**Lab 5 System, Convolution and Filter**

|  |  |
| --- | --- |
| **Author** | 周安然 12011214 冯柏钧 12011124 |
| **Introduction**  The Lab5 is an overview about system, convolution and filter. And we learn how to deal with acoustic signal.  1.Designing a filter via Matlab,which is about Butterworth filter design&Different filter type.  2.Using filtering to generate speech shaped noise and extract signal envelop  3.Adjusting the signal intensity: adjusting signal-to-noise ratio (SNR) level &normalizing signal energy.  **Lab results & Analysis**：  Question(1)    Results  The SSN:    The 2 speech signal:    Question(2)    Results    Analysis  To adjust the SNR, i choose to let x2 = (x/norm(x))\*norm(SSN)\*(10^(-0.25)); Then it transform SNR to -5db.  As for y = x + SSN, let y2 = (y/norm(y))\*norm(x).  snr is the original SNR of x and ssn. The snr2 is the modified SNR of x and ssn. The snr3 is the SNR of modified y and x. There exists an neglected error between modified y and x.  **Note**: Please indicate meaning of the symbols in all expressions. Please indicate the coordinate and unit in all figures. | |
| **Experience**  Design different types pf filter is quite interesting. I made a neglected error when modifying y and x in Assignment 2. As for Assignment 1, it is easy to handle with the help of PPT slides.  0dd4b232c0c38e733c6f93df037c9579e1028a78276034fde06bc77e22f360b5662a259407238a6353d9a41b0cf1b04878e761d5195b7a82cf583ead10f1 | |
| **Score** | 95 |

**Code**

**Assignment 1**

clc;clear;

N = 1000; fs = 16000;

noise = 1-2\*rand(1,N);

sig = repmat(noise, 1, 10);

[pxx, w] = pwelch(sig,[],[],5112,fs);

b = fir2(3000,w/8000, sqrt(pxx/max(pxx)));

[h, wh] = freqz(b, 1, 128);

ssn = filter(b, 1, noise);

[pssn, wssn] = pwelch(ssn,[],[],512,fs);

[s1,fs1] = audioread("C\_01\_01.wav");

[s2,fs2] = audioread("C\_01\_02.wav");

figure(1);

plot(wssn,pssn);xlabel("The ssn");ylabel('fs = 16000');

[ps1, ws1] = pwelch(s1,[],[],512,fs1);

[ps2, ws2] = pwelch(s2,[],[],512,fs2);

figure(2);

subplot(211);

plot(ws1,ps1);xlabel('C\_01\_01.wav');ylabel('fs = 16000');

subplot(212);

plot(ws2,ps2);xlabel('C\_01\_02.wav');ylabel('fs = 16000');

**Assignment 2**

clc;clear;

[x,fs] = audioread("C\_01\_01.wav");

N = length(x);fs = 16000;

noise = 1-2\*rand(1,N);

sig = repmat(noise,1,10);

[pxx, w] = pwelch(sig,[],[],512,fs);

b = fir2(3000,w/8000,sqrt(pxx/max(pxx)));

[h,wh] = freqz(b,1,128);

ssn = filter(b,1,noise);

ssn = ssn';

snr = 20\*log10(norm(x)/norm(ssn));

x2 = (x/norm(x))\*norm(ssn)\*(10^(-0.25));

snr2 = 20\*log10(norm(x2)/norm(ssn));

y = x + ssn;

y2 = (y/norm(y))\*norm(x);

snr3 = 20\*log10(norm(y2)/norm(x));