Biomedical Signal Processing
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Lecture – 10
Artifact Removal (Contd.)

In the last session, we are looking into the derivative based filter. So, this is the simple form.

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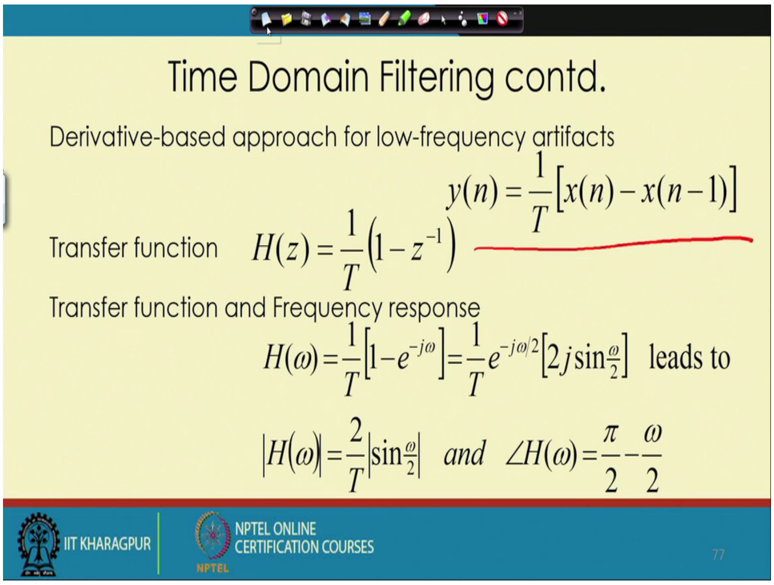

And we have seen that the corresponding that the corresponding the transfer function which is a high pass filter.

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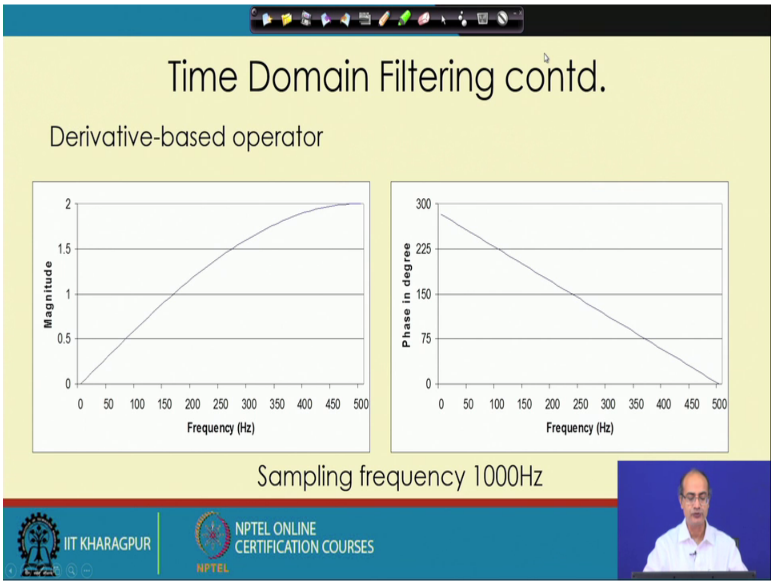

And this is very good in removing the low frequency artifact. But at the same time it creates one problem that if there is any high frequency noise it gives a tremendous boost, which is not a desirable characteristics.

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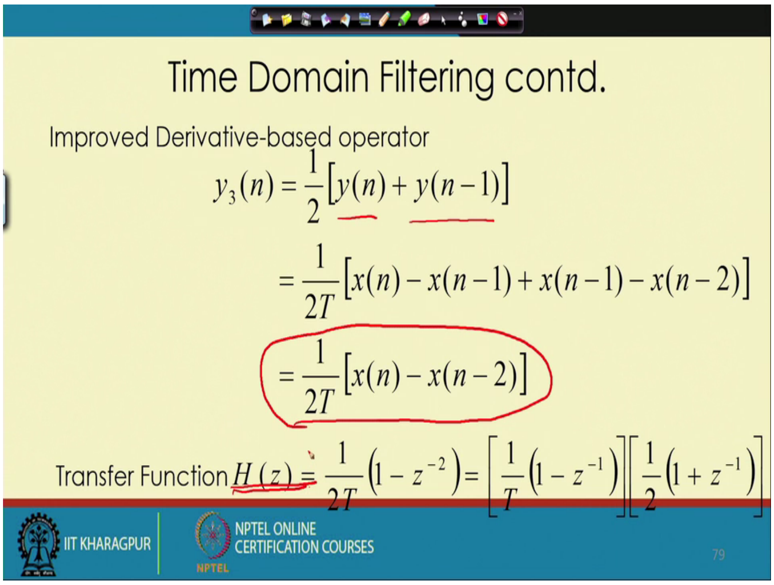

So, the derivative based filter it need some improvement of modification and here one such modification is shown. So, what it is done that simple derivative based filter that has taken to a second stage. So, that has taken as an input and we have taken the two the lags have present and the just passed that output of that filter and that these two are averaged. So, if we replace y n that can y n minus 1, we can do the simple computation and we get this is the form, this is the form of new modified or improved the derivative based filter. And the corresponding transfer function is given here that we get here the transfer function.

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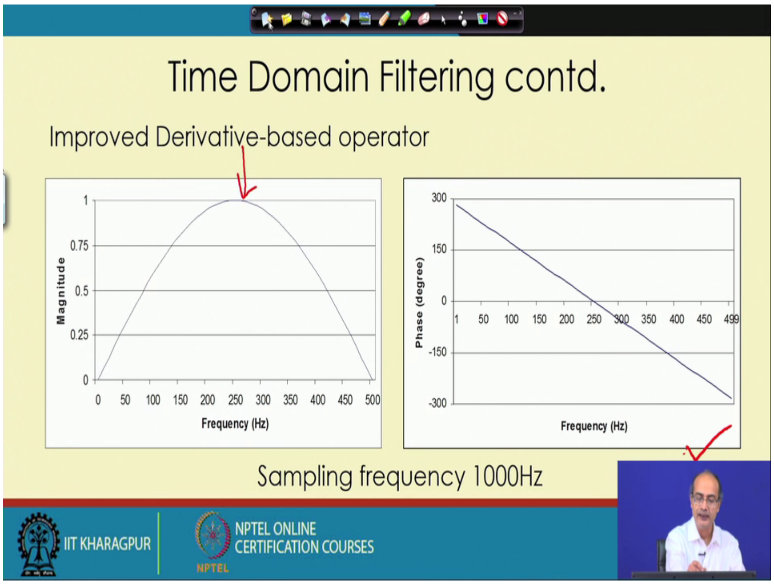

Now, let us see that how does that the magnitude and the frequency response they vary.

We see in this case the frequency response is better it is speaking in the middle here and it is 0 in both the end. So, it will help to get rid of the low frequency noise that is like based line wandering, at the same time if there is very high frequency term that will also get suppressed or at least it will not be magnified by this improve derivative filter and at the same time it has preserve the linear phase here. So, it has preserved that linear phase, so that is the good thing and that we get that this filter is much better for the real life use.

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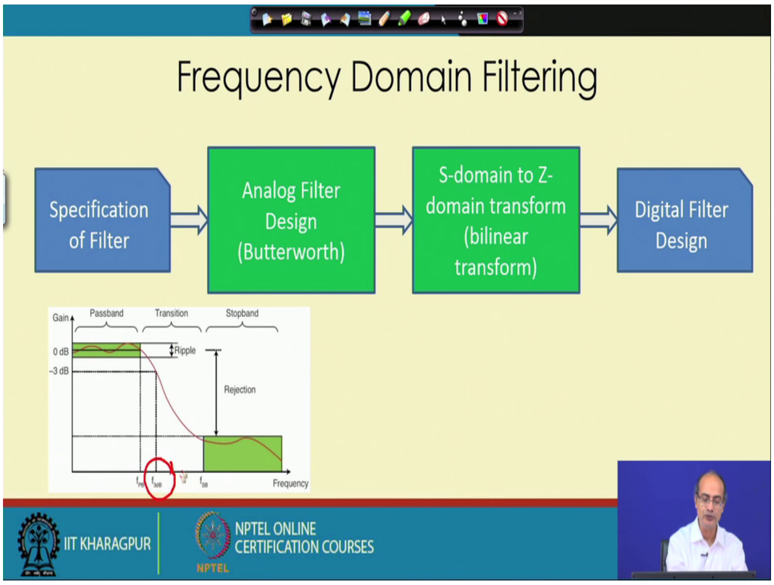

Next, we look at will just pass through the frequency domain the filtering the basic technique, for the frequency domain filtering that this primary we dependent on the analogue filter design that analogue filter design there are so many techniques and they are so matured for digital filter design what we do from the specification that first the is domain filter that is analogue filter is designed and then using some a's domain to z domain transformation like bilinear transform. Then it is actually map to a digital filter.

And how this specifications are given? The first thing we need to give that is the cutoff frequency, sometimes it is given that three (Refer Time: 04:00) bandwidth that this is the cutoff frequency of this filter. So, this is the primary thing this is the minimum we need to give for a low pass filter for a high pass filter also it would be if you just may get it 1 minus the low pass filter we get the high pass filter. So, they are also we need to give this cutoff frequency.

Next we need to give that what is the amount of attenuation or the difference between the pass band and then that stop band. So, what we have taken the maximum of the stop band response and the average of that pass band response we take the, this is a difference this is the rejection or we will tell that so many device attenuation need to be there from pass band to the stop band.

Next some people would like that it should be close to the ideal filter; that means, there should not be any change in the pass band or any undulation should not be there. So, they try to actually control that how much ripple would be actually this is the undulations what is there it is called the ripple what would be the maximum allowed ripple would be there so that is another parameter that comes. And if you want to give further then the people may specify that how sharp would be this transition, so that is given by that what would be the transition band. The sharper the transition we need the transition band should be smaller.

Now, because the this analogue filter design are covered in a different subject will not get into the deep of it; however, we will just mention it here for the purpose of actually that practice because sometimes we may have to use this filters for that attenuation of the signal this artifacts and we will make use of them to get rid of the different kind of noise like high frequency noise.

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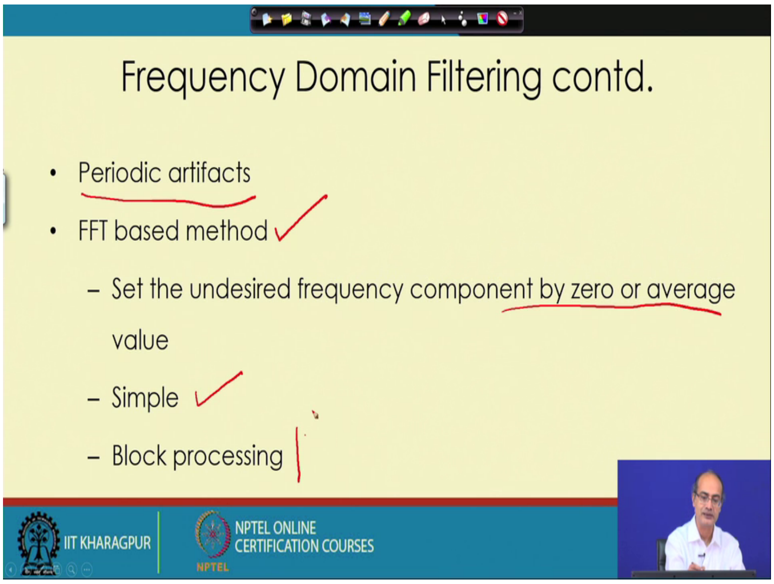

So, now we go for other kind of frequency domain filter which are not covered in those regular courses. The first thing that if we look at the filters that frequency domain filter here it is primarily act actually sensitive or effective for the periodic artifacts. So, what could be the example of periodic artifact? That first thing which comes in my mind that is that power frequency noise which is completely periodic is nature and those kind of artifact can be removed by the frequency domain filter very effectively. So, the simplest way to do that is using the effective based method and in the effective based method what is done we take a block of signal and that block of signal is transformed to the frequency domain taking FFT.

And then as you get the different frequency component we actually set those values of the frequency component which are undesired we said them to 0 or we if you feel that it should not go to 0 that will change the spectrum then we take their nearby values and we replace it by the average of the neighboring frequency component. And that is how that the elimination of the noise takes place here and then we do the inverse transform and we get back the time domain signal.

Now, the biggest advantage of it is, it is simple, but that one actually special characteristics we get that this is actually block processing technique or it needs for a block of signal to do this operation. So, though it is simple and that makes actually one difference that unless we have a block of signal we cannot actually process it and give the output and that makes it difficult for real time applications or unsuitable for the real time applications. So, will look for some alternative that which can keep the benefits of that simplicity and effectiveness of the frequency domain technique, but yet they are not block processing technique.

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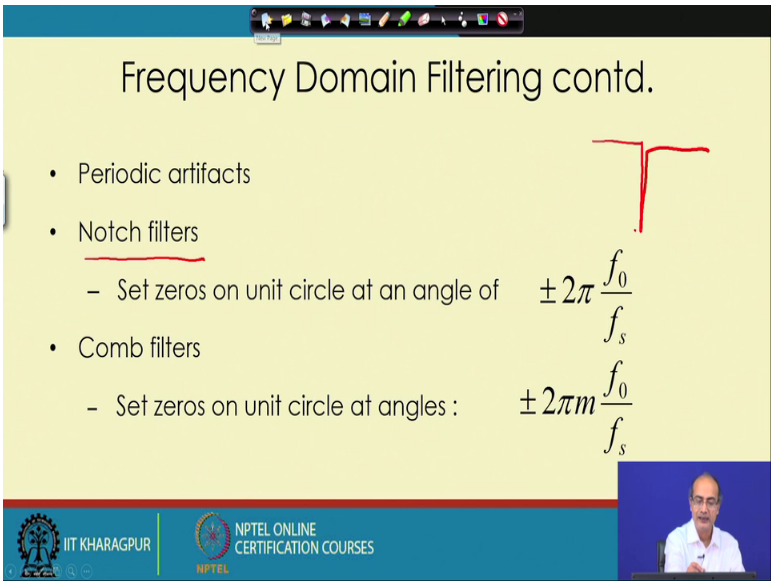

So, we look at a particular filter that is called notch filter. Notch filtered means that it will have a 0 at a particular frequency. So, this is the notch. So, here what we do we set some zeros on the unit circle at a particular frequency and we are taking actually a conjugate pole pairing in this case the primary reason is that that it will give us a real filter. If you do not take conjugate pole pair then it will not give us the real filter, but our signals are real, so we would prefer to use a real filter to reduce the computation.

And if we have multiple harmonics; that means, the periodic artifact in not just present in one frequency, but at multiple harmonics then we go for comb filter. So, it sets again 0s, but at multiple angles at pre frequencies. So, that is the difference with the north filter.

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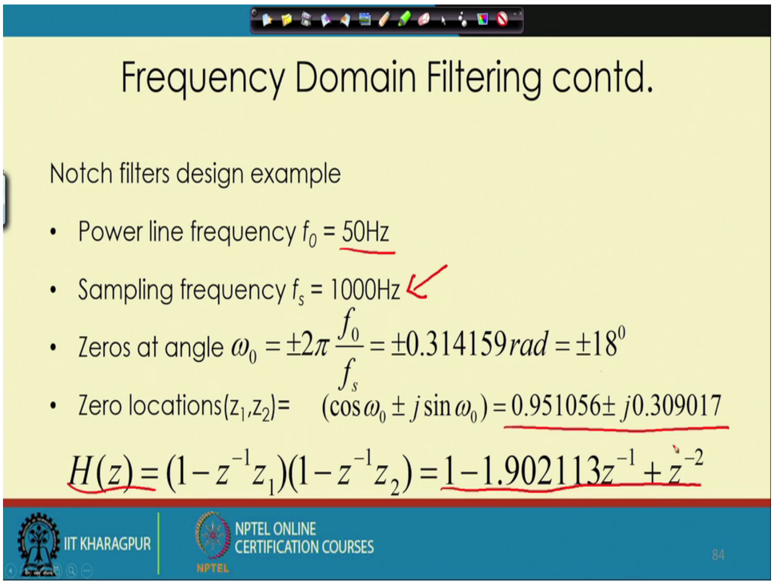

So, let us look at an example. In our case that we have a power line frequency that is 50 hertz. So, for this 50 hertz power line frequency and our sampling frequency is 1000 hertz here that to eliminate that what we need to do we need to first apply the, that the two 0s at plus minus 2 pi f 0 by f s. So, we get the angle at radian and that can be converted to the degree. So, we get plus minus 18 degree if we have two 0 that can help us to get rid of the 50 hertz component of the signal.

So, using this information we get the 0 locations the cosine omega 0 plus minus j sin omega 0 and we get here the location of the 0s and using that information we can reconstruct the exit which comes as a that second order filter which is symmetric.

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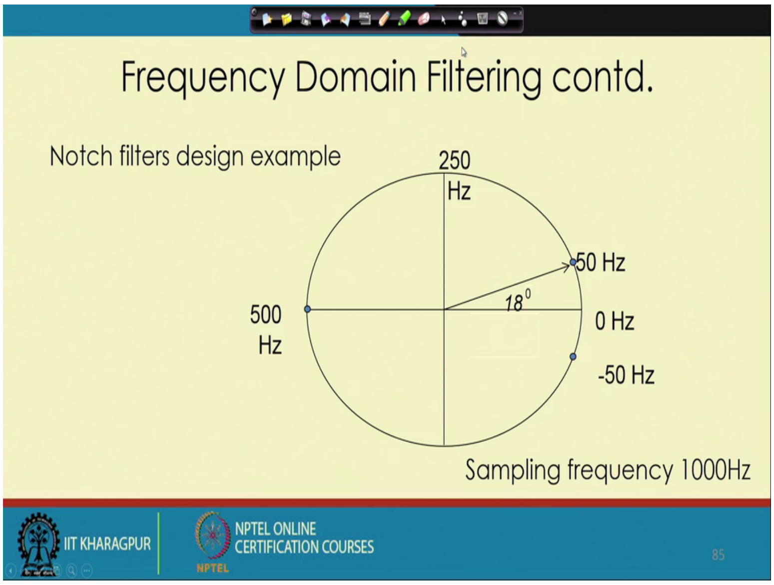

Here we see that our actually that the angle how it is shown that we are getting the plus minus 18 degree angle we are getting the output that the 0s are there in the pole 0 plot and with these that we are able to eliminate the 50 hertz that signal which is actually a power harm or a noise for the biomedical signal.

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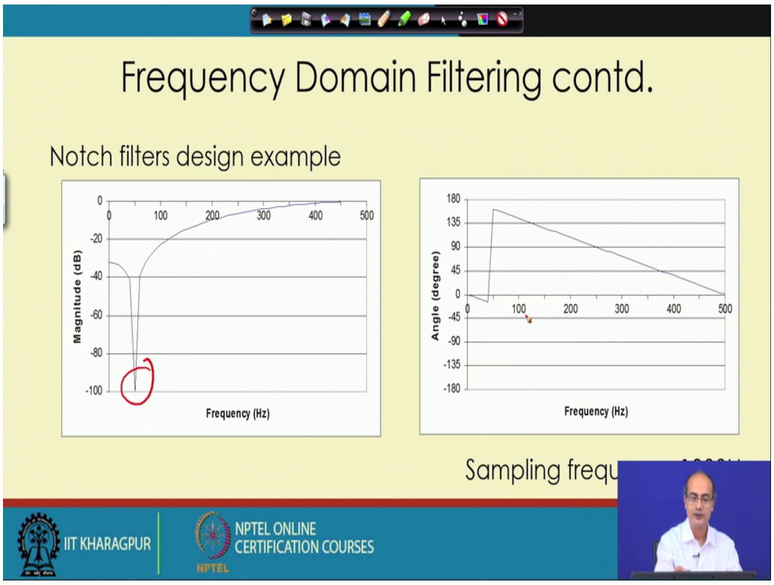

So, here next we show that frequency domain characteristics of that 50 hertz that notch filter. So, here we get that there is a notch here at 50 hertz which is very sharp and we get that corresponding point we have a discontinuity in the phase. So, the phase is as the weights are symmetric here the phase is linear, but it is piecewise linear it is not linear from for the entire length 0 to 500 it is linear from 0 to 50 and again 50 to 500, there is a discontinuity at the point 50 hertz. So, that is how we get the notch filter here.

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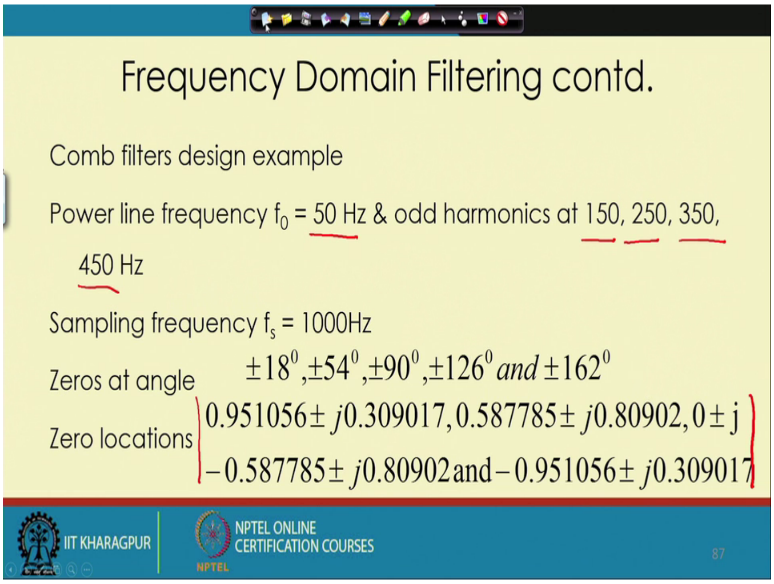

Next, we look for the comb filter. The primary reason for the comb filter is that the power frequency that we get it is not actually a single tone or monotone signal usually there are some harmonics there and this harmonics are created because of the actually that the transformers which are operating at the that saturation region. So, because of them some low that that odd harmonics they are actually injected in the power line signal.

So, we get along with our, that 50 hertz power line signal we get third harmonic at 150 hertz, fifth harmonic, seventh harmonic, ninth harmonic. And as you go for higher and higher harmonic we see the strength actually goes actually down at every step. So, if you take 2 to 3 harmonics that would be actually sufficient here we have taken four harmonics along with that primary frequency. So, the sampling frequency here again remains with the same that is 1000 hertz, corresponding to that, we know for 50 hertz we need to have the 0s at 18 degree. So, for the third, in third harmonic, 3 into 18 that is 54 degree we need to have the 0s, then at we need to have them for fifth harmonic at plus minus 90 degree and for seventh harmonic we need to have at 126 degree plus minus 126 and for nineth harmonic we need to have it plus minus 162 degree.

So, corresponding to that these angles we can get the 0 locations which are given here. That is cosine omega plus minus j sin omega that is the formula we get the location of the 0s or we can says 0 pairs because again we need to get actually the real filter to efficiently clean a real signal.

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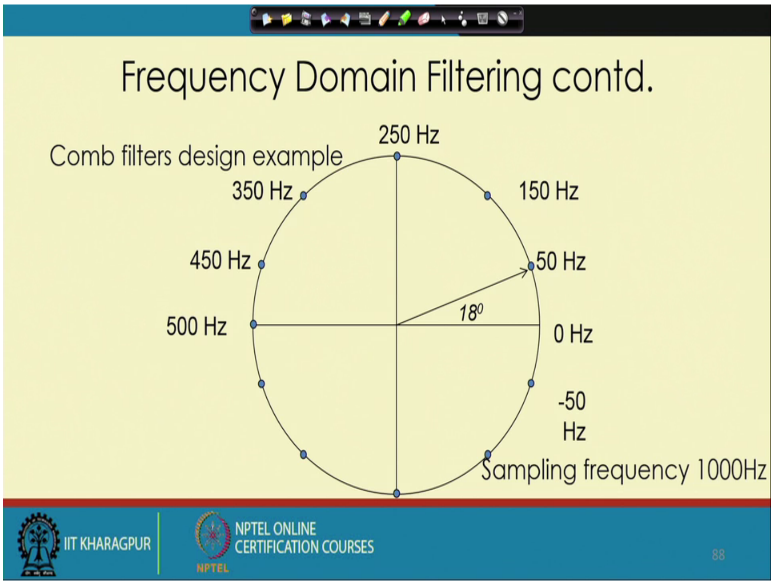

So, here we show that that at different that angles that from 18 degree then at 54 degree, then 90 degree, so that for 50 hertz, 150 hertz, 350 hertz, 450 and that way that all the 0s and corresponding that the conjugate pairs of the 0s, they are given in that pole 0 diagram. So, that is how that can we actually show this that frequency domain filtering and the notch filter and we complete this session.

Thank you.