

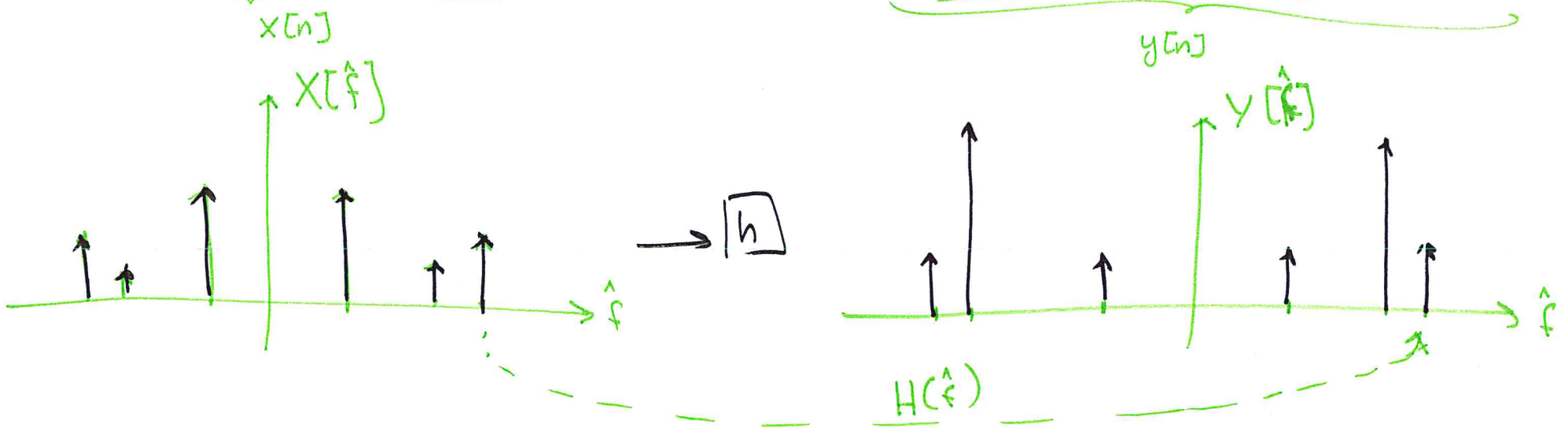
Lecture 19: Spectrum of Filter Output

Filter frequency response

$$e^{j2\pi \hat{f} n} \longrightarrow \boxed{h} \longrightarrow \underline{H(\hat{f})} \cdot e^{j2\pi \hat{f} n}$$

If input is sum of sinusoids

$$\underbrace{a_1 e^{j2\pi \hat{f}_1 n} + a_2 e^{j2\pi \hat{f}_2 n} + \dots}_{x[n]} \longrightarrow \boxed{h} \longrightarrow \underbrace{a_1 H(\hat{f}_1) e^{j2\pi \hat{f}_1 n} + a_2 H(\hat{f}_2) e^{j2\pi \hat{f}_2 n} + \dots}_{y[n]}$$



Spectrum of output @ frequency \hat{f} :

$$\boxed{Y(\hat{f}) = X(\hat{f}) \cdot H(\hat{f})}$$

Two way to calculate filter output :

- 1) convolution in time domain: $y[n] = x[n] * h[n]$
- 2) multiplication in frequency domain



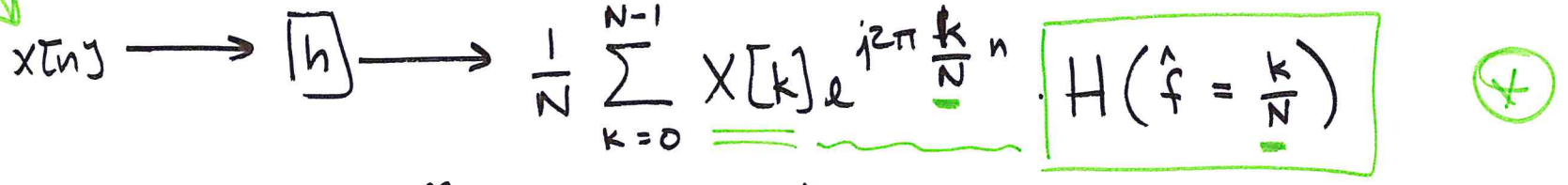
Spectrum of $(x[n] * h[n]) = X(\hat{f}) \cdot H(\hat{f})$

convolution in time domain \iff multiplication in frequency domain

DFT expansion of a signal with N samples:

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

\Rightarrow expressing signal as a sum of sinusoids w/ frequencies $0, \pm 1/N, \pm 2/N, \dots, 1/2$



Recall $H(\hat{f}) = \sum_{l=-\infty}^{\infty} h[l] e^{-j2\pi l \hat{f}} =$ frequency response

$H[k] = \sum_{l=0}^{N-1} h[l] e^{-j2\pi \frac{k}{N} l} =$ DFT of filter impulse response

$$H(\hat{f} = \frac{k}{N}) = H[k]$$

$$= \sum_{l=-\infty}^{\infty} h[l] e^{-j2\pi l \frac{k}{N}} = \sum_{l=0}^{N-1} h[l] e^{-j2\pi l \frac{k}{N}}$$

if $h[l]=0$ for $l < 0$ and $l \geq N$

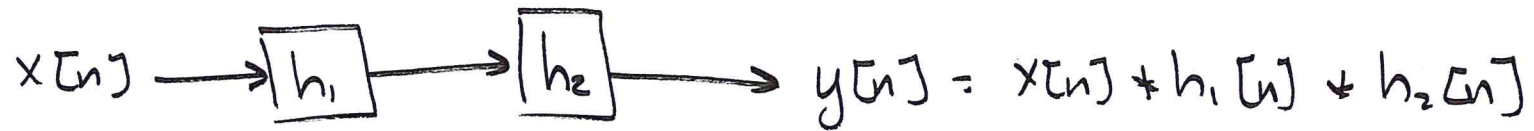
$$= H[k] = \text{DFT of } h$$

back to \otimes

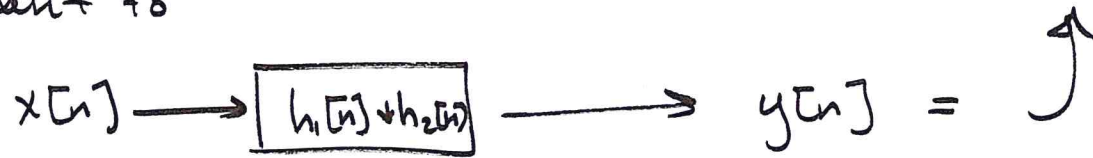
$$x[n] = \frac{1}{N} \sum_{k=0}^{N-1} X[k] e^{j2\pi kn/N} \rightarrow \boxed{h} \rightarrow y[n] = \frac{1}{N} \sum_{k=0}^{N-1} X[k] H[k] e^{j2\pi kn/N}$$

multiplication in
frequency domain

Filter cascade:



equivalent to



Using the spectrum of filter outputs:

$$[X(\hat{f}) \cdot H_1(\hat{f})] \cdot H_2(\hat{f}) = X(\hat{f}) \cdot [H_1(\hat{f}) \cdot H_2(\hat{f})]$$

To compute filter output:

1. DFT of $x[n] \longrightarrow X[k]$
2. DFT of $h[n] \longrightarrow H[k]$ (use zero padding)
3. Multiply $X[k]H[k] \longrightarrow Y[k]$ ↳ $h[n]$ and add zeros onto end to make it length-N
4. IDFT of $Y[k] \longrightarrow y[n]$

in matlab:

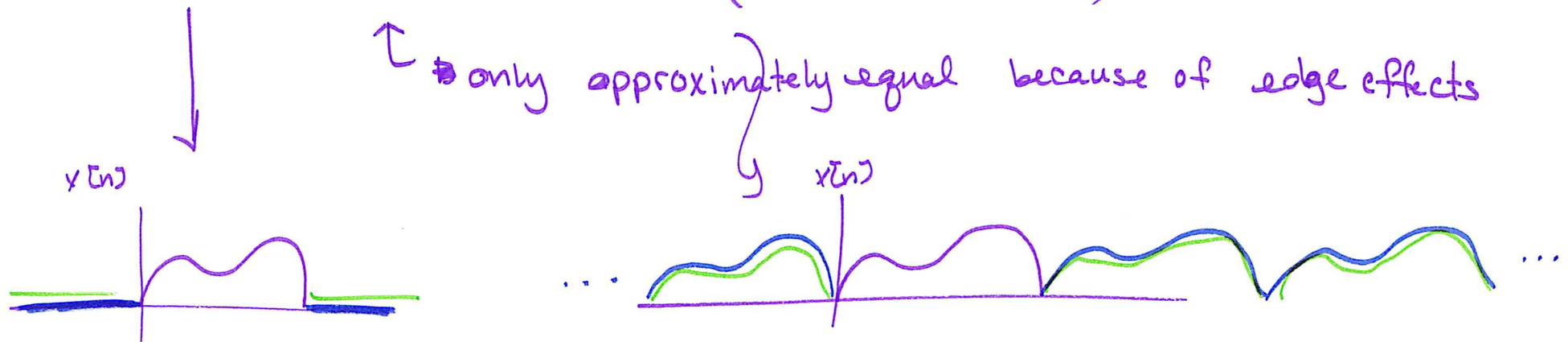
$N = \text{length}(x);$

$M = \text{length}(h);$

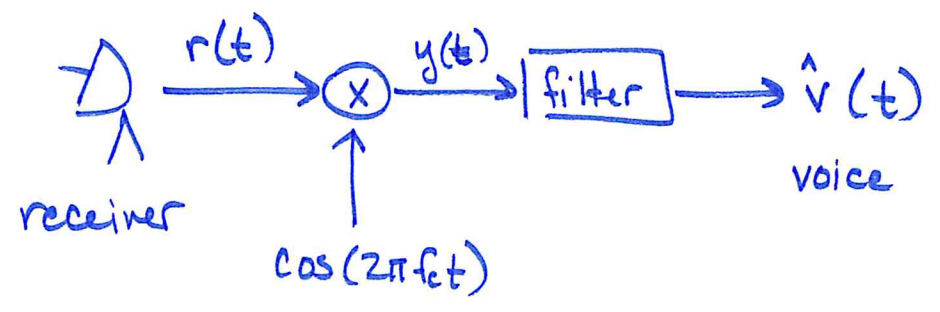
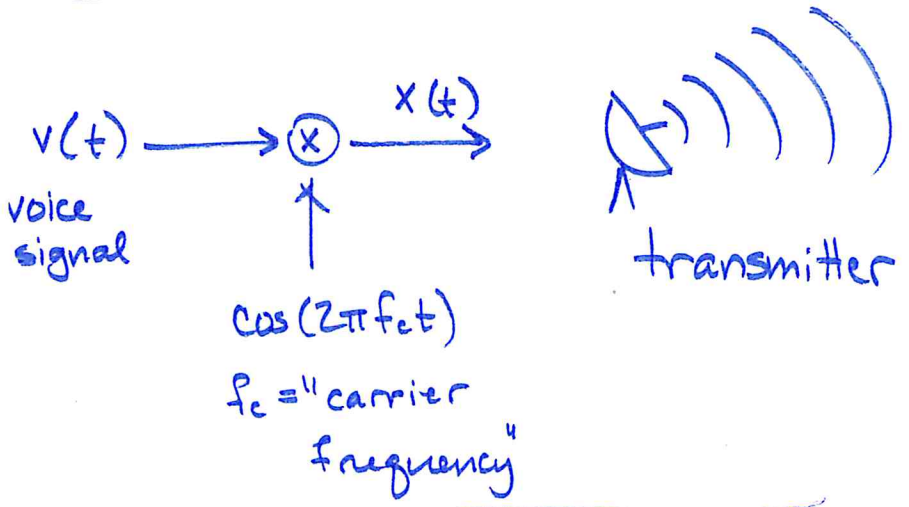
$h = [h, \text{zeros}(1, N-M)];$

$\text{conv}(x, h) \approx \text{ifft}(\text{fft}(x) .* \text{fft}(h))$

↑ only approximately equal because of edge effects



AM Communications



modulation: shifting spectrum of signal so it can be transmitted more easily

demodulation - extract $v(t)$ from modulated signal.

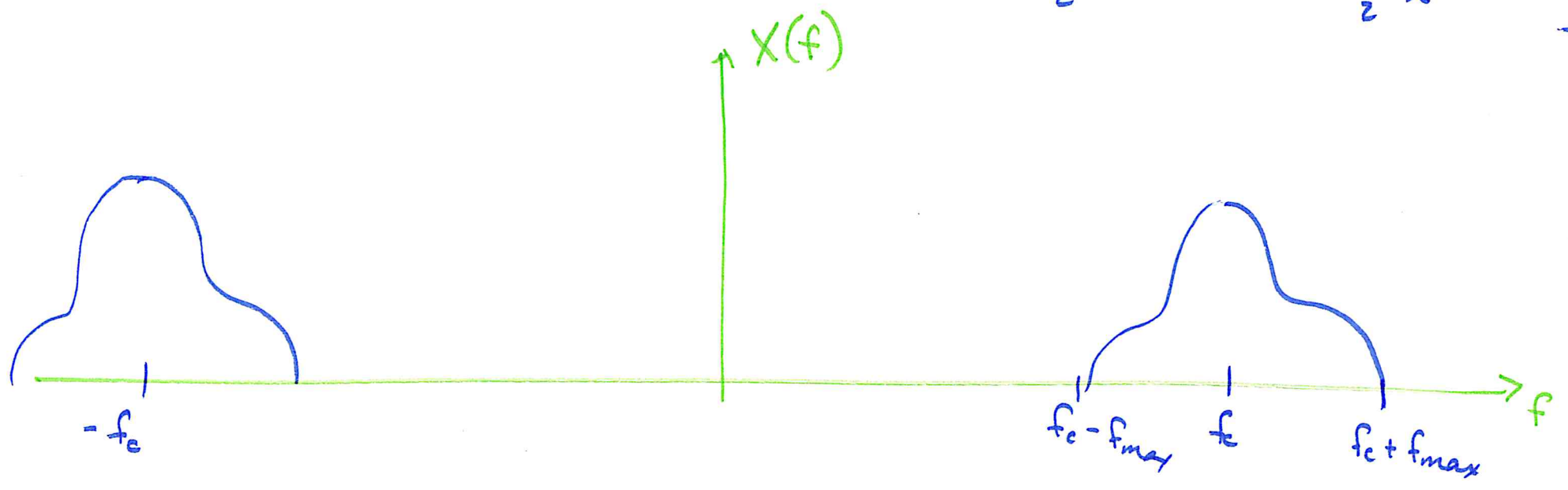
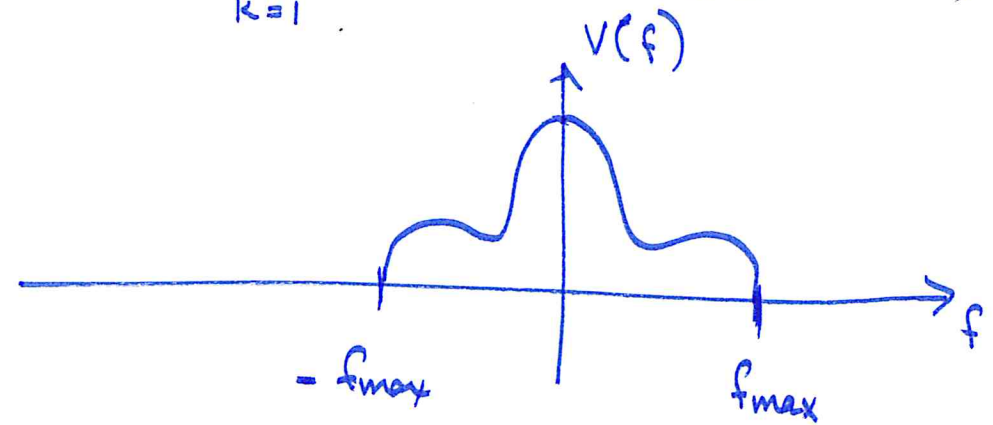
Modulation:

$$x(t) = v(t) \cdot \cos(2\pi f_c t)$$

$$\text{where } v(t) = \sum_{k=1}^N (a_k e^{j2\pi f_k t} + a_k^* e^{-j2\pi f_k t})$$

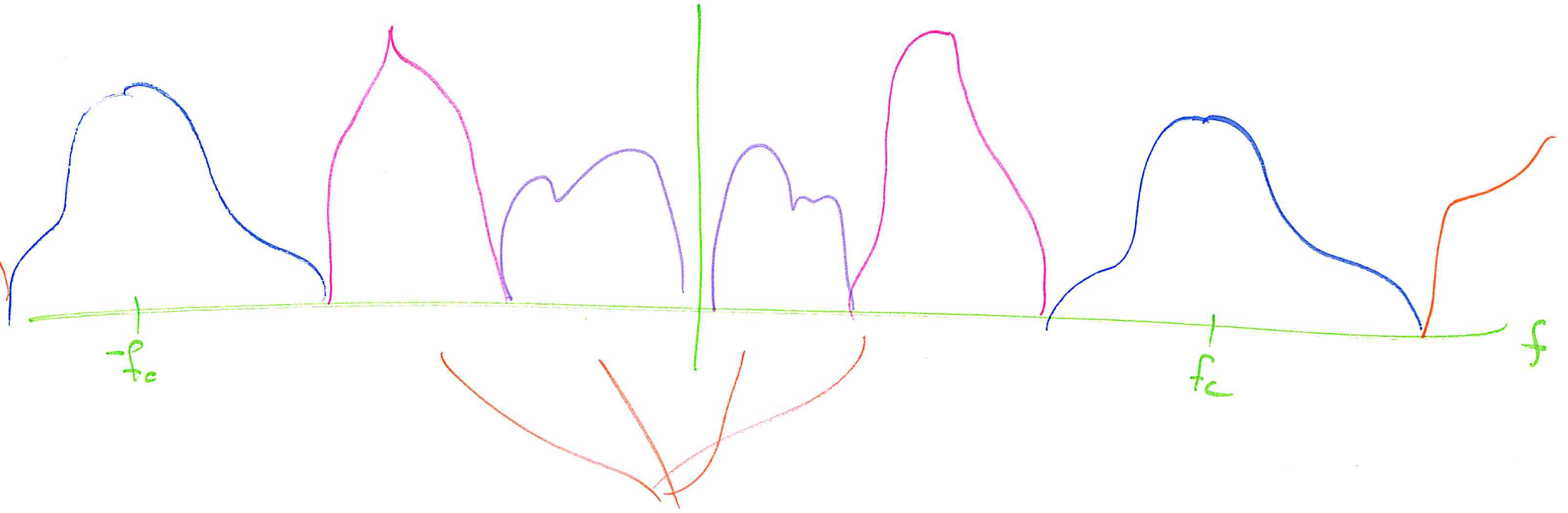
$$= \sum_{k=1}^N (a_k e^{j2\pi f_k t} + a_k^* e^{-j2\pi f_k t}) \cdot \left(\frac{1}{2} e^{j2\pi f_c t} + \frac{1}{2} e^{-j2\pi f_c t} \right)$$

$$= \sum_{k=1}^N \left[\frac{a_k}{2} e^{j2\pi(f_c + f_k)t} + \frac{a_k^*}{2} e^{j2\pi(f_c - f_k)t} + \frac{a_k^*}{2} e^{-j2\pi(f_c - f_k)t} + \frac{a_k}{2} e^{-j2\pi(f_c + f_k)t} \right]$$



$r(t)$ = received signal = transmitted signal $x(t)$ + signals from other stations

$R(f)$



other stations


$$r(t) = x(t) + \text{o.s. (other stations)}$$

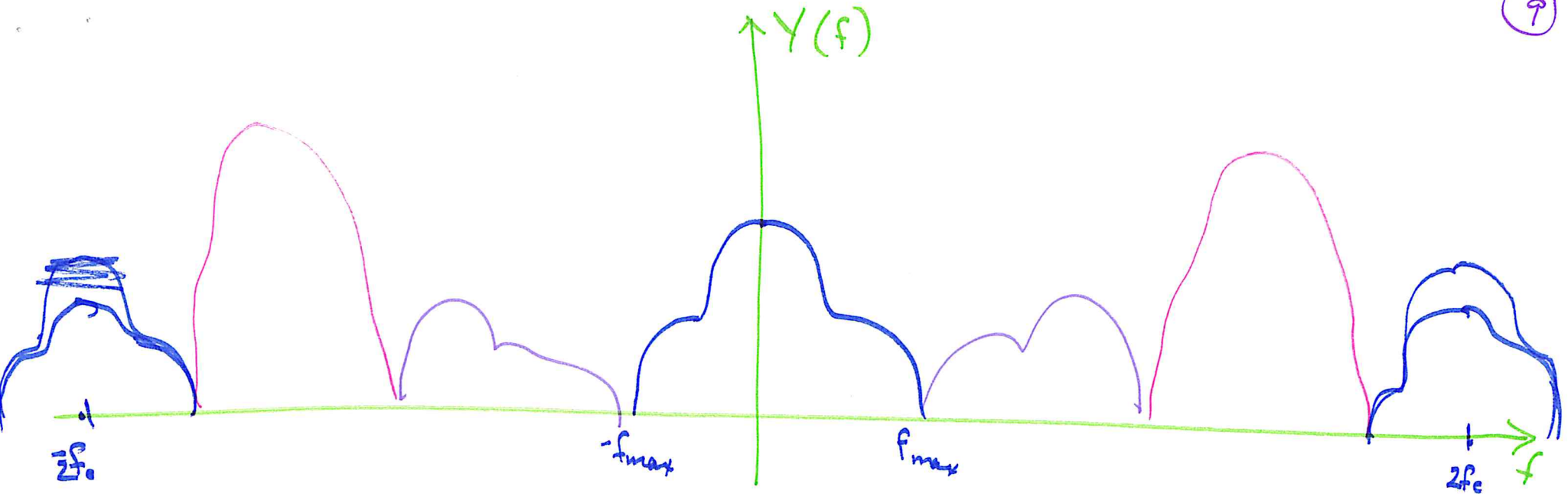
How to get $v(t)$?

- 1. multiply $r(t)$ by $\cos(2\pi f_c t)$
- $$y(t) = r(t) \cos(2\pi f_c t)$$
- $$= (x(t) + \text{o.s.}) \cos(2\pi f_c t)$$

$$= (v(t) \cos(2\pi f_c t) + \text{o.s.}) \cos(2\pi f_c t)$$

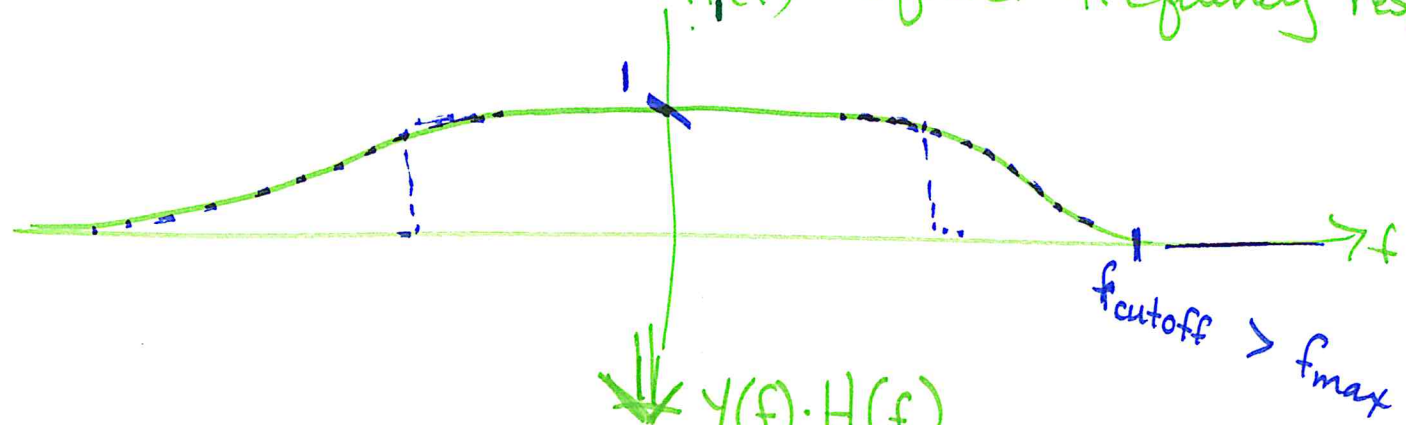
$$y(t) = \frac{1}{2} v(t) + \frac{1}{2} v(t) \cos(4\pi f_c t) + \text{o.s.} \cdot \cos(2\pi f_c t)$$

$$\begin{aligned} & \cos(2\pi f_c t)^2 \\ &= \left(\frac{1}{2} e^{j2\pi f_c t} + \frac{1}{2} e^{-j2\pi f_c t} \right)^2 \\ &= \frac{1}{4} e^{j4\pi f_c t} + \frac{1}{4} e^{-j4\pi f_c t} + \frac{1}{2} \\ &= \frac{1}{2} \cos(4\pi f_c t) + \frac{1}{2} \end{aligned}$$


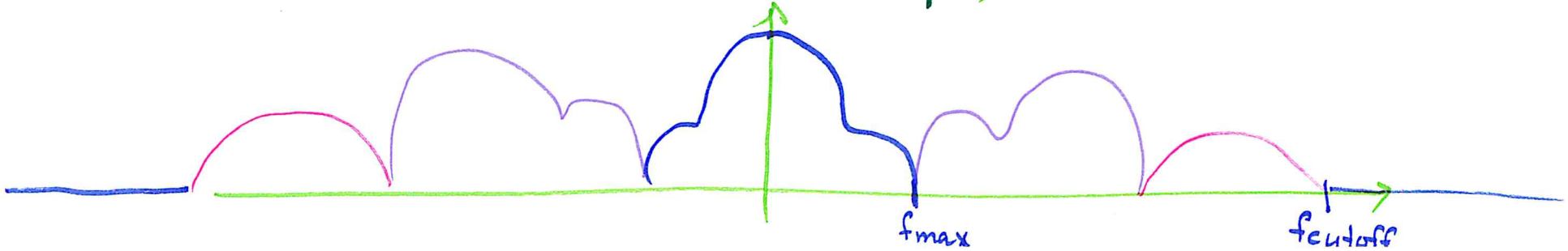


2. Apply a low pass analog filter

$H_1(f)$ = filter frequency response



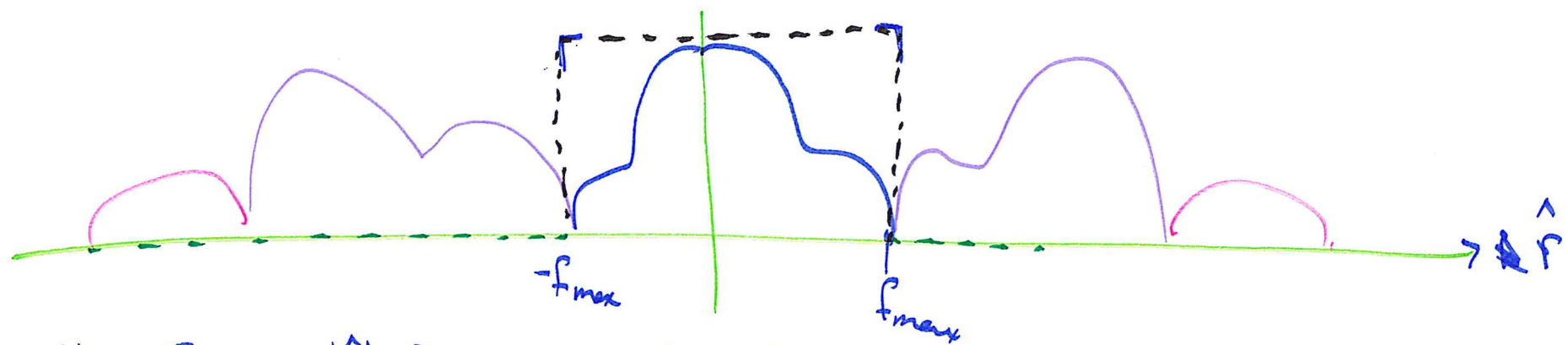
$Y(f) \cdot H_1(f)$



3. Sample with an A/D converter @ frequency $f_s \geq 2 f_{\text{cutoff}}$

4. Digital filter

$$Y[\hat{k}] H[\hat{k}]$$



$$H_2(\hat{f}) = \begin{cases} 1 & \text{if } |\hat{f}| < \frac{f_{\text{max}}}{f_s} \\ 0 & \text{otherwise} \end{cases}$$

$(Y[\hat{f}] H_2[\hat{f}]) H_2[\hat{f}]$ → only "passes" frequencies below f_{max}

