Real-Time Digital Signal Processing
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Module No # 09
Lecture No # 41
Graphic Equalizer

Nameste, welcome back to real time digital signal processing course today we will discuss about graphic equalizer. So we had seen the demo but again we will be demoing after completing the theory.

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So in the previous class we saw that how to do echo cancellation and even generation in the previous class what we had seen the thing why we have to cancel the echoes? So if it is required as a hobby you can generate the echoes and if you are not, want the echoes which are generated by different sources so you can do the cancellation.

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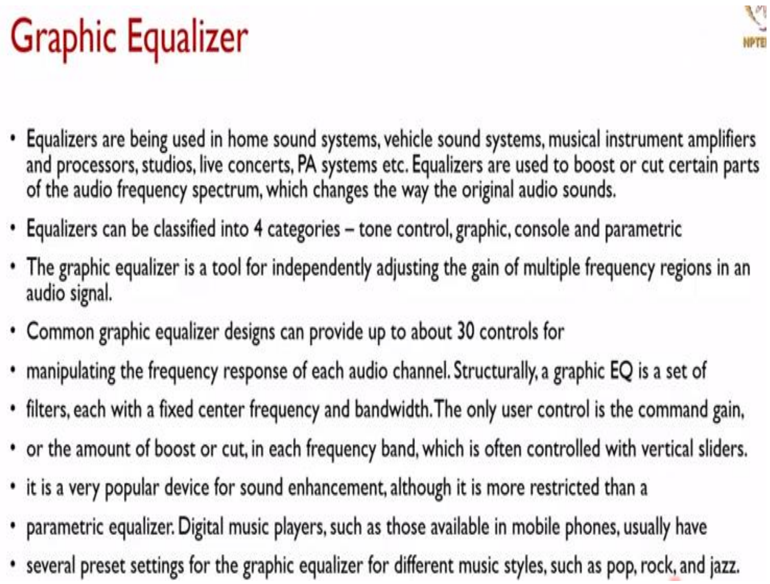

So today what we will see is the graphic equalizer. So why do we need equalizers? So we know that we use in home sound systems, vehicle sound systems also, in musical instruments amplifiers and processors, even in studios and even live concerts including our public address systems and then you go on listing where we will be using the equalizers. So it is used to boost or cut certain parts of the audio frequency spectrum, so we change the way of the original audio sounds.

So equalizers can be classified into 4 categories one is tone control the other one is graphic so console and then parametric. So it is the graphic equalizer is a tool for independently adjusting the gain of multiple frequency regions in an audio signal. So the common graphic equalizer designs can provide up to about 30 controls for manipulating the frequency response of each audio channel.

And structurally a graphic equalizer is set of filters, so if you are designing in the digital domain each with a fixed center frequency and bandwidth. The only user control is the command gain, what it is provided or the amount or boost or cut, in each frequency band which is often controlled with vertical sliders. So it is a very popular device for sound enhancement although it is more restricted than a parametric equalizer.

Digital music players such as those available in mobile phones, because most of us use the thing, usually have several preset settings for the graphic equalizer for different music styles, such as pop, rock and then jazz. So one can investigate what are the things you need it for your music players.

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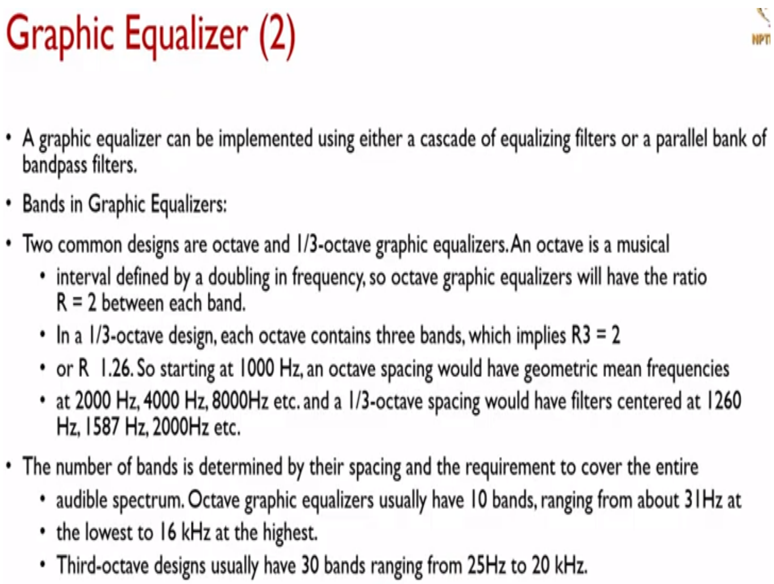

So continuing with the thing so this can be implemented using either a cascade of equalizing filters or a parallel bank of band pass filters. So we have discussed about the filters one of them you can consider in the cascade form or in the parallel form. Bands in the graphic equalizers are 2 common designs are octave and one third octave graphic equalizers what we have it. So an octave is a musical interval which is defined by doubling in frequency, so octave graphic equalizers will have the ratio that is we call it as R = 2 between each band.

So in one third octave design so each octave contains 3 bands which implies that R3 = 2 or R is given as 1.26. So starting at 1000 hertz that is 1 Kilo hertz an octave spacing would have geometric mean frequencies at 2 kilo hertz, 4 Kilo hertz, 8 and so on. And then one third octave spacing would have filters centered at 1260 hertz, 1587 hertz, 2000 hertz and etc. to call you back we generated a musical note.

So if you remember that complete thing what we call it as one octave so we started with the base frequency 240 hertz there and we went up to 480 hertz and different base what you can select. I think music lovers will be knowing c c sharp and whatever their tone is going to be you can set it and then you can go on adjusting your frequencies. So the number of bands is going to be determined by their spacing and the requirement to cover the entire audible spectrum what it is going to be considered.

So what we call it is octave graphic equalizer usually have 10 bands ranging from about 31 Hertz at the lowest to 16 Kilo Hertz at the highest. So the third octave designs usually have 30 bands ranging from 25 Hertz to 20 Kilo Hertz. So you will be seeing that how the difference of frequencies or bands what we have to select in your design part of it.

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Octave Frequency Bands

Preferred octave frequency bands according to the ISO standard

Lower frequency f_l (Hz)	Geometric mean frequency f_c (Hz)	Upper frequency f_u (Hz)
22	31.5	44
44	63	88
88	125	177
177	250	355
355	500	710
710	1,000	1,420
1,420	2,000	2,840
2,840	4,000	5,680
5,680	8,000	11,360
11,360	16,000	22,720

So to show that octave frequency bands you can go to net and then you can see, search for the thing you will be getting it. Here whatever the preferred octave frequency bands according to the ISO standard what it has been listed? So you will be seeing that lower frequency f_l Hertz is 22 to 11360 that is 11.360 Kilo Hertz what it is going. So in the geometric mean frequency that is f_c which is the cut off frequency center frequency what we call it. So which varies between 31.5 Hertz to 16 Kilo Hertz in this? And the upper frequency one can have is from 44 Hertz to 22.72 Kilo Hertz.

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Octave Frequency Bands (2)

ISO standard for 1/3 octave frequency bands

f_l (Hz)	f_c (Hz)	f_u (Hz)	f_l (Hz)	f_c (Hz)	f_u (Hz)
22.4	**25**	28.2	708	**800**	891
28.2	**31.5**	35.5	891	**1,000**	1,122
35.5	**40**	44.7	1,122	**1,250**	1,413
44.7	**50**	56.2	1,413	**1,600**	1,778
56.2	**63**	70.8	1,778	**2,000**	2,239
70.8	**80**	89.1	2,239	**2,500**	2,818
89.1	**100**	112	2,818	**3,150**	3,548
112	**125**	141	3,548	**4,000**	4,467
141	**160**	178	4,467	**5,000**	5,623
178	**200**	224	5,623	**6,300**	7,079
224	**250**	282	7,079	**8,000**	8,913
282	**315**	355	8,913	**10,000**	11,220
355	**400**	447	11,220	**12,500**	14,130
447	**500**	562	14,130	**16,000**	17,780
562	**630**	708	17,780	**20,000**	22,390

So this is the table shows and then that is for complete one octave and if you see for one third octave frequency bands the ISO standard is given as this. So what you have is f_l and then you will be seeing little variation with the thing it varies from 22.4 to 562 what you can see it. And your center is frequency somewhere from 25 Hertz to 630 Hertz what it goes. So what it says is bold font represents the center frequencies in this case.

So because we said it is 30 bands what it has. So this is represented in one column. So in the next column you will be seeing the upper frequencies what it has it and then from the 500 and this thing 62 it goes from to 708 and then you will be seeing up to 17.78 Kilo Hertz. So totally you have 30 bands in it. And these are the center frequencies what you will be providing for them and these are the upper frequencies.

So for these frequencies, these are the center and these are the upper frequency and for this set you will be seeing this as center frequency then upper frequency. So this is how synthetically you can generate your equalizers.

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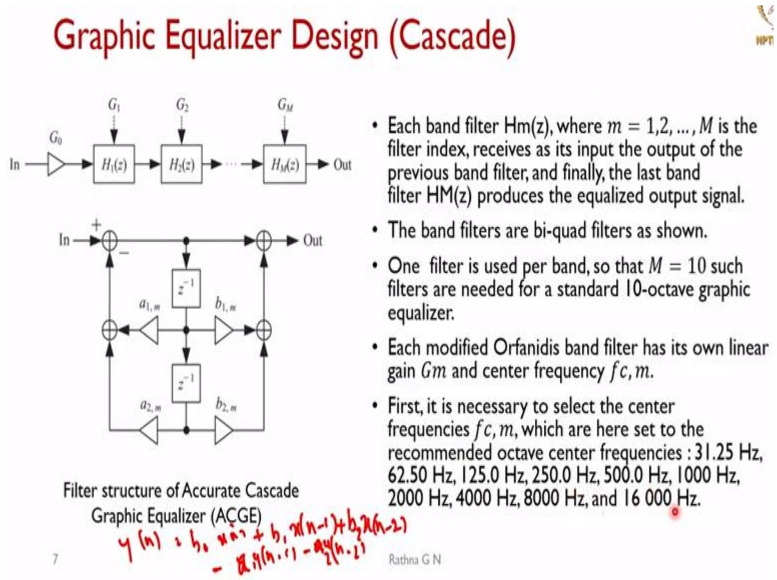

So how to go about designing it, so in this case first will consider the cascade part of it. So if you want you can go to theory and then see how the design is going to be done. So in this case we will consider we are going to give, this is the input audio what it going to be and this is the overall G0 gain for the audio what we can give it. And then for each band here it is 3 band or M bands what you are going to decide.

It can be one octave or one third octave what you can select. So if it is one octave it is 10 equalizers what we have it and one third you know that 30 what you need it equalizers basically and for each one the gains can be varying from G1, G2 to G M basically. So what it say is M is varying between 1 to m is the filter index receives at input. The output of the previous band filter and finally the last band we call it as HM of z produces the equalized output signal.

So this is what; how you provide octave basically, so the band filters are bi-quad filters I think it should be something is ringing in your mind so when we use bi-quad basically in IIR filters. So as you will be seeing that it is a bi-quad section what we have choosen this is again input and this is plus and minus this is the feedback word path and this is the feed forward path. So if you want to have b_0 you can give the thing most of the time it is going to be 1 that is why it has been removed.

And then this is $b_{1,m}$ because number of stages $b_{1,m}$ m will be varying between l to M. so many stages what you can do concatenation basically. That is cascading what you can do this is a b_2

co-efficient comma m and this $a_{2,m}$ and this $a_{1,m}$. So you will be seeing this is our output after adding it and then because we know that our feedback coefficients are negative. That is the reason why what you have put it as negative in this and these 2 getting added and then after that it is going to be subtracted. So if you remember the equation.

So what we have it is I will write here $y(n) = b_0x(n) + b_1x(n-1) + b_2x(n-2) - y_1(n-1) - y_2(n-2)$ so this is a_1 and this is $a_2y(n-1) - y(n-2)$. So this is the equation what we have it for the bi-quad section. So a_1 and a_2 are the co-efficients for the poles basically and then b_0 b_1 b_2 are the zeros. So this is what our bi-quad equations so which is written in this way and then $z\ -\ 1$ will be in zee domain.

The delay which is provided for x and then our y what we see it. So what we can do is one filter is used for per band what it says so that $M = 10$ such filters are needed for a standard 10 octave graphic equalizer. So each modified filters which is named with Orfanidis band filter has its own linear gain Gm what we call it and center frequency fc, m for each this thing what it is defined. So now coming to the think first it is necessity to the select center frequencies f c,m which are here set to the recommended octave center frequencies what it is given with a previous slide what you have seen the thing that is, these are the center frequencies mean frequencies what we have it for the 10 band same thing what it is going to be considered here in the case. So, next one is 62.5 and then 125 Hertz and then so on up to 16 Kilo Hertz. So 10 band center frequencies what it has been chosen.

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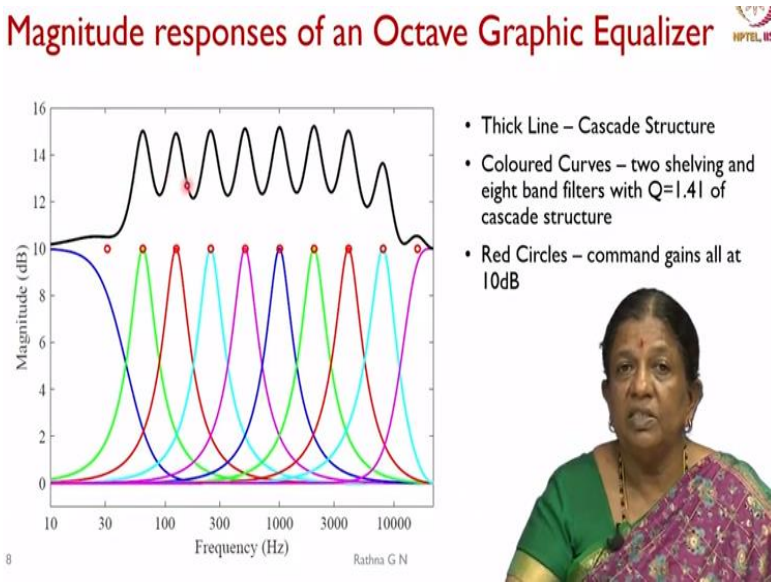

So now will we see what are the magnitude responses of an octave graphic equalizer? So you will be seeing that the thick line what it shows is the cascade structure so you are seeing the this is the input and this is the output. And this coloured curves what it shows is a 2 shelving and 8 band filters what it has been considered here. So you will be seeing that this is a low pass filter afer that what we have is a band pass filters.

So each one represents the center frequency of each one of them. So we have it 1, 2, 3, 4, 5, 6, 7, 8 9 and then 10. So you have what it shows is 8 band filters with what we say Q=1.41 of the cascade structure. So what it says is red circles command gains all at the 10db as we can see that all of them are at 10db and this is the frequency in Hertz what we are going so that is whatever in the previous one we had said that 31.5,62.50.

So we will be seeing they are the center frequencies what it has been chosen along the x axis and the y axis will give us the magnitude of it. So the cascaded what it shows in the thick line here.

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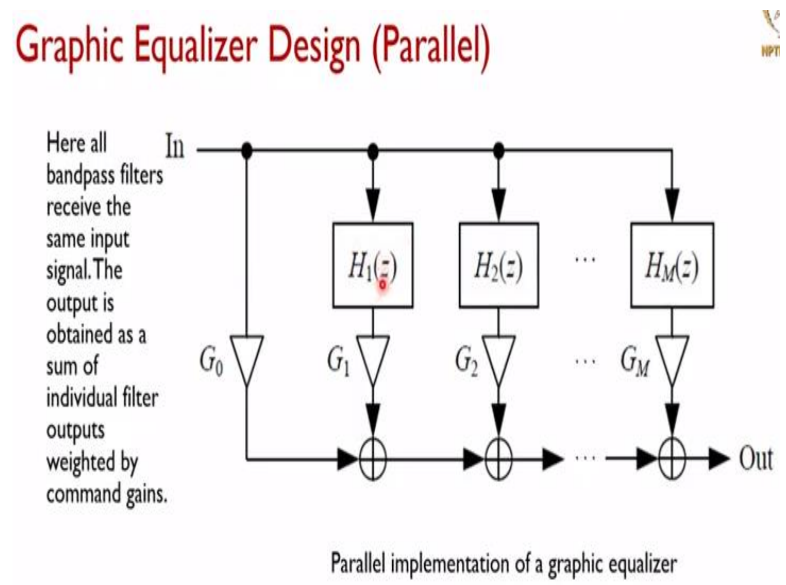

Now how we can design the same thing in parallel that is what the figure shows. So we will be having the overall gain G_0 here and then each one will have its own impulse response $H_1(z)$ and they will be this thing multiplied with their gains in parallel. And then you will be seeing that each one is getting added and then what will be getting get out. So what it says is here all bandpass filters receive the same input signal.

And the output is obtained as a sum of indivdual filter outputs that they are weighted by the command gains. So, one has to think now whether we want to have cascade graphic equalizer as it is shown here or the parallel. Can you think of it because we have seen even in ir filter design we said for our hardware because we are using the dsp processor or if you want to implement in FPGA.

So we said that multiplication overflow is taken care of where as the addition we are going to have a overflow we need so we have to check every time if there is output is overflowing then we have to scale them and then will be taking care of all the overflows. So it is better to go with cascade section in hardware than the parallel structure. Although we have both the things so if you are able to take care of whatever the overflow which is going to come from the addition in the input itself.

Then we can use the parallel section because we know that this is much faster compared to our cascade section, what I will put it as this is parallelism what I can use it here. There I had to use the pipelining in between I had to put a what I will say is a buffer basically or a delay unit. So that each one will be pipeline structure what I can use. So you will be recollecting your pipelining and parallelism what we used in the theory here and then can go for implementation of it.

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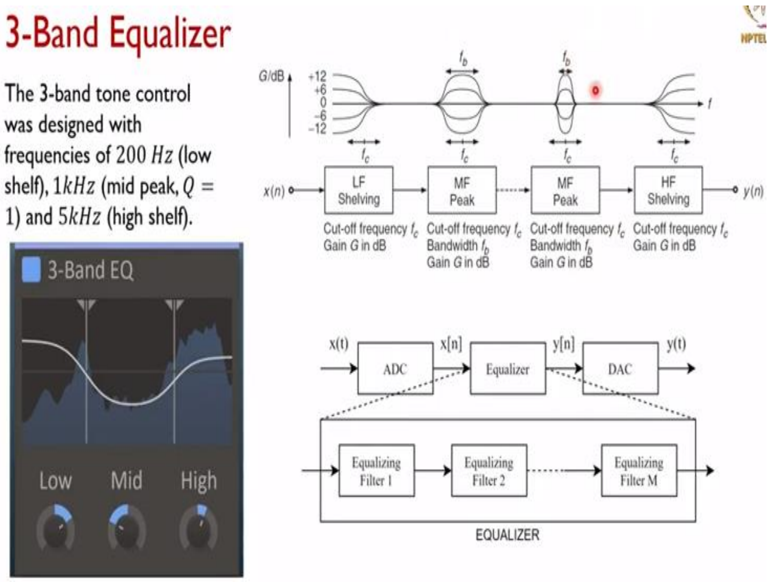

So the next is will take up as an example one can design a 3-band equalizer. So it depends on some of that books gives that we can go with the 3-band is enough that is what you will be having tone control was designed with frequencies of that is 200 Hertz that is low shelf what you will be considering it. And 1 Kilo Hertz mid peak and then here Q is taken as equal to 1 and 5 Kilo Hertz what it says is this is the high shelf. So with this also you can design your graphic equalizer.

So you will be seeing that what is it so the dB what it is shown from here that is -12 to 12 dB what you will be selecting it on different spaces and then this is your f_c is your cut-off frequency and then your you call it as low filter shelving what you have it. And here cut-off frequency f_c gain G in dB what you will be providing it to the system. And then the next one this is a low pass filter as we know and here it is a bandpass filter that is the middle frequency what we can call it as peak that is cut-off frequency f_c bandwidth or f_b gain G in dB what you will be taking it.

So this our cut-off frequency and this is the frequency range fd that is passband region what you will be selecting it. And the next one so on, what you can design and next is the your this thing multiple middle frequencies what you can see and then their peak. And here your cut-off frequency is what you will be specifying with their gain as well as your bandpass of the structure what it is shown here f_d.

And in the last what you will have is does a highpass filter, so which is HF which is given with cut-off frequency as ususl f_c and it says high filter shelving the last stage. And cut-off frequency is fc and its gain is G in dB what it is given. So whatever dB you want to select you can select them and $y(n)$ is your output. So how you can decide design the thing all of us know that audio file is given in the time domain.

So $x(t)$ t is our input which is coming from the, it can be from the mic or audio system. So then we have to convert into digital domain that is will be using ADC depending on number of bits what we want to we can select ADC and after converting into digital domain will be getting $x(n)$ as our input which will be passing it through the equalizer and $y(n)$ is our output. Then what is it? So if i want to convert this DAC to audio that is our time domain then I have to use the DAC. Then $y(t)$ will be the output.

But I do not want to use this ADC and DAC but I want to see that whatever I have stored it. Whether I can provide the equalizing filters design for this equalizer filter 1 2 and then M if it is whatever band yor will be selecting it. So that is what the cascade section what you are seeing it which goes into your equalizer. So now you will be seeing that in the market a 3-band equalizer is shown this way.

So what is it? You have low frequency knob which you can adjust it. And then this is the mid frequency what you have it or middle one so different one what you can select here one of it lower high whatever fb bandwidth what you want to have it. And this is the high frequency what you will be selecting it. So you are seeing this is our low-pass fiter and this is what it is shown in the inverse of it so that is bandstop what you are seeing it.

And whatever frequecies you want to pass it so you can put it as a bandpass filter and then other one is the high frequency filter what you are having it. And some of the frequencies how you can adjust it what it is shown in this blue colour. So this is the commercial which have taken it from the net and am showing you how it can be designed.

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So this completes our adaptive filter and that filter applications what we have considered for bandpass filter, lowpass filter and highpass filter, how it can be used in scrambling, echo generation and cancelling and then in the equalizers. So in the next class we will be discussing about the speech coding techniques that is available in the literature and will be considering one of the example how we can design both in Matlab and then in code composer studio. So thank you, happy learning for this material and then will meet in the next class. Thank you.