\left(\frac{nk}{N}\right)}$$

-If there are N samples of the time signal, then there are N values of X[k] (which is the DFT).

-The N values of the DFT (magnitude only shown):



-Only need the 1st half of the DFT values (2nd half is a complex conjugate mirror image for real-valued time signals) :



-Frequency resolution of the frequency plot: $\Delta f = f_s/N$, so that the highest frequency on the plot is $f_s/2$.

-This agrees with Nyquist theory—if we sample at f_s samples/sec, the highest frequency content of the analog signal is assumed to be $f_s/2$ Hz.