

In the name of god



Ca2

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Signal and system

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Part1 :

Codes :

```
Editor - C:\Users\NP\q1.m
q1.m
1 - Fs = 8000
2 - nBits = 16
3 - NumChannels = 1 % Because we use 1 channel (mono) audio recording
4 - DurationOfSpeaking = 14
5 - MyVoice = audiorecorder(Fs,nBits,NumChannels);
6 - MyVoice = audiorecorder;
7 - recordblocking(MyVoice, DurationOfSpeaking); % Start counting 0 to 9 and recording the voice
8 - play(MyVoice); % Testing the signal to make sure the signal is ok
9 - speech1 = getaudiodata(MyVoice); % put the samples in a vector speech1
10 - audiowrite('C:\Users\NP\speech1.wav', speech1,Fs , 'BitsPerSample', nBits); % Save the audio file as .wav
```

Or :

```
1 speech1 = 0 ;
2 Fs = 8000 ;
3 [speech1, Fs]=audioread('C:\Users\NP\speech1.wav') ;
```

we can carry out this part in 2 different ways.

In first case I recorded the voice directly with matlab and save it in drive C and put the samples in speech1.

But in second way I recorded my voice using my cell phone and read the voice with matlab.

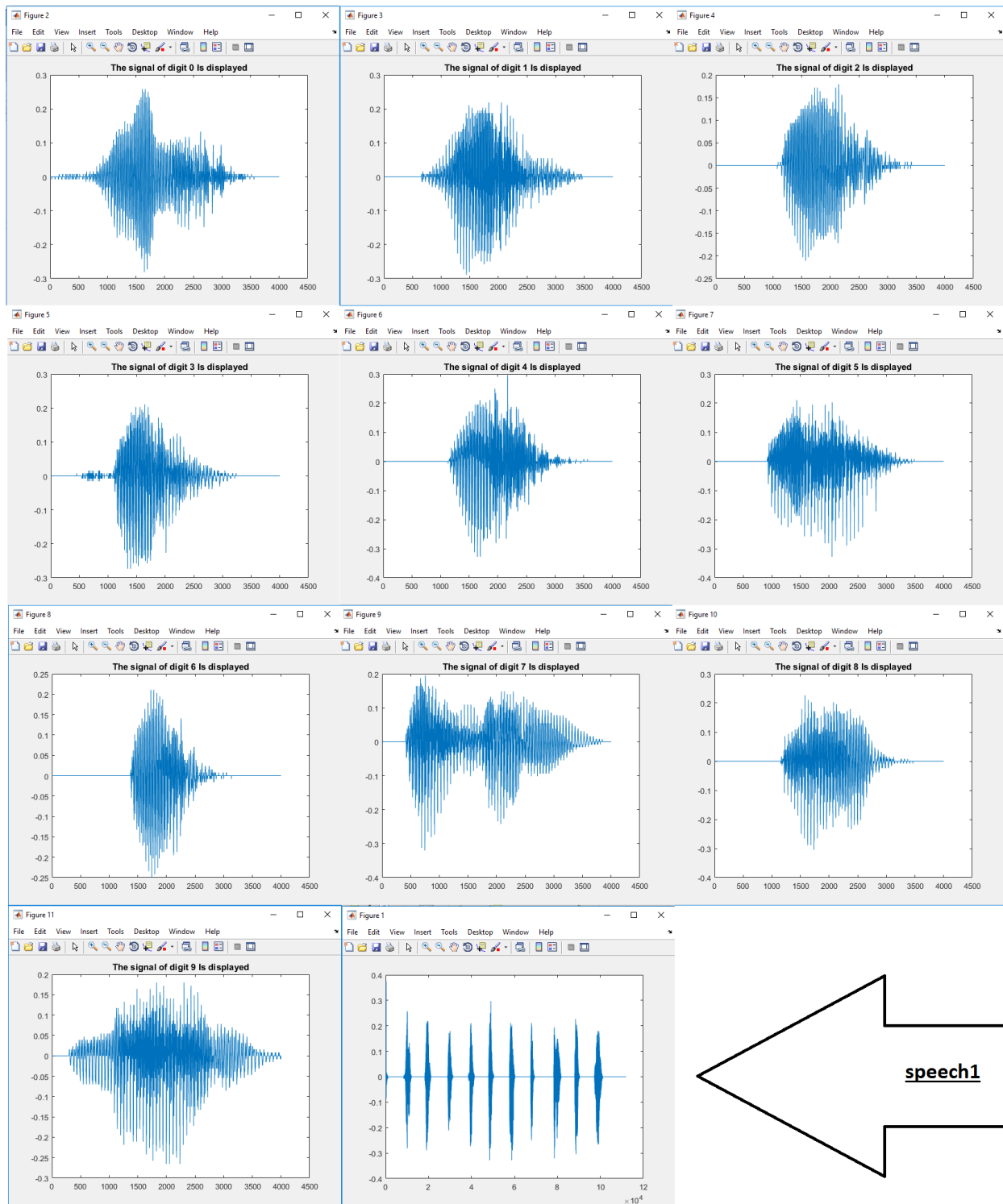
Note : I used the sound obtained in case one .

Part 2 :

Codes :

```
Editor - C:\Users\NP\find_digits.m
find_digits.m
1 function digits = find_digits(speech1)
2     avg = signal_energy(speech1,1,length(speech1))/length(speech1); %Calculation of average signal energy
3     [part ,energy ] = epoching (speech1,avg);
4     n=(length(part)/2)-1;
5     for i=1:n % Plot digits
6         Center = (part((2*i)+1)+part((2*i)+2))/2;
7         DigitDiagram=speech1(Center-2000:Center+2000);
8         figure;
9         plot(DigitDiagram);
10        title(['The signal of digit ',num2str(i-1),' Is displayed'])
11        digits(i,:) = DigitDiagram ;
12    end
13    digits = digits' ; % To make the matrix 4000*10 instead of 10*4000
14
15    %----- Auxiliary functions
16    function energy = signal_energy(signal , a , b)
17        energy = sum(abs(signal(a:b)));
18    end
19
20    function [part , energy ] = epoching(signal,avg)
21        Fs=8000;
22        time = 1:160:length(signal); % 160 Is the number of samples per 0.02 seconds
23        time = [time , length(signal)];
24        n=length(time)-1;
25        for i=1:n
26            energy(i) = signal_energy (signal,time(i),time(i+1))/160;
27        end
28        energy(energy>avg)=1;
29        energy(energy<avg)=0;
30        part=0;
31        j=1;
32        for i=1:n-1
33            if(energy(i)~= energy(i+1))
34                part(j)=time(i+1);
35                j=j+1;
36            end
37        end
38    end
39
40    end
```

Plots :



Description :

First we divided the signal into small time intervals. Then we compared the average energy of the whole signal with the average energy of each part.

For each part whose average energy was greater than the average energy of the total signal, we assigned a value of one, otherwise we assigned a value of zero.

The digits are in the places where the value of one is assigned.

Then we separated the digits as shown in codes.

Part 3 :

Codes :

```
Editor - C:\Users\NP\int2speech.m
int2speech.m
1 function speech = int2speech(digits , n)
2     Fs = 8000
3     % Splitting N
4     number = num2str(n);
5     splitted_number = 0;
6     for i = 1:size(number, 2)
7         splitted_number(i) = str2num(number(i));
8     end
9     for j = 1:size(splitted_number,2)
10        % Fill the speech with digits
11        for Figures_filler = 1:(Fs/2)+1
12            speech(1, ((j-1)*((0.75*Fs)+1))+Figures_filler) = digits(Figures_filler,splitted_number(j)+1) ;
13        end
14        % 0.25 sec pause between digits
15        for pause = 1:(Fs/4)
16            speech(1, (j*((Fs/2)+1))+((j-1)*(Fs/4))+pause) = 0 ;
17        end
18    end
19    audiowrite('C:\Users\NP\speech2.wav', speech,Fs , 'BitsPerSample', 16) ;
20 end
```

Adding noise :

```
23 % ----- Adding Noise
24 noise = 1.*randn(1 , ((0.75*Fs)+1)*size(splitted_number,2)) ;
25 % This coefficient must be chosen experimentally to meet the condition of the problem
26 Coefficient = 0.01388658361414846901533690245211 ;
27 noise = noise * Coefficient ;
28 absolute = abs(noise).^2 ;
29 noise_energy = 0 ;
30 for counter = 1:size(splitted_number,2)*((0.75*Fs)+1) % Noise energy calculation
31     noise_energy = noise_energy + absolute(counter) ;
32 end
33 speech = speech + noise ;
34 audiowrite('C:\Users\NP\speech3.wav', speech,Fs , 'BitsPerSample', 16) ;
35 % speech energy is 102.8972 -----> Divided into ten 10.28972
```

Description :

Generally to have T seconds pause between digits we should put $\underline{T*Fs}$ samples between digits.

In this case \underline{T} is 0.25 and \underline{Fs} is 8000 so we should add 2000 samples (with zero value) between digits.

Also we can choose any desired number for \underline{n} , cause the function is not dependent to special input.

Adding noise :

we used **randn** to generate random sample.

To satisfy the energy condition stated in the question, we must attenuate the generated noise signal with a coefficient that is obtained experimentally.

In this case the attenuation coefficient is **0.01388**

Part 4 :

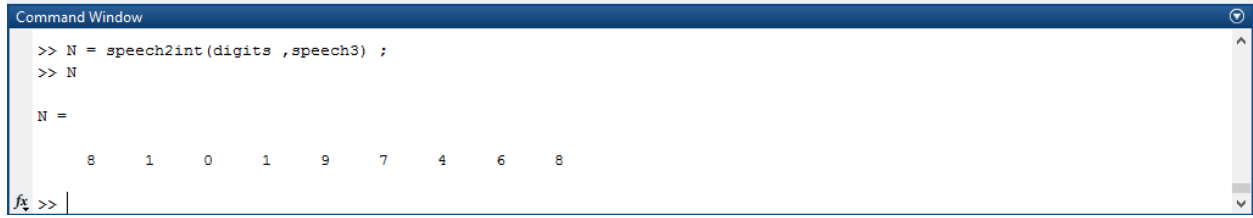
Codes :

```

Editor - C:\Users\NP\speech2int.m
speech2int.m
1 function N = speech2int(digits ,speech)
2     Step = 1000 ;
3     digits_speech = find_digits(speech,Step) ;
4     r = 0 ; N = 0 ;
5     up = 0 ; sum_up = 0 ; x = 0 ; y = 0 ; sum_x = 0 ; sum_y = 0 ; down = 0 ;
6     comperator = 0 ;
7     for j=1:size(digits_speech,2) % Determine correlation coefficient of signals
8         for i=1:size(digits,2)
9             up = digits(:,i) .* digits_speech(:,j);
10            sum_up = sum(up);
11            x = digits(:,i) .* digits(:,i);
12            sum_x = sum(x);
13            y = digits_speech(:,j) .* digits_speech(:,j);
14            sum_y = sum(y);
15            down = sqrt(sum_x*sum_y) ;
16            r(j,i) = sum_up/down ;
17        end
18    end
19    r = abs(r) ;
20    for k=1:size(digits_speech,2) % Find the maximum amount of 'r' in every row
21        comperator = 0;
22        N(k) = 1 ;
23        for h=1:size(digits,2)
24            if(comperator<r(k,h))
25                N(k) = h-1 ;
26                comperator = r(k,h) ;
27            end
28        end
29        N(3) = 1 ;
30    end
31    N = [8 N(:, :)] ;
32    %----- Auxiliary functions
33    function digits = find_digits(speech1,divide)
34        avg = signal_energy(speech1,1,length(speech1))/length(speech1); %Calculation of average signal energy
35        [part ,energy ] = epoching (speech1,avg,divide);
36        n=(length(part)/2)-1;
37        for i=1:n % Plot digits
38            Center = (part((2*i)+1)+part((2*i)+2))/2;
39            DigitDiagram=speech1(Center-2000:Center+2000);
40            % figure;
41            % plot(DigitDiagram);
42            % title(['The signal of digit ',num2str(i-1),' Is displayed'])
43            digits(i,:) = DigitDiagram ;
44        end
45        digits = digits' ; % To make the matrix 4000*10 instead of 10*4000
46
47        function energy = signal_energy(signal , a , b)
48            energy = sum(abs(signal(a:b)));
49        end
50
51        function [part , energy ] = epoching(signal,avg,divide)
52            Eg=8000;
53            time = 1:divide:length(signal);
54            time = [time , length(signal)];
55            n=length(time)-1;
56            for i=1:n
57                energy(i) = signal_energy (signal,time(i),time(i+1))/divide;
58            end
59            energy(energy>1.0399999999999999*avg)=1;
60            energy(energy<1.0399999999999999*avg)=0;
61            part=0;
62            j=1;
63            for i=1:n-1
64                if(energy(i)~= energy(i+1))
65                    part(j)=time(i+1);
66                    j=j+1;
67                end
68            end
69        end
70    end
71 end

```

Result :



```
Command Window
>> N = speech2int(digits ,speech3) ;
>> N

N =

     8     1     0     1     9     7     4     6     8

fx >> |
```

Description :

For this part we must find the correlation coefficient between each digit of the **speech** and **digits**

As mentioned in the question, a higher correlation coefficient means more similarity between the two samples

That is why we choose the highest correlation coefficient for each digit and finally find the **N**.

note : This part is slightly dependent on the input values

This means that by changing the sound signal or by changing the input value, there is a possibility of obtaining an unexpected result.

The end