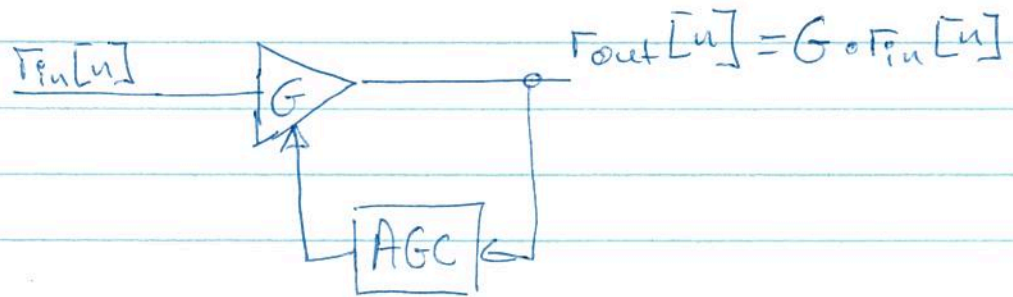


P.1:

Power can be measured by computing:

$$P_{meas}[n] = \frac{1}{N_{avg}} \sum_{k=n}^{n-N_{avg}+1} (r_{out}[k])^2$$

Want $P_{meas}[n]$ to approach P_{des} by adjusting G .

\Rightarrow Objective Function:

$$\begin{aligned} J(G;n) &= (P_{des} - P_{meas}[n])^2 \\ &= \left(P_{des} - \frac{1}{N_{avg}} \sum_{k=n}^{n-N_{avg}+1} (r_{out}[k])^2 \right)^2 \\ &= \left(P_{des} - \frac{1}{N_{avg}} \sum_{k=n}^{n-N_{avg}+1} (G[A] \cdot r_{in}[k])^2 \right)^2 \\ &= \left(P_{des} - \frac{G[n]^2}{N_{avg}} \sum_{k=1}^{n-N_{avg}+1} (r_{in}[k])^2 \right)^2 \end{aligned}$$

Iterative update:

$$G[n+1] = G[n] - \mu \cdot \left. \frac{\partial J(G;n)}{\partial G} \right|_{G=G[n]}$$

Need:

$$\left. \frac{\partial J(G; n)}{\partial G} \right|_{G=G[n]} = 2 \cdot \left(P_{des} - \frac{G[n]}{N_{avg}} \cdot \sum_{k=n}^{n-N_{avg}+1} (r_{in}[k])^2 \right) \cdot \underbrace{\sum_{k=n}^{n-N_{avg}+1} (r_{in}[k])^2}_{= P_{meas}[n]}$$

$$\left(-2 \cdot \frac{G[n]}{N_{avg}} \cdot \sum_{k=n}^{n-N_{avg}+1} (r_{in}[k])^2 \right) \cdot \underbrace{\sum_{k=n}^{n-N_{avg}+1} (r_{in}[k])^2}_{P_{meas}[n]/G[n]}$$

$$= -4 \cdot (P_{des} - P_{meas}[n]) \cdot \frac{P_{meas}[n]}{G[n]}$$

⇒ Update:

$$G[n+1] = G[n] + 4\mu \cdot (P_{des} - P_{meas}[n]) \cdot \frac{P_{meas}[n]}{G}$$

Note: this formula ensures that if $P_{meas} < P_{des}$ then gain G is increased and vice versa.

See AGC.m (in http://spec.gmu.edu/~ypparis/classes/resources_460)

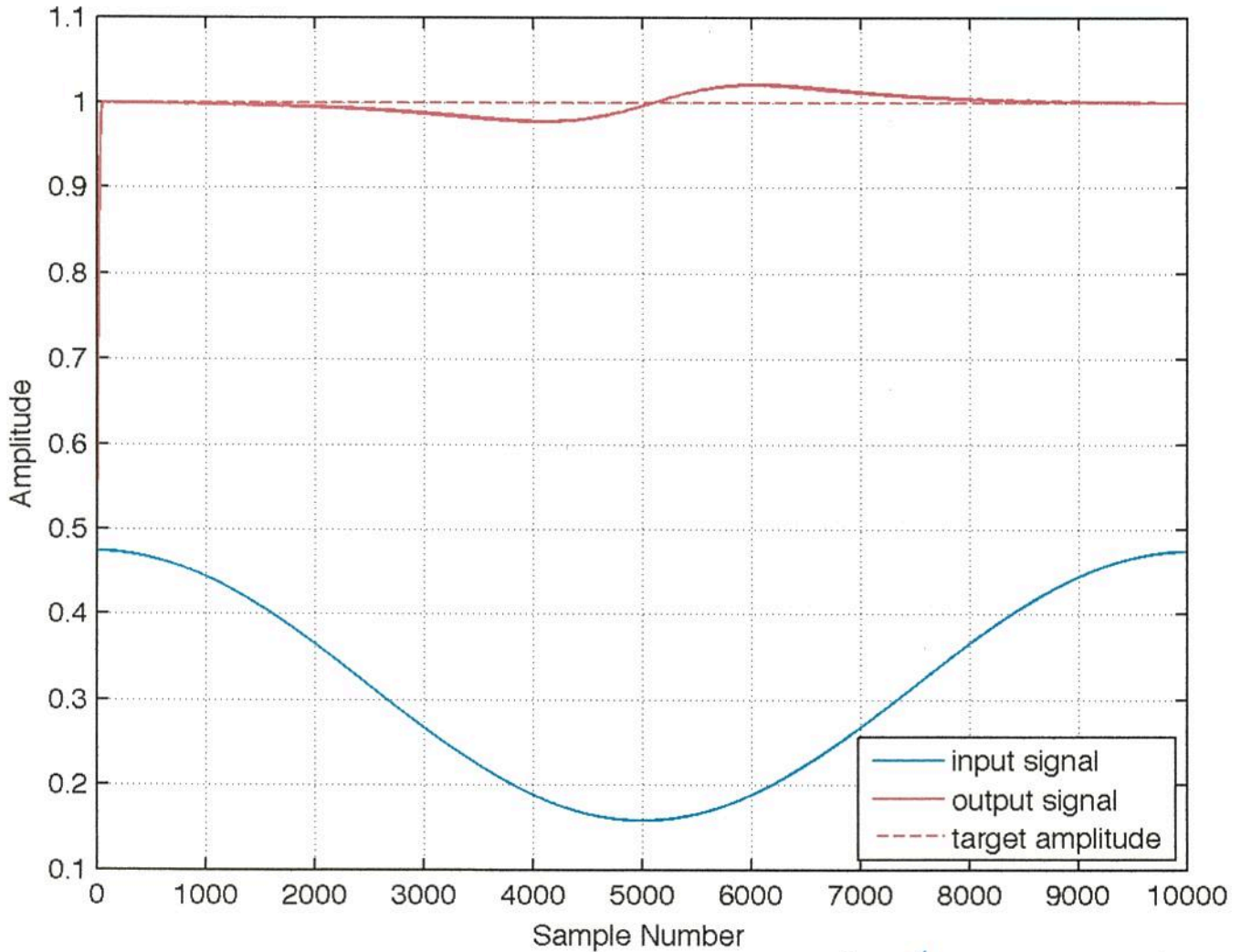
for MATLAB Demo.

See attached plots.

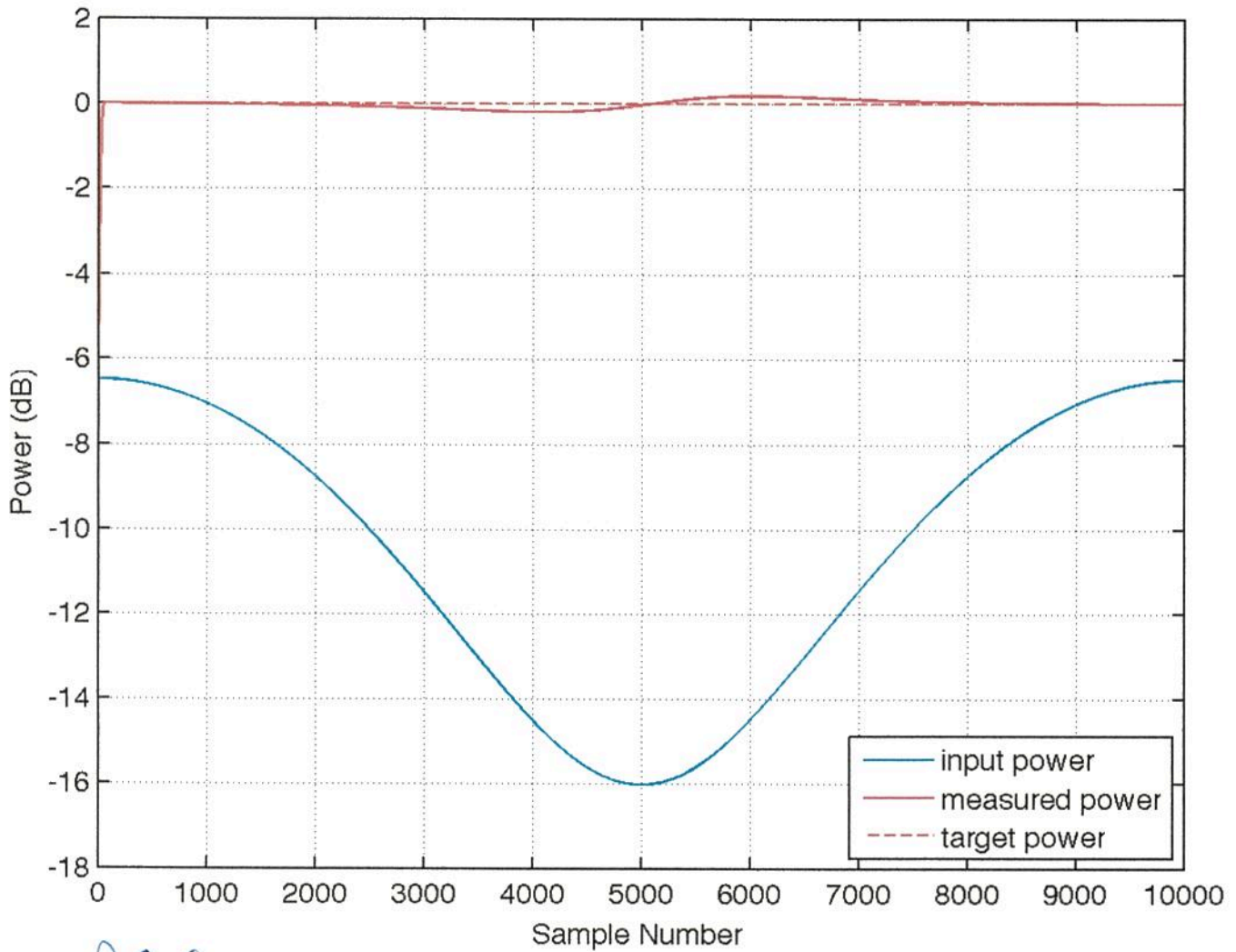
↑
all MATLAB files are here.

AGC, m:

- constant (all ones) input signal
- amplitude/power is slowly time-varying
- + see blue line below



⇒ AGC tracks gain such that output amplitude (solid red line) remains close to target amplitude ($\sqrt{P_{des}}$).



AGC:

- plot of powers in dB
- blue: input power / time-varying
- solid red: output power with AGC remains with < 1dB of target power (red dashed)

P3

Amplitude modulation function is implemented in

- `ampModulate.m`

- See comments in that function for implementation details:

`demo_ampModulate.m` provides test script for this function:

Demo 1: $f_m = 1$ - sinusoidal message
 $f_c = 8$ - carrier frequency

see attached plot of passband signal

Demo 2: - message signal is a speech signal
- $f_c = 8$ kHz - carrier frequency

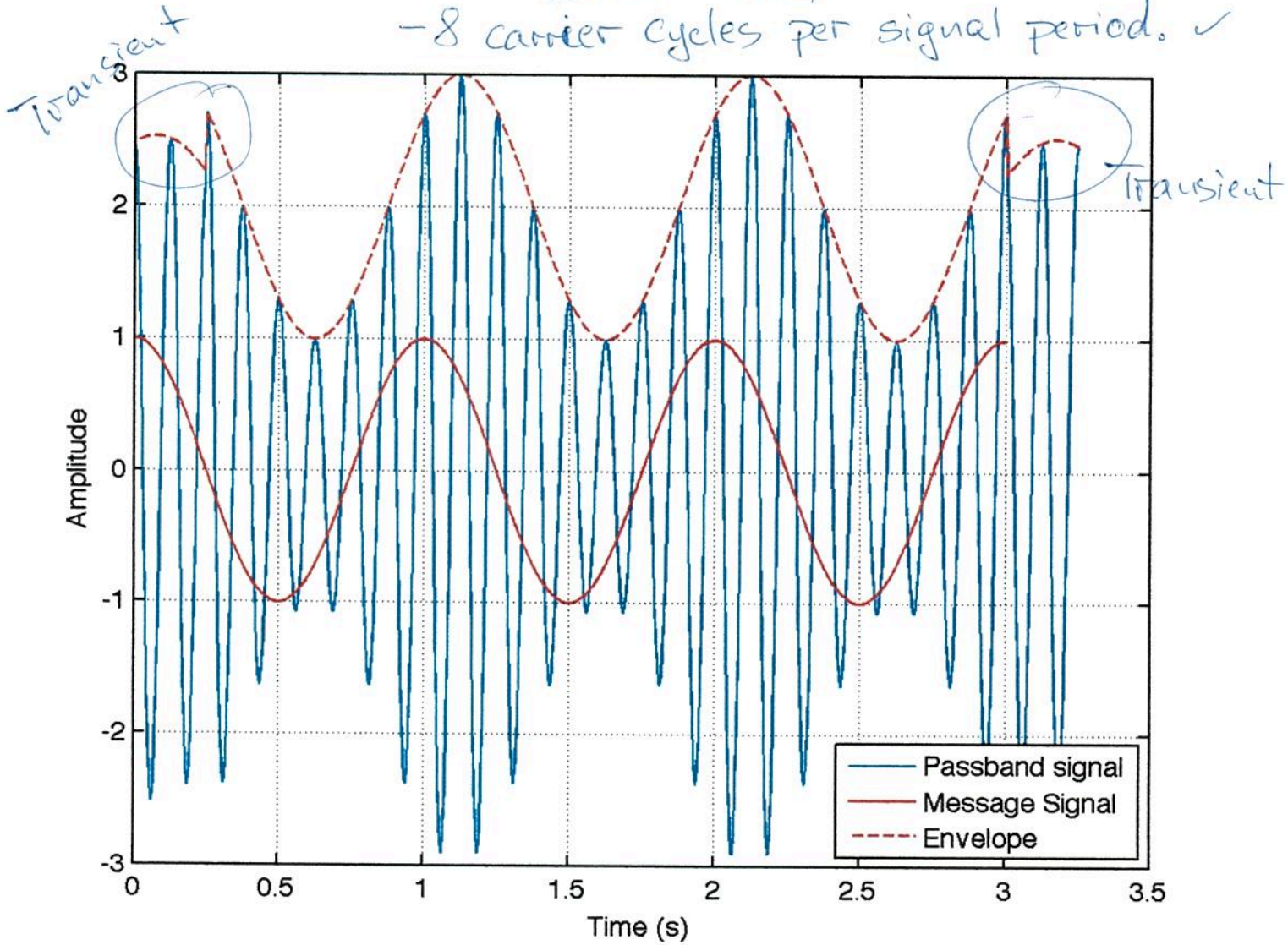
see attached plot of PSD of passband signal

P4: Function - `ampDemodulate.m` (no such)

Demo: - `demo_ampDemodulate.m`
shows demodulation of 2 signals above.

dema ampModulate.m; Demo 1:

- sinusoidal message signal $m(t)$ (solid red)
- modulated signal / passband signal (solid blue)
- 8 carrier cycles per signal period. ✓

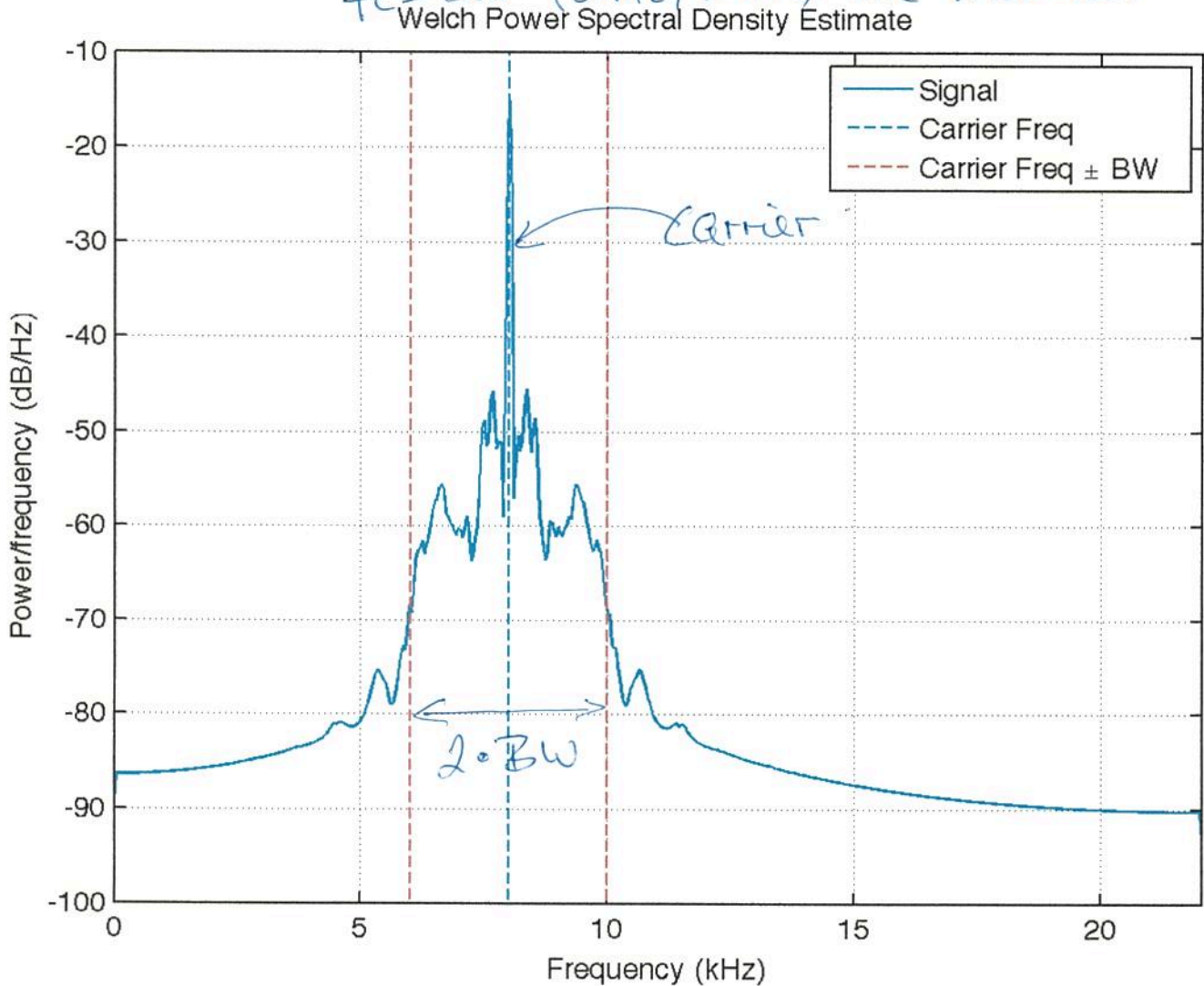


- dashed red: envelope signal:
 $A_c + m(t)$.

- Notes Filter in ~~the~~ modulator induces:
- delay (25 samples)
- transients at start and end.

demo_ampModulate.m: Demo 2

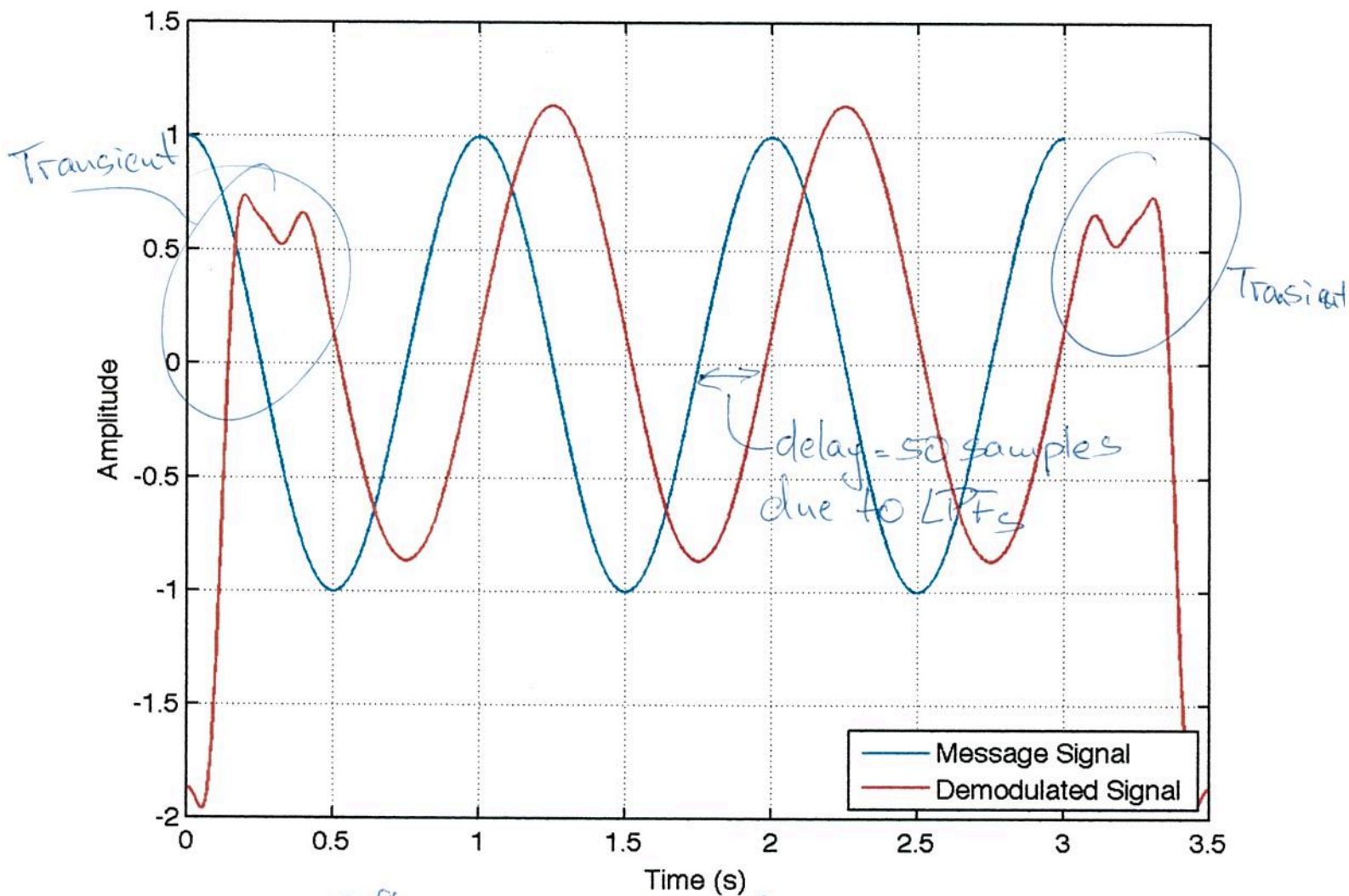
- $m(t)$ = speech signal
- plot shows power spectral density (PSD) of passband signal.
- location of center frequency (8kHz) and $f_c \pm BW$ (6kHz/10kHz) are marked.



- signal is clearly centered at f_c .
- carrier is visible ($a_{mod} = 50\%$)
- At passband, signal is limited to $f_c \pm BW$ ($BW = 2\text{kHz}$)

demo_ampDemodulate: Demo 1

- plot shows
 - + original signal (blue)
 - + demodulated signal (red)



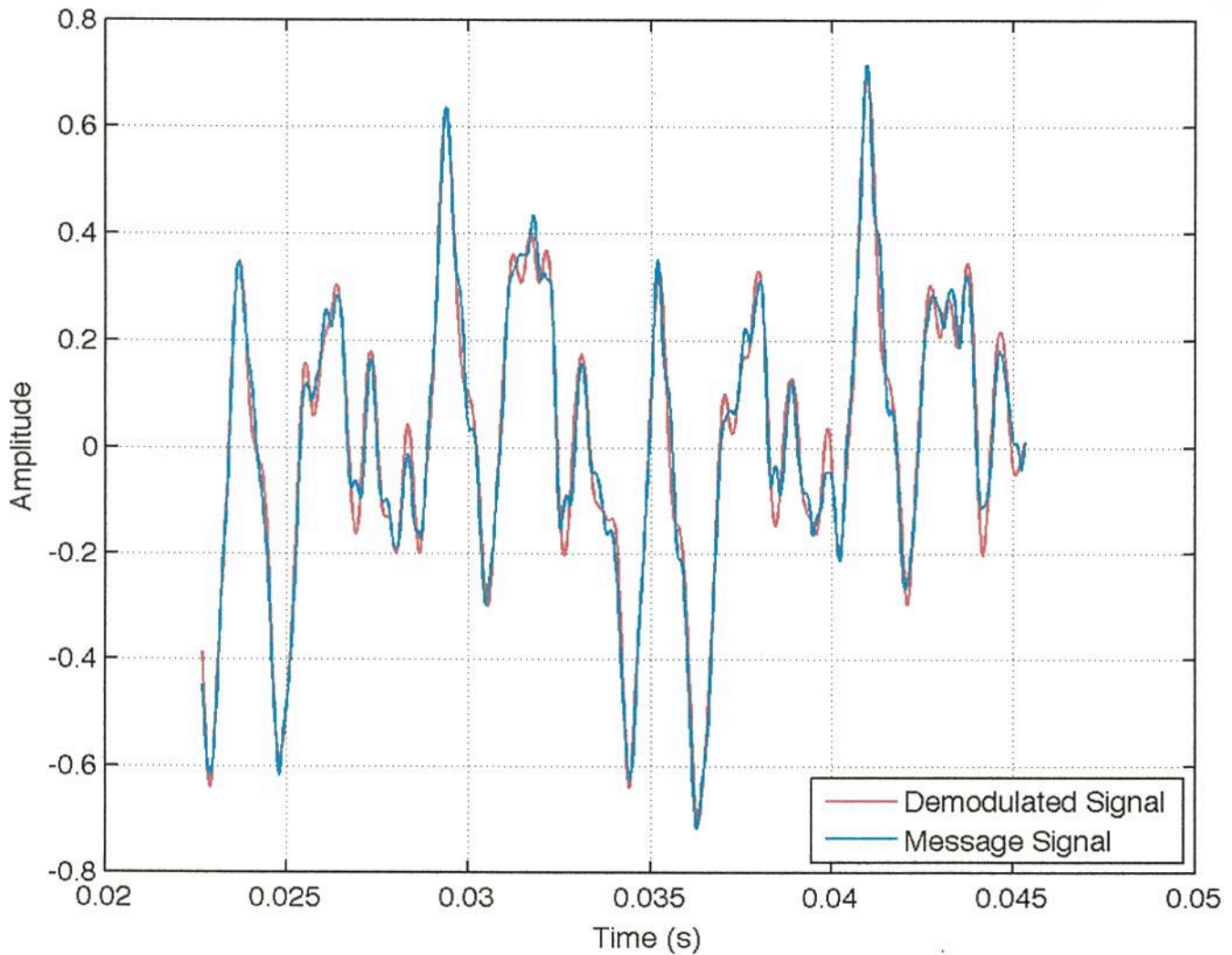
- differences arise from:

- transients due to filtering at both ends of demodulated signal
- delay (50 samples) from filtering
- small vertical offset due to incomplete carrier removal.

demo_ampDemodulation: Demo 2

- Plot shows: (~20ms)
+ original message signal (blue)
+ demodulated signal (red)

- Note: delay induced by 2 LP filters (50 samples) was compensated to align signals



- Very good agreement:
- differences are due to filtering or imperfect suppression of double-frequency tones.