

## Problem Set #2

Math 8600

November 12, 2002

**Exercise 1** (*Oversampling*) In this problem you will show that a positive effect of sampling a band limited signal  $f(t)$  at a rate faster than the Nyquist rate is that the expansion for  $f(t)$  in terms of the sampled values converges at a faster rate. (The negative effect is that you have to sample more often.)

1. Suppose  $f$  satisfies the hypotheses of the Shannon sampling theorem proven in the notes; in particular it is a band limited signal with  $\hat{f}(\lambda) = 0$  for  $|\lambda| \geq \Omega$ . Fix  $a > 1$  and reprove the theorem to show that

$$\hat{f}(\lambda) = \sum_{n=-\infty}^{\infty} c_{-n} e^{-\frac{in\pi\lambda}{a\Omega}}, \quad c_{-n} = \frac{\pi}{a\Omega} f\left(\frac{n\pi}{a\Omega}\right).$$

2. Let

$$\hat{g}_a(\lambda) = \begin{cases} 0 & \text{if } |\lambda| > a\Omega \\ \frac{\lambda+a\Omega}{(a-1)\Omega} & \text{if } -a\Omega \leq \lambda < -\Omega \\ 1 & \text{if } -\Omega \leq \lambda < \Omega \\ -\frac{\lambda-a\Omega}{(a-1)\Omega} & \text{if } \Omega \leq \lambda \leq a\Omega \end{cases}$$

Show that

$$g_a(t) = \frac{\cos \Omega t - \cos a\Omega t}{\pi(a-1)\Omega t^2}.$$

3. Since  $\hat{f}(\lambda) = 0$  for  $|\lambda| \geq \Omega$ , we see that  $\hat{f}(\lambda) = \hat{f}(\lambda)\hat{g}_a(\lambda)$ . Prove that

$$f(t) = \sum_{n=-\infty}^{\infty} \frac{\pi}{a\Omega} f\left(\frac{n\pi}{a\Omega}\right) g_a\left(t - \frac{n\pi}{a\Omega}\right).$$

Since  $g_a(t)$  has a factor of  $t^2$  in the denominator, this expression for  $f(t)$  converges faster than the expression in the Shannon theorem. Note that the  $n$ th term behaves like  $1/n^2$ , rather than  $1/n$ .

**Exercise 2** (Filtering with the FFT) Let

$$f(t) = e^{-\frac{t^2}{10}} (\sin 2t + 2 \cos 4t + 0.4 \sin t \sin 50t).$$

Discretize  $f$  by setting  $y_k = f(2k\pi/256)$ ,  $k = 1, \dots, 256$ . Use MATLAB's FFT to compute  $\hat{y}_k$  for  $0 \leq k \leq 255$ . (Note that  $y_{n-k} = \overline{y_k}$ . Thus the low frequency coefficients are  $\hat{y}_0, \dots, \hat{y}_m$  and  $\hat{y}_{256-m}, \dots, \hat{y}_{256}$  for some small integer  $m$ . Filter out the high-frequency terms by setting  $\hat{y}_k = 0$  for  $m \leq k \leq 255 - m$  with  $m = 6$ . Then apply the inverse FFT to these filtered  $\hat{y}_k$  to compute the filtered  $y_k$ . Plot the results and compare with the original unfiltered signal. Experiment with several different values of  $m$ .

**Exercise 3** (Compression with the FFT) Consider the signal  $f(t)$  as given in the previous problem. Let  $\text{tol} = 1.0$ . In the previous problem compress the transformed signal by setting  $\hat{y}_k = 0$  whenever  $|\hat{y}_k| < \text{tol}$ . Apply the inverse FFT to the compressed transformed signal to get a compressed signal  $y_k$ . Plot the results and compare with the original uncompressed signal. Experiment with several different values of  $\text{tol}$ . Keep track of the percentage of Fourier coefficients that have been filtered out.

**Exercise 4** Construct the unitary rep  $\tilde{\mathbf{T}}_n$  of  $H_R$  induced by the one-dimensional rep  $\tilde{\mathbf{T}}_0^n(a_1, a_2, y_3) = e^{2\pi i n y_3}$  of the subgroup  $H^1$ , where  $n$  is an integer (not necessarily positive). Determine the action of  $H_R$  on the rep space. Under what conditions on  $n$  is  $\tilde{\mathbf{T}}_n$  irred?

**Exercise 5** Suppose  $f \in L_2(\mathbb{R})$  such that  $f_{\mathbf{P}} \neq 0$  almost everywhere. Prove that the set  $\{e^{2\pi i(m_1 x_1 + m_2 x_2)} f_{\mathbf{P}} / |f_{\mathbf{P}}|, m_1, m_2 = \pm 1, \pm 2, \dots\}$  is an ON basis for the lattice Hilbert space  $V^!$ . Find an explicit expression for the corresponding ON basis of  $L_2(\mathbb{R})$  obtained from the mapping  $\mathbf{P}^{-1}$ .

The following MATLAB routines should be useful:

```
function fc=compress( f, r)
% Input the vector f and ratio r: 0<= r <=1.
```

```

% The output is the vector fc in which the smallest
% 100r% of the terms f_k, in absolute value, are set
% equal to zero.
if (r<0) | (r>1)
    error ('r should be between 0 and 1')
end;
N=length(f); Nr=floor(N*r);
ff=sort(abs(w));
tol=abs(w(Nr+1));
fc=(abs(w)>=tol).*w;

```

You can discretize the interval  $[0, 2\pi]$  and read in the signal as a vector by using the commands

```

t=linspace(0,2*pi,2^8);
f=exp(-t.^2/10).*(sin(2*t)+2*cos(4*t)+0.4*sin(t).*sin(50*t));

```

If  $\hat{f}$  is the FFT of  $f$ , you can filter out high frequency components from  $\hat{f}$  with a command such as

```

filterhatf=[ hatf(1: m) zeros(1, 2^8-2*m) hatf(2^8-m+1:2^8)]

```

```

function L2error =fftcomp(t,f,r)
% Input: time vector t, signal vector f, compression rate r, (between
% 0 and 1)
Output: graph of f, graph of the compression of f, and the relative L2
error
if (r<0) | (r>1)
    error ('r should be between 0 and 1')
end;
hatf=fft(f);
hatfc=compress(hatf,r);
fc=ifft(hatfc);
plot(t,f,t,fc)
L2error=norm(f-fc,2)/norm(f)

```