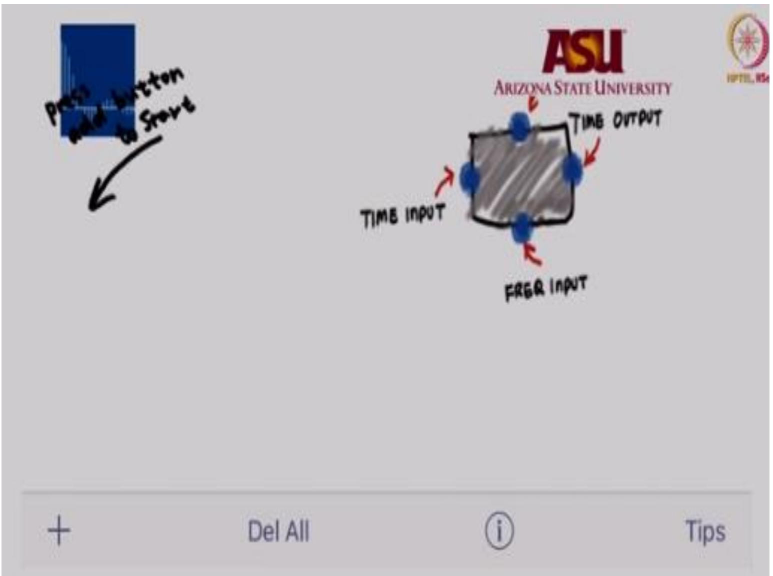
Real-Time Digital Signal Processing
Prof. Rathna G. N.
Department of Electrical Engineering
Indian Institute of Science, Bengaluru

Module No # 11
Lecture No # 55
Lab: Using JiDSP

Welcome back to real time digital signal processing lab, so today we will see how you can use your own mobile either it is android platform or iPhone you can run your some of the real time signal processing experiments on it.

(Refer Slide Time: 00:46)



So, this has been developed by using Java from Arizona State University say they call it as what I am using today for the recording is JDSP basically. So, they have for the android AJDSP so you can use that or in the normal thing they have the JDSP basically Java based DSP project which has been developed so you can use them. So, what I am using as I said you is seeing the screen that we have the unit.

(Refer Slide Time: 01:20)

So, here manuals have been given so you can go and then check with them so here we will do some experiments whatever we had done the thing. So, we will see that how signal generator is going to generate some signals.

(Video Starts: 01:36)

So, whether we want the sinusoid signal, rectangular or triangular or delta exponentially any of the functions what you can select from this. So, here we will select the sinusoid so I will be doing the save actually. And the gain is given as 1 and pulse width is 20 and we have not taken any time shift for this and frequency in pi radians that is 0.2π radians what it has been selected so default what it is there I am taking it.

So, we are adding a block as you can see it is a block level design what you will be doing it next what we will do is I will see what kind of sine wave I am going to get it so we will do a plot. So, it asks our ad plot so I have added the thing so now the connectivity is going to be between output of the sine generation and then the plot so we will click on it.

So, I want to see the magnitude of it so you are seeing the sine wave how it is looking like so you can have it as linear or in DB if you want to see the thing continuous or discontinuous what you want to see you can do it. So, linear sine wave what I am displaying it here so number of samples there if you want you can increase also. So what we will do is if you click on the thing so it will ask me to delete it so I can delete this. So, now what we will do is I will select one more sine generator so here also I am clicking on the sinusoid same.

In this case what I will do is I can select the values what I want so we will select it as 0.3 in this case and then save it and then the pulse width if you want you can increase the think so we will make it as a 30 and then save it. And then add them so I have a 2 sine waves what I have it now what I can do is I can take a sum of this and see how my output is going to look like so I will you have a adder so we will select the add this thing and then put it here and connect these 2 and output I can put it onto the plot.

So, now we will see what is the magnitude of it, so you are seeing this is your output basically what you are getting from the 2 of it so if I want I can add one more sine wave or what I can do is I can do the filtering. So, I will disconnect this one so we will move it a little on the left hand side of it so that I have a place for me to put I can do the FFT magnitude also. So, first we can see FFT it is FFT size is 256 so I can add it here so what I get a FFT magnitude and I can do the plot. So, once I click on the thing so we will see the magnitude plot.

So, you are seeing different plots for the frequencies whatever you can see this, this is going to give you approximately 0.2 pi. So, and then here more this is your draw this thing 0.2 pi and then multiple what you are generating 0.6 to 0.3 is not a multiple of 0.2 so that is the reason why what you will be seeing here approximately 0.94 what the repetition is happening. So, now what we can do is we can create a filter in between and then see that so first I have to define the filter coefficient.

So, here it is going to give me a 0 is always 1 and then if I do not give any part of the thing only b coefficients what I am going to specify so it is going to create me FIR filter so I will give the moving average basically I can save b_1 and then b_2 I can give it as a 0.5. We will see what is the thing is going to happen with the filter coefficients and then we will put the filter basically. So, we will add the filter here and coefficients are given from here and then input is from here and then what we will do is we will connect this to FFT here so now we will see what is the plot I am going to get it magnitude plot.

So, you can see that it has it is a moving average so you have seen that the peaks have been amplified so if I want I can change my this thing pulse width I will keep it as a 20 here for the same as the previous one and I will increase it to this one to 0.4 I can update the thing and see what is the plot I am going to get magnitude plot. So, you will be seeing that different kinds of it what it is generated.

So like this you can create your own this thing either FIR filter or IIR filter whatever filter operations you want to do the thing so what I will do is delete all means the workplace is going to be deleted there is no way you can save these workspaces in these. So, now what we will see some of the examples what we have it here that is all of you know what is a midi basically. So, if I click on the thing so you will be looking at your piano basically.

So, if you want to play and then see what is the music tone it is going to generate so we will only work on the black one, so you will be seeing or different tones also coming along with it? So and you will be seeing what are the frequencies as in our lab using the sine generation we created a (FL). So, you will be seeing same kind of it what it is generated here. So, this is one of the example and if you want to see the in between the magnitude what it will be giving up to pi what you have it so you if you click on the thing

So what is the peak value what you will be seeing it 0.04 pi what it is for the first white one and so next one you can see that what is the displacement you have between the thing. And for this what is the peak you will be getting it point approximately 0 5. So, this is at this distance one twelfth of the distance basically in the piano what they frequency what it will be getting aligned basically so this is one of the examples.

So, we will delete this also now other examples all of you know that are DTMF. So, if I click on the thing so you will be seeing the numbers as in your 4 telephone basically so I can click one of them 1 2 3 or whatever may be the thing so and then even the hash function what we have it. So, this is what different frequencies what you will be getting generated. So, if you want to see what are the things generated so what we will do is we will put FFT here so add the thing with the DTMF and then we will do the plot basically.

So, the magnitude plot what you will be seeing because we know that it has 2 frequencies one is the horizontal frequency the other one is the vertical frequencies for the DTMF. So, you will be seeing this is approximately 0.53 and then this is 0.93 so for one these are the 2 frequencies what is generated so if I click on now 8. So, I will go back and you can check the magnitude what it has got created so you will be seeing that it is x axis 0.63 what you will be seeing the thing the other one is 1.04.

So, you can go to the DTMF so each digit what are the frequencies that get generated so you can map it into that. So, this is how you can play along using your mobile also that is what I wanted to bring in so next is using the PZ placement that is your pole 0 placement you can generate your coefficient and then use it for your application. So, this is a demo what you have it so if I want to add a 0 I can add a 0 or if I want to add pole I can do it or you can reset.

So, this is the demo also I will put it as 0 what I have to add it so I am going to put it my 0 in this position so you are seeing what is the magnitude response I am getting get it. And then now this is the frequency that is face what it is at the bottom what it is showing now add a pole to this. So, what we will do is what kind of frequency you want to have it so I can put it in this way also so what is the thing approximately what I am getting is the high pass filter in this case so by adjusting your zeros and then poles you can fine tune so I can add one more 0 and then I can place it here and then add a pole here I will be fine tuning it so you will be seeing that what is my cut off what how it is varying. So, I want to fine tune the thing so you can place the poles and 0s in whatever way you want. So, if you want to delete if you have not satisfied with whatever the response what it is showing so we can move them.

So, or if you want to put the zeros outside so you will be seeing that what is the phase response is going to happen and then how it is getting moved. So, this is how you can play around by placing your poles and 0s using this PZ demo. So, the next one what we have is a little bit on sound record and then player so what the manual says is you can play from one of the unit and then you can get it in the other unit through your Wi-Fi or whatever way it is connected.

So, if you want to see the frequency response demo so you will be seeing the demo here. So, if I want to have a low pass filter to be built in so this is how it shows that what are the components A and B this IIR filter what it has designed it so these are the values what it has taken to get this response? So, if I want to design a high pass filter so this is my poles and these are my 0 what it is coming in in my system.

And same way you can define the band pass filter so these are what you will be seeing poles and then 0s which is getting done. So, this is one of the things so like this you have a sound player. So, if I want to do the convolution demo I can take the thing convolution demo or you can do your own convolution so the demo will whether I want discrete convolution or a linear convolution what you have seen the thing.

So, if I click on the thing continuous convolution if I want to do it or discrete convolution so we want to have a discrete convolution so it will ask me signal 1 is rectangular and Signal 2 is also rectangular what it has been default what it is chosen so we will see what is the output we are going to have it. So, you will be seeing that this is the output y what you will be getting your convolved signal.

So, if you want to do this thing different one will take a triangular one and the other one also will take a triangular and then save it update the thing so you will be seeing the output is what you will be getting to triangular how you will be representing the output y basically. So, one is I can choose it as rectangular and then save the thing update it so you will be seeing that wherever you have the thing convolved signal what it shows.

So, this way you can play around with the thing and then see whether you are a DSP concept is clear or not so the next one is if you want to design a FIR design you can you do use it or if you want to design IIR filter you can design the thing but the coefficients have to be given from the FIR coefficient values will be given from this block basically. If you want to design a window you can do the window design frequency response you can get the thing and filter coefficients have to be fed into your filter design block basically.

So, we will take up FIR filter design so I have asked for this thing it has chosen in this case Hamming window order is 8 and then the design is low pass and cutoff frequencies it is taken it as 0.2. So, we will update the same thing so these are the coefficients what you have it for the eighth order filter so now what we can do is I can generate this thing signal generator I can use it so we will give it as we will see the rectangular what is the output we are going to get it.

So, I will add this so I will be selecting the filter input from here to here and coefficients what I have taken from here. So, now because I have to see what I have got the thing so what we will put is FFT block in this case so I will add a FFT and then I need a plot so I will connect that. So, now we will see what is the output magnitude what I am going to get so we have designed the low pass filter and then cut off frequency what it has chosen is 0.2.

So, sine wave if you want is so one more plot what you want to see it what is the signal generated one with FFT what we can plot it after the filter we can plot the things. So, for that what it has is a junction what it calls so I can put a junction here. So, from here I will be taking input of this so from here I will be taking one of the input to here so I can directly plot the other one let us I will put a FFT there also.

So from here I will be giving to this and then I want to have one more plot I can select it and then I will be putting I will be connecting. So, now we will see what; is this I will see the magnitude response of this and then after filtering. So, what I am going to get is this one so there is no difference between the thing so what we will do is our signal generator directly instead of putting a FFT if you hold it for a thing you can delete the thing I can directly give it to the plot.

So, you will be seeing that magnitude of it so this is a flat response what you have it correct so when I pass it through the filter or because I am passing through the low pass filter the rectangular wave so you will be seeing that this is what the output is going to be so after filtering so I can you can keep playing with the thing one or 2 more demos what I will be showing you the thing. So, if you want to do up sampling or down sampling I have shown you one of the examples and then if I want to do the inverse FFT both FFT and then IFFT can do the thing.

So, if you want to peak up the peak you can do that. And then if you want to signal to noise ratio you can do it, so the other one is if you want to see the spectrogram of your signal what you can do it. So, there is one provision for long signal generator so if you click on it and then. You if you do the thing what it says is it is a male speaker and then they gain what they have are 1 and frame size 256 frames are there.

And then overlap there is no overlap between the frames and then you will be seeing that current frame is there are 32 frames in it so I can go with the thing so you are seeing that where your voice is going to come so you will be picking up the peak of it. So, you can use that one for your usage in any of the thing so you will be seeing that one of the example you have is a linear predictive coding what you can use it so we have a little demo on the thing there is a Quant LMS demo is also there for your adaptive filter.

So, whatever the quantization and then pole 0 to L s is of what you have the thing so we will take the LMS demo. So, if I click on the thing so you are seeing that initially original filter is value has this and then the estimated filter what you have is this one. So by doing the next this is a default what it has taken so if you give the next step so you will be seeing that it is getting adjusted to these values whatever you had the thing. Step size you can choose between the things so we will give it as 0.1 is our step size.

So, then we will see that how the adaptation is going to happen so I will be pressing on the next step initially it was 0. So, we will so default what it has taken the thing so we will see with 0.1 as a step size it was 0.5 earlier now it is 0.1 So, you are seeing that it is going into the negative. So, what is the thing happening it is not reaching the 1.0 so they are varying much away from the thing.

So, by varying your step size smaller the step size you will be seeing that it is getting adapted and then by making little more you can see that it is oscillating between positive and negative side of the thing. So, this is how what you can look into the thing and then the there is a quantization set up to what this is a demo basically. So, what it has got created so you will be seeing that what it has is a long signal generator and then this is a LPC one coder basically and then quantizer what it has in it and then there is a filter and it will be going for the player.

So, you can see the play sound in your this thing it will be coming out I think sound because it is taking my recording so here you would not be able to hear this sound. So, you will be seeing the long signal here it has been chosen as male speaker so if you want to see a different speakers are there whether you want to have a music you can play with it or a female speaker or noise or sine 1 2.

And then you can have a music and noise combined and then you can pass it through your adaptive filter and then see that your noise is eliminated using your filtering and then you will be getting only the music back. So, these are the play things what we what one has so the other one is quantization if you want to see the thing. So, we will LPC quantization so what it will happen is signal to noise ratio what it is going to give you as the output.

So, here it is a long signal generator same thing what you have is a male speaker and it has 32 frames which will be continuously played so and then from the junction one is signal to noise ratio what you are calculating the other one is you have pass it through the linear predictive coding LPC coding. And then you have the quantization, 2 level quantization what you are doing the thing and then you are doing the filtering from that and giving to signal to noise ratio what we will do is we will see the thing here.

That is in DB what it is going to give you so if you whatever the male voice getting played is going to be showing its signal to noise ratio. So, if I play the thing so you will be seeing continuously changing with it so that is from the sound recorder what you can connect with the signal to noise ratio and then see what is your signal to noise ratio is coming. So, one is the direct signal what you have taken the other one is a noise what you have whatever the output of LPC coding what you have got it.

The difference between the thing that DB what it is showing here so this is how you can use this help files are there in this when you click on the thing so if you want you can go to the information.

(Video Ends: 27:26)

(Refer Slide Time: 27:27)

So, you will be having the IJDSP video demo one can look at it and then you have the DSP with the book with simulations this has to be bought basically. So, you can web page is also available so if you go to the demo so you will be seeing the so it will show you how to go about using it with the demo. What are the things available? And how is it signal generator how you can do it whatever demo have seen the thing.

So, you can go with this demo and then use in your own design basically. So, hopefully you will be enjoying this course so with this thank you very much for listening to me, so you can go up and then play around with your mobile phone with the block level simulations also thank you.