
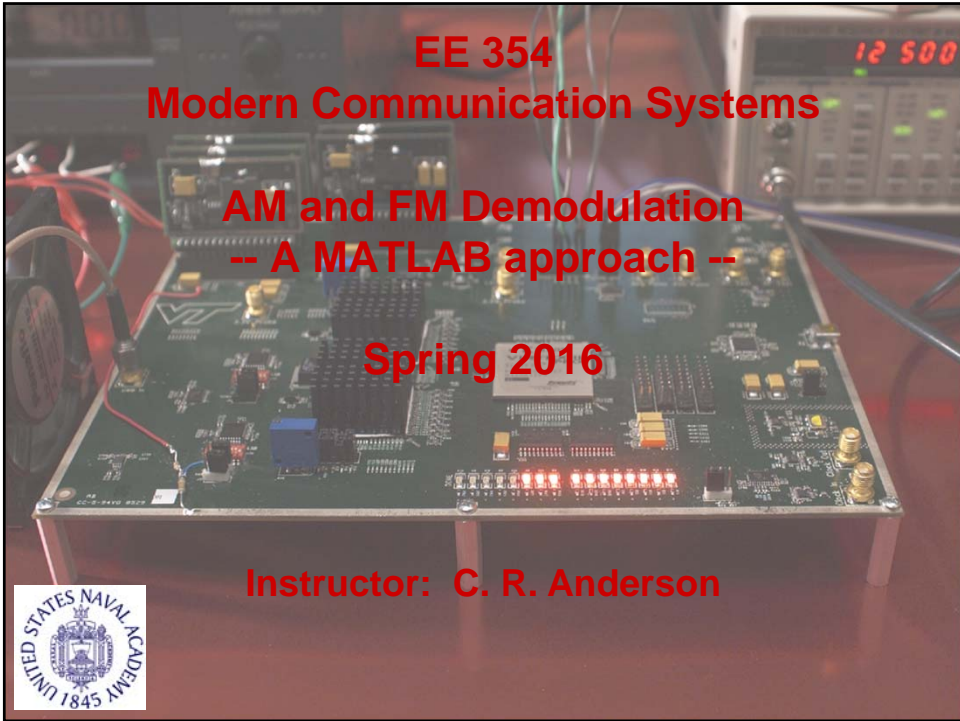


**EE 354**  
**Modern Communication Systems**

**AM and FM Demodulation**  
**-- A MATLAB approach --**

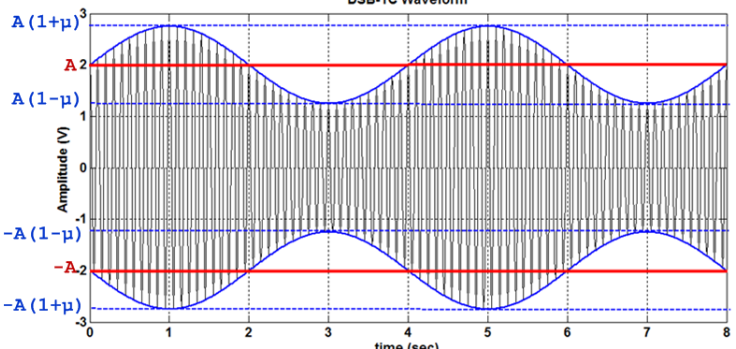
**Spring 2016**

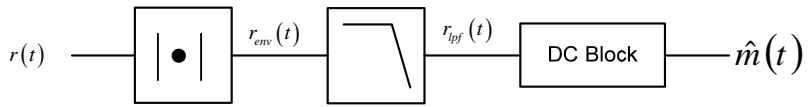
**Instructor: C. R. Anderson**

### Recovering the Message: Envelope Detector

DSB-TC Waveform





**Observe:** If we can trace out the Envelope of the AM signal, we can effectively recover the underlying information signal.

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## Envelope Detector Mathematically

**Received Signal:**  $r(t) = A_c [1 + a_{\text{mod}} m_n(t)] \cos(2\pi f_c t)$

**Absolute Value Operation:**  $r_{\text{env}}(t) = A_c \left| [1 + a_{\text{mod}} m_n(t)] \cos(2\pi f_c t) \right|$

If  $1 + a_{\text{mod}} m_n(t)$  is constrained to be always positive (i.e.,  $0 \leq a_{\text{mod}} \leq 1$ )

$$r_{\text{env}}(t) = A_c \left[ 1 + a_{\text{mod}} m_n(t) \right] \left| \cos(2\pi f_c t) \right|$$

**Note:** The full-wave rectified cosine can be expanded in a Fourier Series

$$r_{\text{env}}(t) = A_c [1 + a_{\text{mod}} m_n(t)] [a_0 + a_1 \cos(2\pi(2f_c)t) + a_2 \cos(2\pi(4f_c)t) + \dots]$$

**LPF:** Block everything except the  $a_0$  term.

$$r_{\text{env}}(t) = a_0 A_c (1 + a_{\text{mod}} m_n(t))$$

**DC Block does the rest:**  $\hat{m}(t) = a_0 a_{\text{mod}} m_n(t)$

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## AM Demodulation in Software

```
fs_rf = 1e6; % sampling frequency of the AM signal
fco = 0.02e6; % cutoff frequency for the low pass filter
% Will need to downsample the recovered signal to an audio-
% level sampling frequency for output to sound card.
downsample_rate = floor(fs_rf./44.1e3);

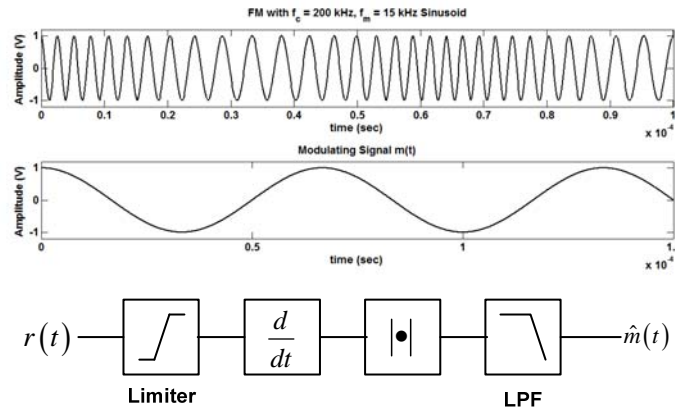
% Perform AM Demod using Envelope Detection.
r_env = abs(am_samples); % Envelope Detector
r_lpf = filter_audio(x_env,fco,fs_rf); % Low Pass Filter
% Downsample to audio sampling frequency
fs_audio = fs_rf./downsample_rate;
m_hat = downsample(r_lpf, downsample_rate);

% Perform AM Demod using Square-Law Detector
r_sq = am_samples.^2; % Square-Law Detector
% Low Pass Filter and Square Root operation
m_hat = sqrt(filter_audio(r_sq,fco,fs_rf));
% Downsample to audio sampling frequency
fs_audio = fs_rf./downsample_rate;
x_bb = downsample(m_hat, downsample_rate);

% Output the demodulated signal to the sound card
sound(m_hat, fs_audio, 16);
```

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## FM Demodulation: Implement the Instantaneous Frequency Equation



$$r(t) = A_c \cos\left(2\pi f_c t + 2\pi k_f \int_0^t m(\tau) d\tau\right) \quad f_i = \frac{1}{2\pi} \frac{d\theta(t)}{dt} \quad \hat{m}(t) = 2\pi A_c k_f m(t)$$

Note that there's another way to do this...

## FM Demodulation in Software

```

fs_rf = 1e6; % sampling frequency of the FM signal
fco = 0.02e6; % cutoff frequency for the low pass filter
% Will need to downsample the recovered signal to an audio-
% level sampling frequency for output to sound card.
downsample_rate = floor(fs_rf./44.1e3);

% Perform FM Demod using Instantaneous Frequency Method.
% Note: A hard limiter would go here to reduce noise

diff_sfm = diff(sfm); % Take Derivative of FM signal

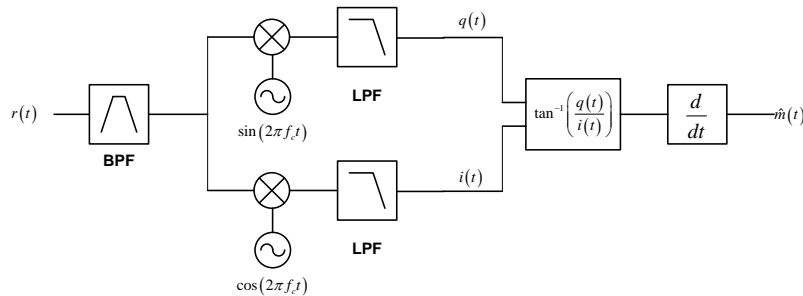
% Matlab 'diff' reduces length by 1 sample, need to add a
% sample back in its place.
diff_sfm = [0 diff_sfm];
sfm_abs = abs(diff_sfm); % Envelope Detector
m_demod = filter_audio(sfm_abs,fco,fs_rf); % Low Pass Filter

% Downsample to audio sampling frequency
fs_audio = fs_rf./downsample_rate;
x_bb = downsample(x_demod, downsample_rate);

% Output the demodulated signal to the sound card
sound(x_bb, fs_audio, 16);

```

## FM Arc tangent Demodulator



$$i(t) = \frac{1}{2} A \cos(\theta(t)) \quad \hat{m}(t) = \tan^{-1} \left( \frac{q(t)}{i(t)} \right) = \tan^{-1} \left( \frac{\frac{1}{2} A \sin(\theta(t))}{\frac{1}{2} A \cos(\theta(t))} \right) = \tan^{-1}(\tan(\theta(t))) = \theta(t)$$

$$q(t) = \frac{1}{2} A \sin(\theta(t))$$

Note:  $\theta(t) = 2\pi k_f \int_0^t m(\tau) d\tau$

**Need a 4-Quadrant ArcTan or Phase Unwrap.**

**“Nobody does this---I’m shocked!” --- fred harris**

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## FM ArcTan Demodulation in Software

```
fs_rf = 1e6; % sampling frequency of the FM signal
fco = 0.02e6; % cutoff frequency for the low pass filter
% Will need to downsample the recovered signal to an audio-
% level sampling frequency for output to sound card.
downsample_rate = floor(fs_rf./44.1e3);

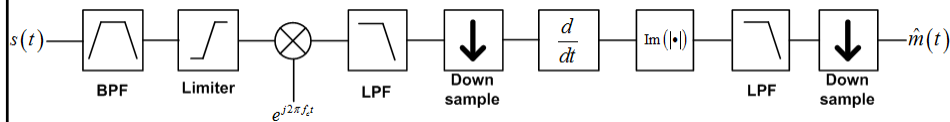
% Perform FM Demod using ArcTangent Method.
% Note: A hard limiter would go here to reduce noise
% Create the Inphase and Quadrature Signals
q_sig = sfm.*sin(2*pi*fco*t);
i_sig = sfm.*cos(2*pi*fco*t);

% Perform 4-Quadrant ArcTangent
arctan_sfm = unwrap(atan2(q_sig, i_sig));

% Matlab 'arctan2' and 'diff' reduces length by 1 sample, need to add a
% sample back in its place.
arctan_sfm = [0 arctan_sfm];
diff_sfm = diff(sfm); % Take Derivative of ArcTangent signal
diff_sfm = [0 diff_sfm];
m_demod = filter_audio(sfm_abs, fco, fs_rf); % Low Pass Filter
```

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## FM Complex Exponential Demodulator



```

fs_rf = 1e6; % sampling frequency of the FM signal
fco = 0.02e6; % cutoff frequency for the low pass filter
% Will need to downsample the recovered signal to an audio-
% level sampling frequency for output to sound card.

```

```

fs_soundcard = 44.1e3;
fc = XXXXXX; % Center frequency of the FM Signal
% Decimate to sampling rate ~ just greater than the FM BW
dec_rate_rf = floor(fs_rf./(fs_soundcard.*XXXX));
fs_bb = fs_rf./dec_rate_rf;
dec_rate_bb = ceil(fs_bb./fs_soundcard);
fs_audio = fs_rf./(dec_rate_rf.*dec_rate_bb);

```

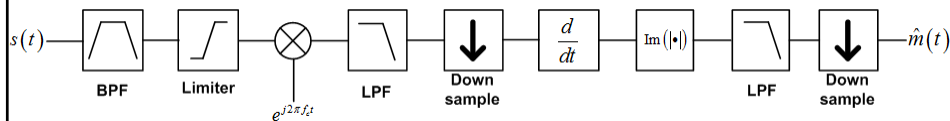
```

% Bandpass Filter
x_fm = filter_IF(sfm,flow,fhigh,fs_rf);

```

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## FM Complex Exponential Demodulator



```

% Downconvert to baseband and filter
x_bb = filter_bb(x_fm.*exp(-j.*2.*pi.*fc.*time),frf_co,fs_rf);
x_bb_ds = downsample(x_bb,dec_rate_rf); % Downsample, first stage
x_limit = x_bb_ds./abs(x_bb_ds); % Limiter (could move earlier)

```

```

% Generates an "Ideal" differentiator using an FIR Filter
hd = firls(30,[0 100000 0.8.*fs_bb fs_bb]/fs_bb, [0 1 0 0],'differentiator');
x_diff = imag(conv(x_limit,hd,'same').*conj(x_limit));

```

```

% Now, just extract the audio signal
x_demod = filter_bb(x_diff,fco,fs_bb);
t_bb = downsample(time, dec_rate_rf*dec_rate_bb);
x_bb = downsample(x_demod, dec_rate_bb);

```

```

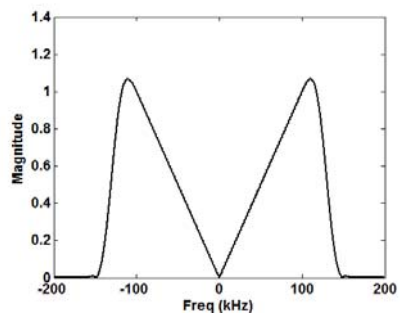
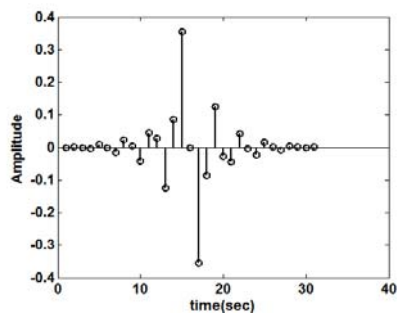
soundsc(x_bb,fs_audio,16)

```

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## Note: FIR Filter as a Differentiator

```
hd = firls(30,[0 100000 0.8.*fs_bb fs_bb]/fs_bb, [0 1 0 0],'differentiator');  
% Note: This is a digital version of a CT Differentiator  
%  $F[d/dt] = j*2*\pi*fc*t \rightarrow$  Linear frequency response upto the cutoff freq.  
  
% Plot the filter, and its frequency response  
  
subplot(121);  
stem(hd);  
Subplot(122);  
f = [-fs_bb./10:fs_bb./10-1];  
plot(f,abs(fftshift(fft([hd zeros(1,2.*(fs_bb./10)-31)]))));
```



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