

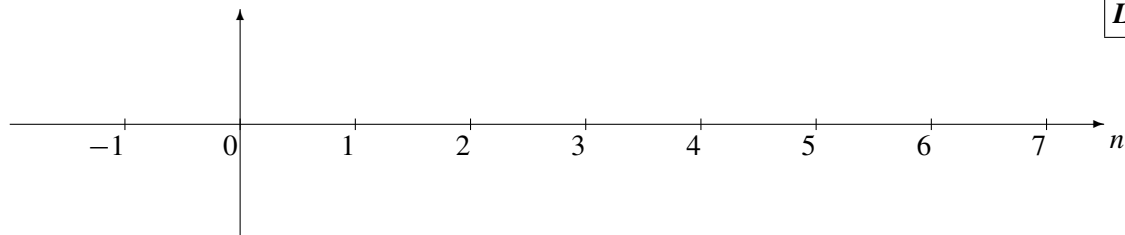


**PROBLEM SPR-06-Q.2.1:**

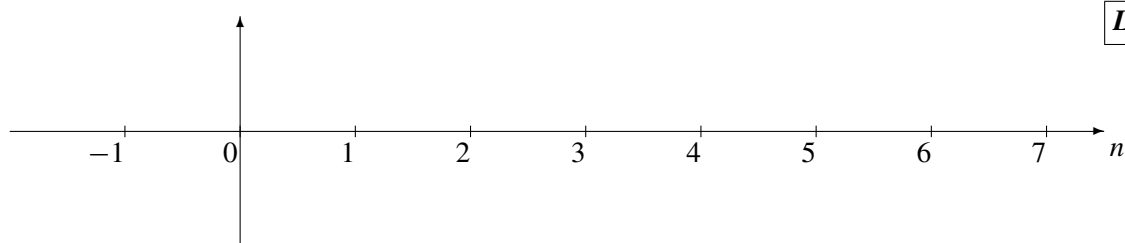
(a) Evaluate the convolution:  $\mathbf{yn} = \mathbf{conv}([1 \ -1 \ 1 \ -1 \ 1], [10 \ 20 \ 30] )$ ;

*Note:* a MATLAB vector implicitly defines a signal to have its starting point at  $n = 0$ .

Give your answer as a *stem plot*.



(b) Make a *stem plot* of the signal  $s[n] = -200(u[n - 1] - u[n - 6])$ .

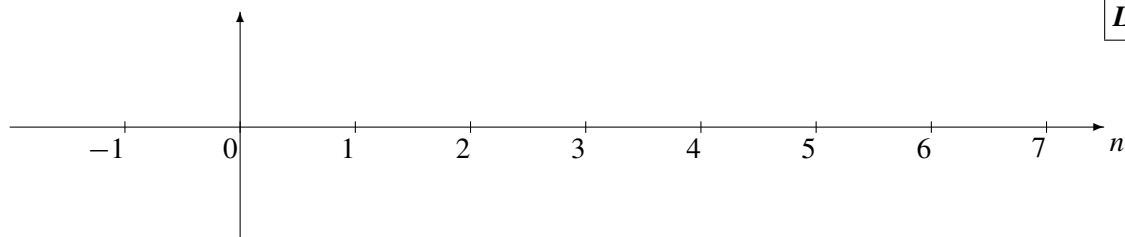


**PROBLEM SPR-06-Q.2.2:**

(a) Determine the impulse response of the system:

$$y[n] = 400x[n - 1] + 400x[n - 5]$$

Give your answer as a *stem plot*.



**Label all points**

(b) Determine the frequency response of the FIR system:

$$y[n] = 400x[n - 1] + 400x[n - 5]$$

Give your answer as a formula **in the following form:**  $H(e^{j\hat{\omega}}) = e^{-j\beta\hat{\omega}} A \cos(\mu\hat{\omega})$   
by finding numerical values for  $A$ ,  $\beta$  and  $\mu$ .

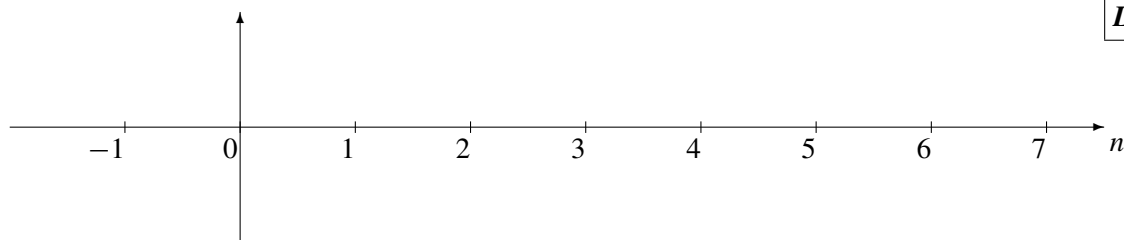
$A =$	$\beta =$	$\mu =$
-------	-----------	---------

**PROBLEM SPR-06-Q.2.3:**

Suppose that the system function of an FIR filter is

$$H(z) = 50z^{-1} (1 + e^{j2\pi/3}z^{-1}) (1 + e^{-j2\pi/3}z^{-1})$$

- (a) Determine the impulse response,  $h[n]$ , of the FIR filter. Give your answer as a *stem plot*.



- (b) Evaluate the magnitude and phase of the frequency response of the FIR filter at  $\hat{\omega} = 2\pi/3$ .

Magnitude =

Phase =

### PROBLEM SPR-06-Q.2.4:

Here are some operations that are often done in MATLAB. In each case, you should analyze the code and determine the answer via mathematics.

- (a) Suppose that a student enters the following MATLAB code:

```
nn = 0:4480099;  
zz = (3 + 4i) * exp(j*(1.7*pi*nn + 389));  
soundsc( real(zz), 3000 )
```

Determine the analog frequency (in Hertz) that will be heard. (There might be folding or aliasing!)

FREQ =  Hz

*Explain your reasoning.*

- (b) Suppose that a student writes the following MATLAB code to generate a sine wave:

```
tt = 0:1/15000:10000;  
xx = cos( 2*pi*6000*tt + pi/3 );  
soundsc( xx, fsamp );
```

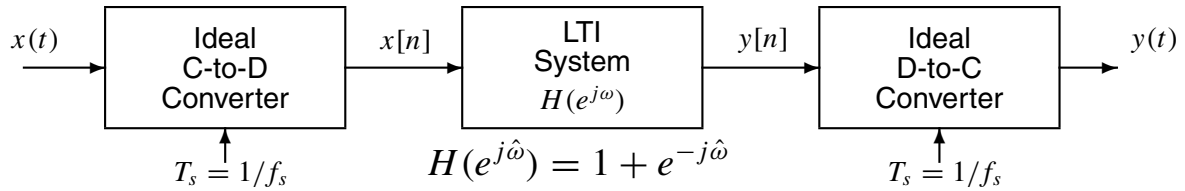
Although the sinusoid was not written to have a frequency of 600 Hz, it is possible to play out the vector `xx` so that it sounds like a 600 Hz tone. Determine the value of **fsamp** that should be used to play the vector `xx` as a 600 Hz tone.

fsamp =  Hz

*Explain your reasoning.*

**PROBLEM SPR-06-Q.2.5:**

Consider the following system for discrete-time filtering of a continuous-time signal:

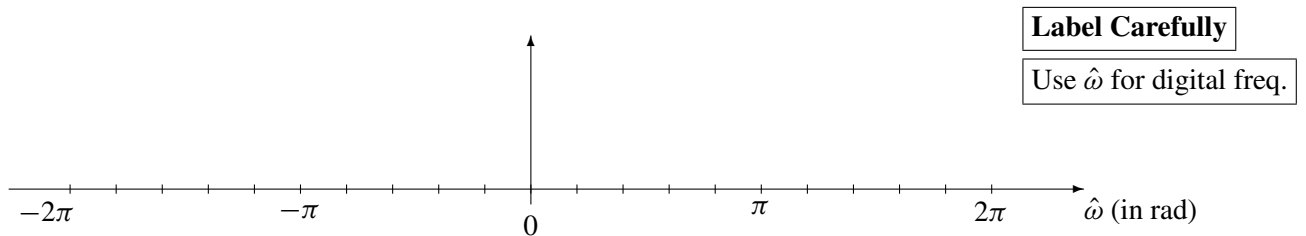


For all parts below, the input to the ideal C-to-D converter is  $x(t) = 500 \cos(300\pi t)$ .

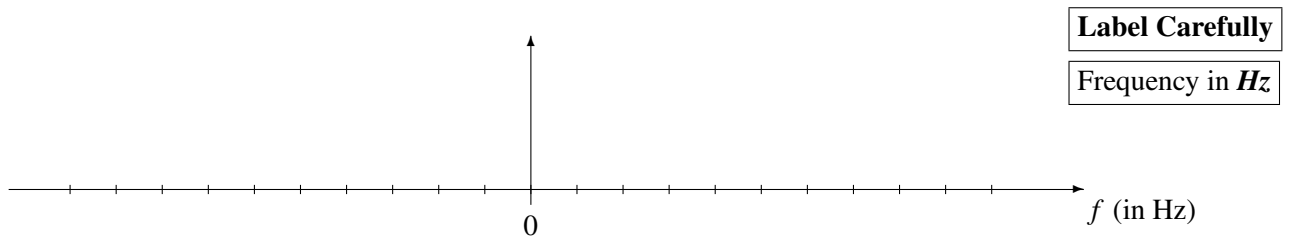
- (a) Determine the Nyquist rate (in hertz) for sampling the input signal  $x(t)$ . Explain.

$f_{\text{Nyquist}} =$    $\text{Hz}$

- (b) If the sampling rate of the C-to-D converter is  $f_s = 400$  samples/sec, make a plot of the spectrum of the discrete-time signal  $x[n]$  over the range of frequencies  $-2\pi \leq \hat{\omega} \leq 2\pi$ . Make sure to show **all spectrum lines** and label the frequency, amplitude and phase of each spectral component.



- (c) If the sampling rate of the the ideal D-to-C converter is  $f_s = 400$  samples/sec, draw the spectrum for the continuous-time output signal,  $y(t)$ . Use  $x(t)$  defined above.



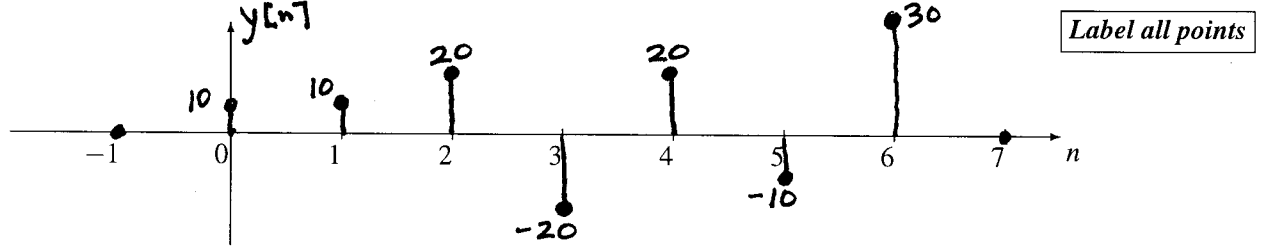


**PROBLEM SPR-06-Q.2.1:**

(a) Evaluate the convolution:  $y_n = \text{conv}([1 \ -1 \ 1 \ -1 \ 1], [10 \ 20 \ 30])$ ;

Note: a MATLAB vector implicitly defines a signal to have its starting point at  $n = 0$ .

Give your answer as a *stem plot*.

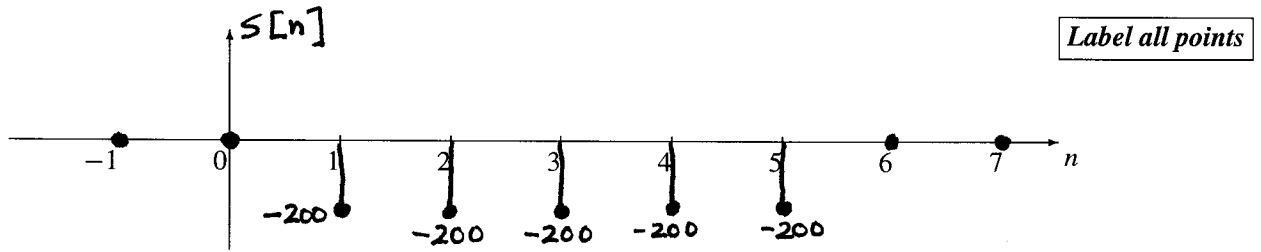


The convolution table is:

$$\begin{aligned}
 y_n &= [10 \ -10 \ 10 \ -10 \ 10 \ 0 \ 0] \\
 &+ [0 \ 20 \ -20 \ 20 \ -20 \ 20 \ 0] \\
 &+ [0 \ 0 \ 30 \ -30 \ 30 \ -30 \ 30] \\
 y_n &= [10 \ 10 \ 20 \ -20 \ 20 \ -10 \ 30]
 \end{aligned}$$



(b) Make a *stem plot* of the signal  $s[n] = -200(u[n - 1] - u[n - 6])$ .



The difference of two unit-step signals is a pulse. The unit-step  $u[n - 1]$  goes “up” at  $n = 1$ ; the unit-step  $-u[n - 6]$  goes “down” at  $n = 6$ , so the final result has two cases:

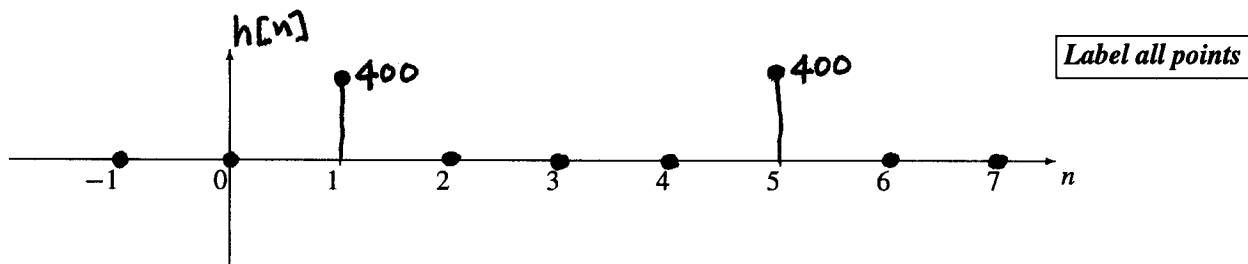
$$-200(u[n - 1] - u[n - 6]) = \begin{cases} -200 & 1 \leq n < 6 \\ 0 & \text{elsewhere} \end{cases}$$



**PROBLEM SPR-06-Q.2.2:**

(a) Determine the impulse response of the system:

$$y[n] = 400x[n - 1] + 400x[n - 5]$$

Give your answer as a *stem plot*.When  $x[n] = \delta[n]$ , you get the impulse response,  $h[n]$ . Thus,

$$h[n] = 400\delta[n - 1] + 400\delta[n - 5]$$

(b) Determine the frequency response of the FIR system:

$$y[n] = 400x[n - 1] + 400x[n - 5]$$

Give your answer as a formula **in the following form**:  $H(e^{j\hat{\omega}}) = e^{-j\beta\hat{\omega}} A \cos(\mu\hat{\omega})$  by finding numerical values for  $A$ ,  $\beta$  and  $\mu$ .

$A = 800$	$\beta = 3$	$\mu = 2$
-----------	-------------	-----------

The frequency response is  $H(e^{j\hat{\omega}}) = \sum_{n=0}^M h[n]e^{-jk\hat{\omega}}$ . Thus,

$$H(e^{j\hat{\omega}}) = 400e^{-j\hat{\omega}} + 400e^{-j5\hat{\omega}} = 400e^{-j3\hat{\omega}} (e^{j2\hat{\omega}} + e^{-j2\hat{\omega}})$$

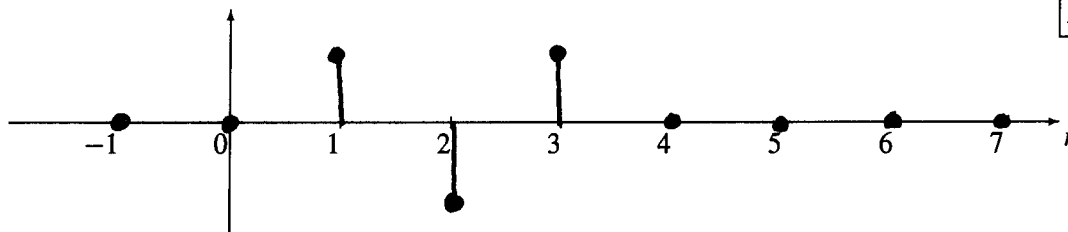
$$H(e^{j\hat{\omega}}) = 800e^{-j3\hat{\omega}} \cos(2\hat{\omega})$$

**PROBLEM SPR-06-Q.2.3:**

Suppose that the system function of an FIR filter is

$$H(z) = 50z^{-1} (1 + e^{j2\pi/3} z^{-1}) (1 + e^{-j2\pi/3} z^{-1})$$

- (a) Determine the impulse response,  $h[n]$ , of the FIR filter. Give your answer as a *stem plot*.



Multiply out the factors and then use the definition of the  $z$ -Transform,  $H(z) = \sum_{n=0}^M h[n]z^{-n}$ , to get the impulse response from the coefficients of the  $z$ -Transform polynomial:

$$H(z) = 50z^{-1} (1 + 2 \cos(2\pi/3)z^{-1} + z^{-2}) = 50z^{-1} - 50z^{-2} + 50z^{-3}$$

$$\implies h[n] = 50\delta[n - 1] - 50\delta[n - 2] + 50\delta[n - 3]$$

- (b) Evaluate the magnitude and phase of the frequency response of the FIR filter at  $\hat{\omega} = 2\pi/3$ .

Magnitude = 100
-----------------

Phase = $-\pi/3 = -1.047$ radians
-----------------------------------

Evaluating  $H(e^{j\hat{\omega}})$  at  $\hat{\omega} = 2\pi/3$  is the same as evaluating  $H(z)$  at  $z = e^{j2\pi/3}$ , so we obtain

$$H(e^{j2\pi/3}) = 50e^{-j\pi/3} (1 - e^{j2\pi/3} e^{-j\pi/3}) (1 - e^{-j2\pi/3} e^{-j\pi/3})$$

which reduces to

$$\begin{aligned} H(e^{j2\pi/3}) &= 50 (e^{-j\pi/3} - e^{-j0} - e^{-j4\pi/3} + e^{-j\pi}) \\ &= 100e^{-j\pi/3} = 100e^{-j1.047} \end{aligned}$$

### PROBLEM SPR-06-Q.2.4:

Here are some operations that are often done in MATLAB. In each case, you should analyze the code and determine the answer via mathematics.

- (a) Suppose that a student enters the following MATLAB code:

```
nn = 0:4480099;  
zz = (3 + 4i) * exp(j*(1.7*pi*nn + 389) );  
soundsc( real(zz), 3000 )
```

Determine the analog frequency (in Hertz) that will be heard. (There might be folding or aliasing!)

**FREQ = 450** Hz

The discrete-time frequency of  $\hat{\omega} = \pm 1.7\pi$  will alias to  $\hat{\omega} = \mp 0.3\pi = \mp 2\pi(0.15)$ , so the analog frequency of the output is  $f_{\text{out}} = \frac{\hat{\omega}}{2\pi} f_s = (0.15)(3000) = 450$  Hz.

- (b) Suppose that a student writes the following MATLAB code to generate a sine wave:

```
tt = 0:1/15000:10000;  
xx = cos( 2*pi*6000*tt + pi/3 );  
soundsc( xx, fsamp );
```

Although the sinusoid was not written to have a frequency of 600 Hz, it is possible to play out the vector **xx** so that it sounds like a 600 Hz tone. Determine the value of **fsamp** that should be used to play the vector **xx** as a 600 Hz tone.

**fsamp = 1500** Hz

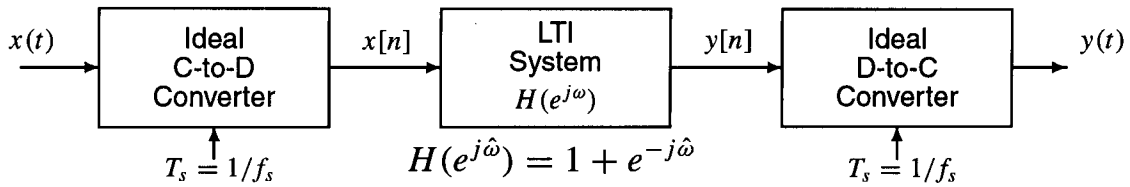
The **xx** signal is actually a discrete-time signal that came from a sinusoid of frequency 6000 Hz, sampled at 15000 Hz. Thus the discrete-time frequency is  $\hat{\omega} = 2\pi(6000/15000) = 2\pi(0.4)$  which does **not** alias. In order to get a specified output frequency of 600 Hz, we must solve for  $f_s$  in

$$f_{\text{out}} = \frac{\hat{\omega}}{2\pi} f_s \quad \Rightarrow \quad 600 = \frac{2\pi(2/5)}{2\pi} f_s$$

Thus, the sampling frequency should be  $f_s = (5/2)(600) = 1500$  Hz.

**PROBLEM SPR-06-Q.2.5:**

Consider the following system for discrete-time filtering of a continuous-time signal:



For all parts below, the input to the ideal C/D converter is  $x(t) = 500 \cos(300\pi t)$ .

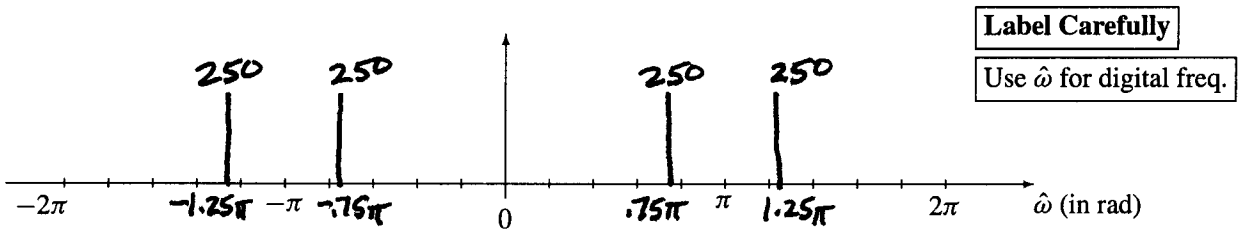
- (a) Determine the Nyquist rate (in hertz) for sampling the input signal  $x(t)$ . Explain.

$$f_{\text{Nyquist}} = \boxed{300} \text{ Hz}$$

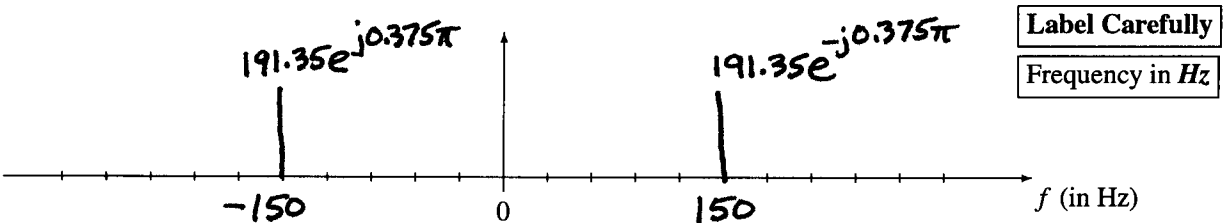
The Nyquist Rate is two times the highest frequency in  $x(t)$ , whose only frequency is 150 Hz. The Sampling Theorem says that you must sample at greater than the Nyquist Rate to be able to recover the signal.

- (b) If the sampling rate of the C-to-D converter is  $f_s = 400$  samples/sec, make a plot of the spectrum of the discrete-time signal  $x[n]$  over the range of frequencies  $-2\pi \leq \hat{\omega} \leq 2\pi$ . Make sure to show all spectrum lines and label the frequency, amplitude and phase of each spectral component.

The signal  $x[n]$  has a discrete-time frequency of  $\hat{\omega} = 300\pi/400 = 0.75\pi$ , so there will be spectral lines at  $\hat{\omega} = \pm 0.75\pi \pm 2\pi$ . In the figure the spectrum lines will be at  $\hat{\omega} = \{0.75\pi, -1.25\pi, -0.75\pi, 1.25\pi\}$ .



- (c) If the sampling rate of the the ideal D-to-C converter is  $f_s = 400$  samples/sec, draw the spectrum for the continuous-time output signal,  $y(t)$ . Use  $x[n]$  determined in the previous part.



The frequency response at  $\hat{\omega} = 0.75\pi$  is

$$H(e^{j\hat{\omega}}) = 1 + e^{-j0.75\pi} = 0.7654e^{-j0.375\pi}$$

Therefore, the output of the filter is  $y[n] = 382.7 \cos(0.75\pi n - 0.375\pi)$ , and converting from  $\hat{\omega}$  to the output frequency ( $\omega_{\text{out}} = \hat{\omega} f_s$ ) gives

$$y(t) = 382.7 \cos(300\pi t - 0.375\pi)$$

Thus the complex amplitudes are  $191.35e^{\mp j0.375\pi}$  at  $f = \pm 150$  Hz.