Biomedical Signal Processing
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Lecture - 31
Waveform Analysis (Contd.)

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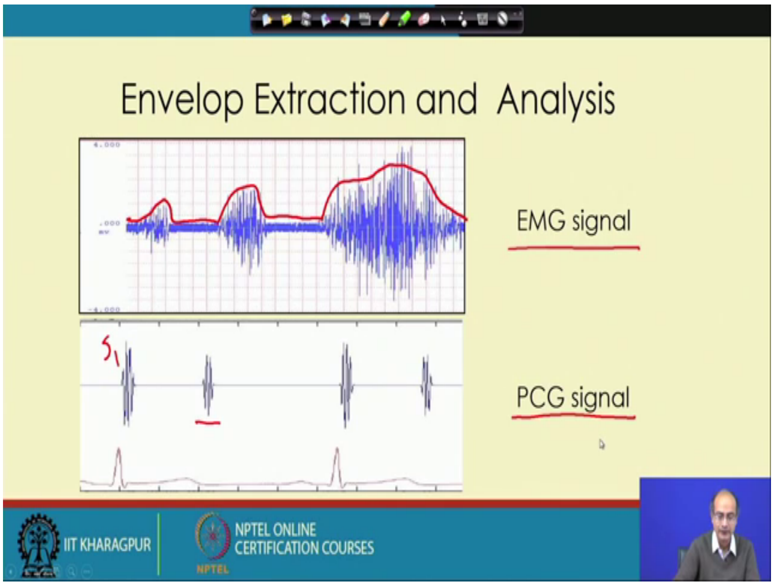

So, next we look at the envelop extraction. So, here we have taken an EMG signal. In case of EMG signal, we know that whenever we exert some actually force through the muscles. So, they are excited and that give raise to some signal and we get the noise like kind of actually pattern and if we look at that one measure of the volusion that how much actually, we are exerting that could be given as the this envelop of the signal. So, that could give us a measure that how much actually effort, we have given. Now if we look at that EMG signal, the EMG signal the main components are within say 200 hertz, some components are they are going up to thousand hertz, but if you look at the envelop the maximum frequency could be of 20 hertz ok.

So, this is one situation we get let us look at one more such example that here, we have taken the PCG signal. In case of PCG signal, along with the R wave; what we get the ventricles, they are getting contracted with the QRS complex and the blood which are accumulated in the ventricles now they start coming out of the ventricle they are getting into the arteries.

So, there would be some noise or vibrations created out of the turbulent motion and the vulgar forced to actually open first for that. So, all these together the opening of the valve the rushing out of the plug out of the ventricle to arteries this causing the turbulence they give raise to this S 1 wave and after that when actually the ventricles they are relaxing then we have that the other wave that is the S T wave where the valves are actually getting closed because the ventricles are relaxing.

So, at that point if the valves does not close, there could be back flow, there is to make sure there is no backflow that these part that there is a that closing. So, that gives raise to the signal S 2.

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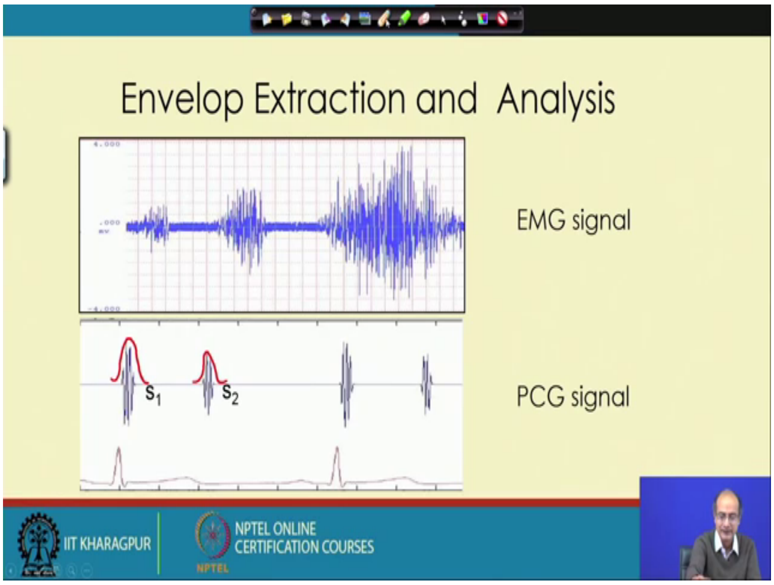

So, S 1 and S 2 these 2 are very important to know the health of the arteries at that point and whether the valves are properly closing or not. So, one measure that how much noise it is creating can again be the some kind of envelop of the signal and again what we get compared to that the frequency of the S 1 and S 2 wave, these envelop should be much more low frequency. So, how actually we can extract them for our purpose that comes as a challenge.

So let us see that how these point was actually taken care.

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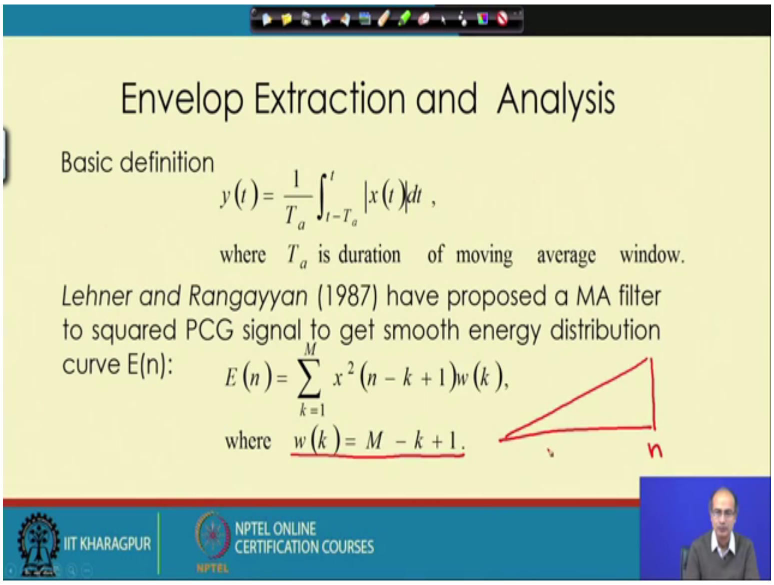

So, from the basic definition that what it is done that we have that first we should take care of the name that who has proposed this that Lehner and Rangayyan in 87. So, what they suggested they suggested to make use of these the basic definition of envelop which suggest we should take the absolute value of the signal intricate over the time and take the average to get actually envelop. So, what they suggested that we should take a MA filter for averaging and to make it positive for the PCG signal that we should take square.

So, we should take a MA filter on the squared value of the our PCG signal. So, that was the suggestion proposed by the Lehner and the Rangayyan. So, they told about a small MA filter which can accumulate the energy which is positive because the purpose of taking the absolute value or the energy, if the same that because the signal is oscillatory that if we do not force it to make positive they will cancel their contribution will cancel each other when we accumulate them. So, either we have to take the absolute value or we need to take the square.

And Lehner and rangayyan they took a strategy that we should take the square and the weight actually we have to get it is not that simple MA filter, they have taken a window which is a triangular window. So, the latest point that n the value would be maximum and then it should actually go to actually 0. So, it would be having a tapper 1; that means, the most recent actually values or most recent points, they will contribute the most and the one which are far in the first will contribute less, but contribution of multiple points are equate to do the low pass filter without that we will just gate the control of the signal itself.

So we do not want that we want that envelop and for that this strategy is needed.

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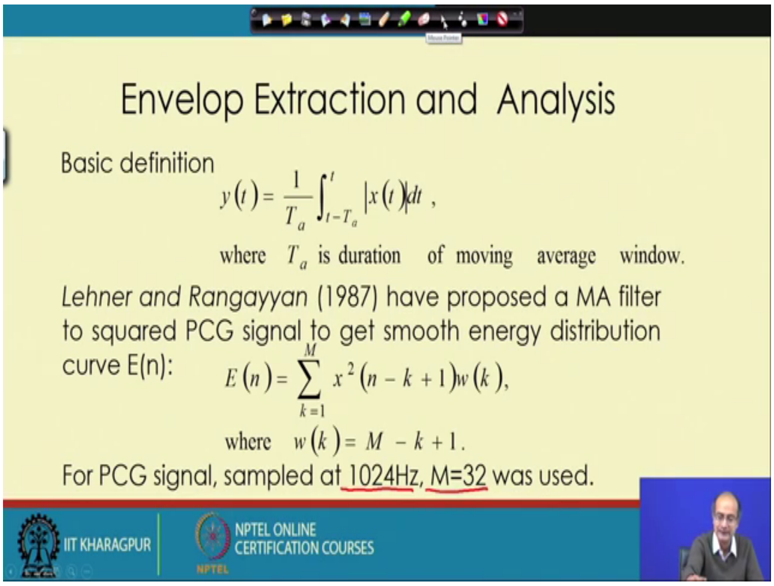

So, for the PCG signal under consideration it was actually sample that that you can say about one kilo hertz 1024 hertz and they found that M equal to 32 is sufficient for that. So, with that that Lehner and Rangayyan, they proposed first to actually get the envelop and make use of that as an energy of the activity or to measure that how much actually energy is present in S 1 and S 2.

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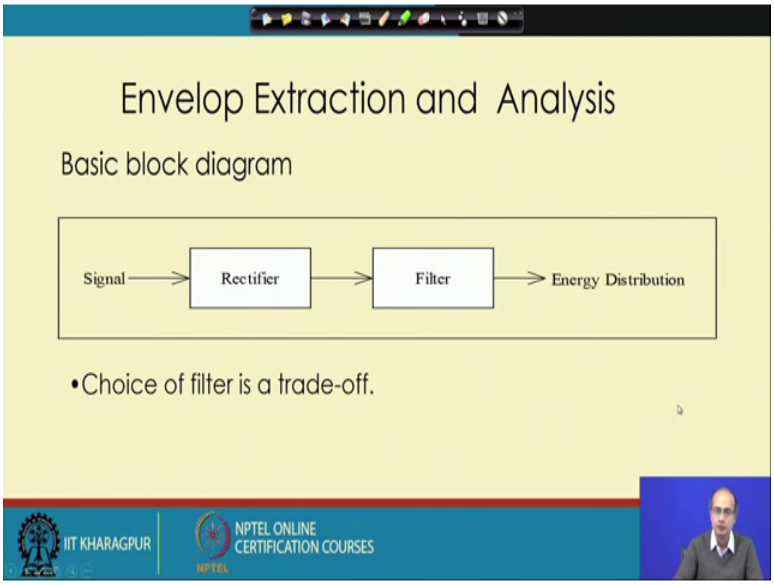

Now, let us proceed to see that what is the basic strategy is here first we should have a rectifier here because the signals are oscillatory. So, this rectifier it will take care of the fact that the signal becomes positive and then we should have a filter the filter is to take care of the refill that after filtering we know whether we are doing that that half way filtering or full way filtering in that case that there would be refill in the signal. So, we want to remove that refill and then that the low pass actually that filter, it will pass on only the that the energy distribution what is changing with the time that is the envelop we are looking at.

And in this valve; however, the choice of the filter is a trade off, what we mean by that that we want to get rid of the riffle, but at the same time we want to actually predict the envelop as fast as possible; that means, as faithful as possible. Now if we suppress all the high frequency terms, if we remove them very drastically what will happen the envelop will become flat and when there is a change specially in case of PCG signal, what we have to seen there is a blast of actually a signal whenever S 1 and S 2 are coming they are actually some blast of energy.

For that the envelop also is moving pretty fast, but if we allow all those components to capture that change if part of the riffle signal also is actually cripping in and that is riding on that envelop. So, that does not give us a actually a good feeling what we get that the signal that is the envelop; what we are extracted that is actually corrupted. So, where would be that cutoff frequency of that low pass filter is a very crucial choice and most of the cases you need to manually actually tweet that thing to get the best result. So, that the challenge we are talking about in these case.

So, people will have explored actually more strategies.

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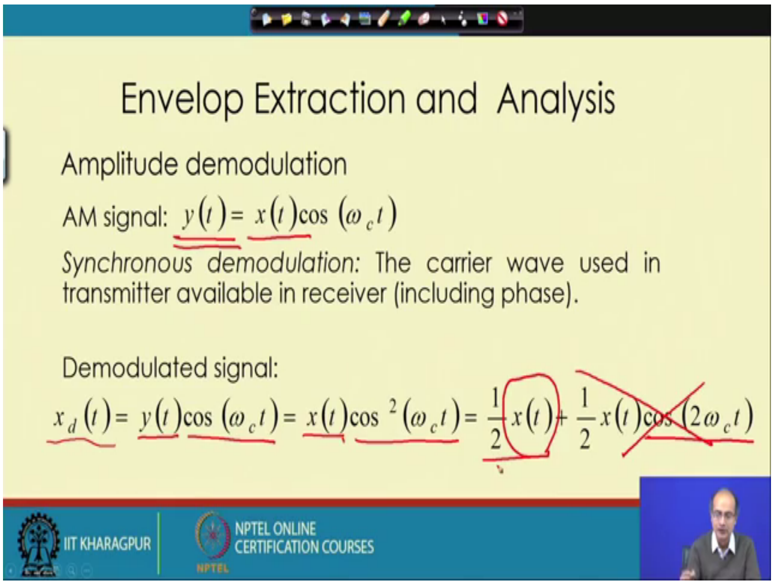

The first thing after release that comes in the mind that is amplitude demodulation we know that from the communication engineering how the amplitude demodulation can be done if you have a signal that is in the form that observe signal is y t which is nothing, but that amplitude modulated signal where x t is the that our the signal of interest and cos omega t is the carrier in that case if we are interested in the x t which we are taking as the envelop we need to remove the carrier signal. So, for that what we can do? First we go for the synchronous demodulation.

In case of synchronous demodulation, what we are actually assuming that we know that the carrier wave the carrier frequency as well as the phase that should be known in the receiver. So, as we know that we can make use of that to demodulate the signal demodulating the signal the purpose is to actually find out the x n from the y n; y n is given from there we need to find out the x n. So, for that what it is done that by demodulation that we have taken that demodulated value x t.

The first the part that we multiply it the observe signal receive signal y t with the same carrier frequency at the same phase. So, we get a term by replacing the y t we get a term involving x t and another term we get cos square omega c t. So, now, the cos square term it can be represented as that in terms of that the 2 parts involving one and other part is cos 2 omega c t.

So, by that what we get that out of these signal x d we have 2 part 1 is at the base line and another is modulated at twice the carrier frequency now if we do the low pass filtering to remove these part we are left with the scale version of the demodulated signal or the signal of the interest that is the x t only a scaling has come. So, that is the there is a way that we can actually get the demodulated signal using the synchronous demodulation.

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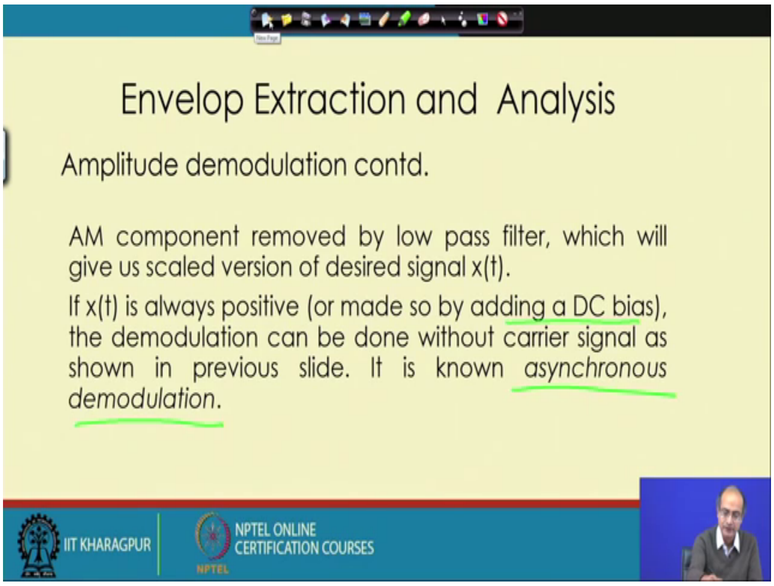

Now, that the low pass filtering again that is it crucial choice, but what is actually interesting here that here as a carrier frequency, we are getting the twice of the carrier frequency that gap between that modulated signal and the modulating signal that gap has increased. So, that makes the job a little easy; however, here one crucial thing, we need to keep in mind, we need to get very accurately that what is the carrier frequency and the what is the phase of it if we have some error in that it can actually lead to the problem. So, that the people that instead of synchronous averaging sometimes they look for the other one which is called the asynchronous demodulation and what is the example of asynchronous demodulation the first approach that what we have taken it can be taken as a part of asynchronous demodulation, let us take some say sinusoid just for a purpose of illustration.

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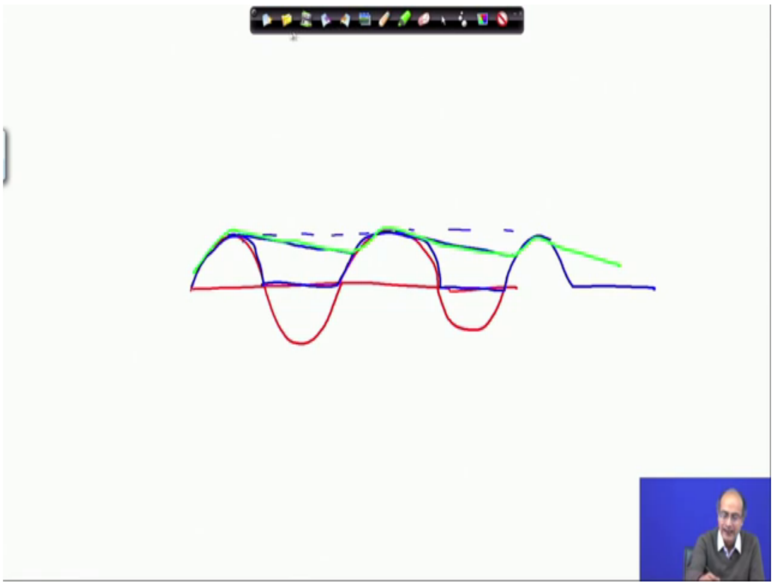

That if we have a sinusoid and if we take a actually half wave filtering we would get the signals like this that we will get the output in this way.

And if it is say modulated what you try to get you try to get this line actually. So, for that purpose we do the low pass filtering and then we would get here a drooping characteristics we would get these kinds of signals. So, the ultimate signal what we would recover would look like in this way the more accurately we can actually remove these things or the riffles it would actually give more and more accurate actually the envelop. So, that is what we call by our; that what we call by our asynchronous demodulation. So, that is the other technique which can compete with the part, but in this case, we need it to either make it positive by filtering or by adding DC bias we can also do that.

That if it is a positive signal, we can still do that and then try to remove that that actually riffle out of that and the we can actually get the envelop in that way ok.

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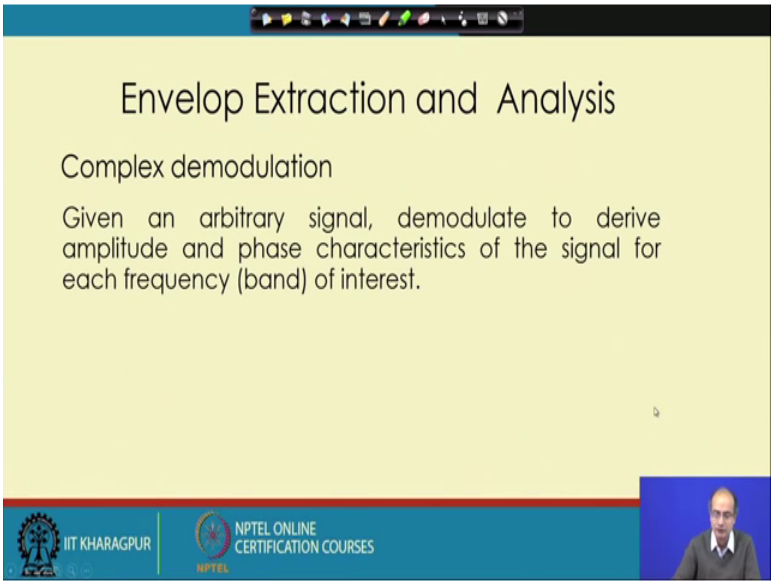

Next is complex demodulation in case of complex demodulation that there is some assumption first of all let us take the case that all the signals the spectra, they are not actually uni module.

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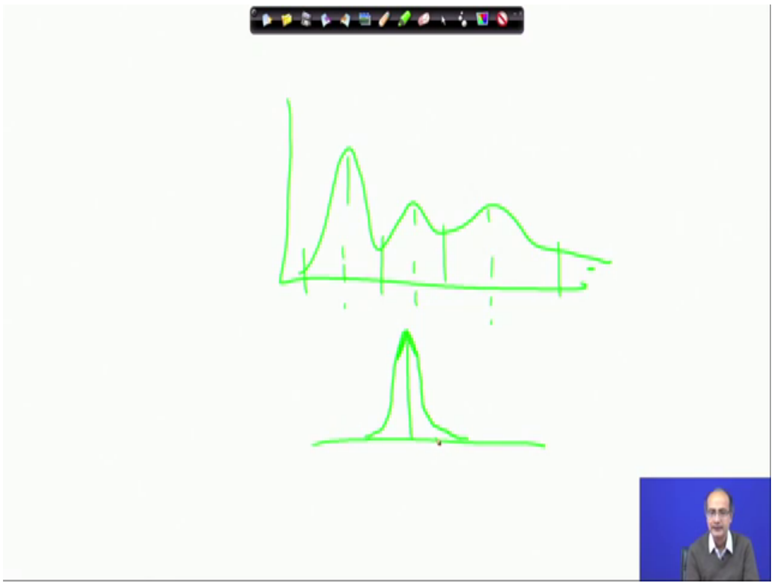

So, we can have multiple say peaks like this and in that case this may be idealized as that it has actually three bands of different central frequency and different band width. So, if we take that in that way then next thing we would like to know that what is the central frequency and what is the actually modulated wave that which gives this signal here, we can start actually from the other point to get the understanding more clear that if we take a sinusoid what would be the frequency, it would be a actually impulse like the p s d, it will give.

It will be all the energy will be concentrated at a particular frequency when we have a modulating signal with on this carrier then what we will see the p s d will actually be a peak taking that central frequency as the peak and it will get displayed. So, here we can think of this complex signal as actually combination of such amplitude modulated signal. So, that is the basic idea in case of the complex demodulation and in that case the idea is if the signal is constituted by couple of such signal which are amplitude modulated signal.

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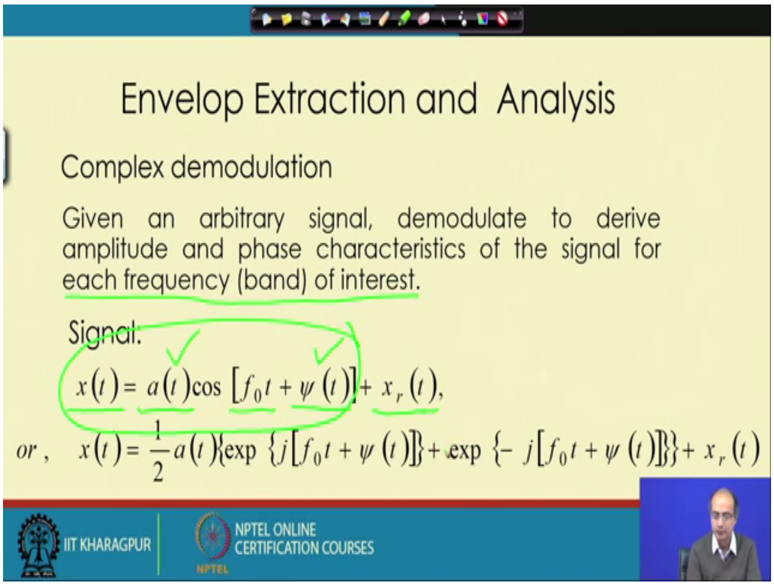

Let us try to find out those carrier frequency and that modulating signal that will give us a description about the signal in a better way.

So, what is the first task, we do we look at that for demodulation of that the amplitude and the phase characteristics of the signal for each frequency band of interest and usually these frequency band is determined by the p s d that we take actually each of the peak as one carrier and. So, the graph actually gives the limit that. So, we take as a band and that kind of that that each band we take one at a time and we try to find out that what is the carrier frequency and the modulating signal in this case. So, for that purpose the signal is represented in this way the signal x t, it is taken as that a modulated signal with the central frequency that is the peak what we get in the p s d within this band and it is modulated by a actually signal a t and there is some starting phase of it also.

And what happens to the rest of the part of the signal; that means, the energy and the band that is taken as the x r. So, this is the form at this movement we are not interested in the part x r that serves as a you can say that the remainder in this case and we are just focusing into this part and try to process this part to find out what should be our f t and what should be our phi t. So, for that what it is done we first do some kind of the mathematical operation to get it more clear the cosine term we expressed in terms of exponentials or a pair of exponentials with that frequency f 0 and minus f 0 and that x r remains in the same way.

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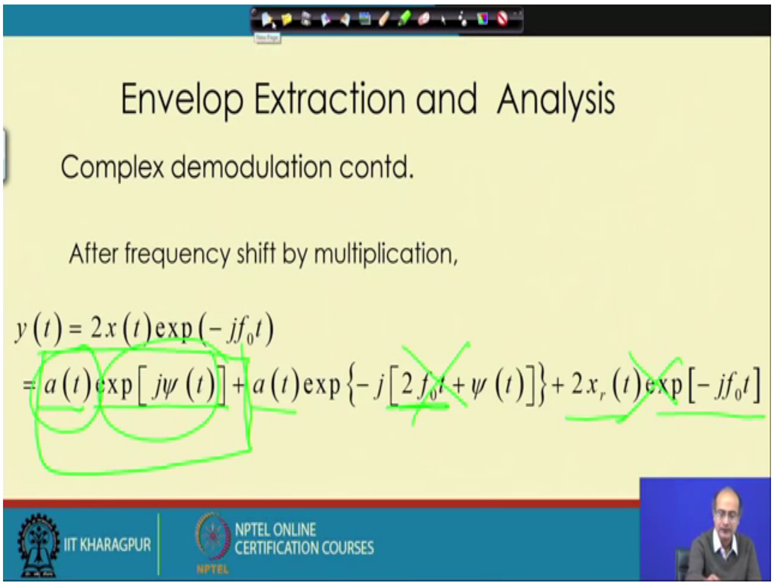

So, with that this complex actually a terms that we can actually multiply it with exponential minus j omega 0 t; that means, now we are whatever the signal we got we are multiplying it again with the one of the carrier frequency or negative of the carrier frequency and that helps us to get actually the first term that is the modulating signal and the corresponding phase of it plus the term with the modulating signal twice the modulating frequency and x r is also getting modulated. So, what we get that these signal using the phase band these part only this is in the base band rest of it they are at higher frequency. So, now, if we use a low pass filter, we can easily actually eliminate these parts and we can get rid these part only.

Out of that the amplitude part of it will give us that the modulating signal and angle will be given will give us the phase..

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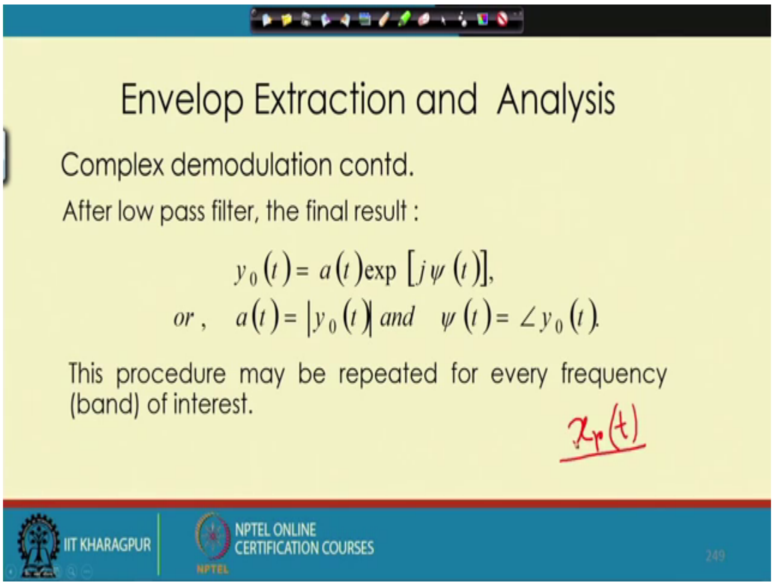

So, that is the way, we get that after the low pass filtering we get something y 0 that is a t into exponential j phi t and if we take the absolute value of y 0 t that gives us the a t. So, modulating signal here please keep in mind what is the assume that it is positive all through.

So, if your signal was really a oscillating signal where the values are going to be negative you would get only the part which is actually positive and the negative component it would be rectified and the we will recover that point. So, if we have a chance to synthesize that signal, then we should make sure that a t should be positive also; however, that may not be true in their life situation, then we should be satisfied with the rectified form of the that modulating signal and if it is a simple wave that the phase instead of becoming the time varying one it should be a constant for a simple example.

And by that actually that in this operation, we have found the result of one particular band we got the modulating signal and the phase of a particular band now once these part is removed that whatever we had as x r having the related the remainder of the energy, it is containing actually that the energy of the other bands. So, we should work on that x r; that means, the after finding out the a t and phi t we should remove these part deduct that part from the signal and gain that remainder part and we should work on that part again in the same way to find out the other that the modulating frequency and the corresponding phase and we should keep on going in that way to find out that modulating signal for each of these band..

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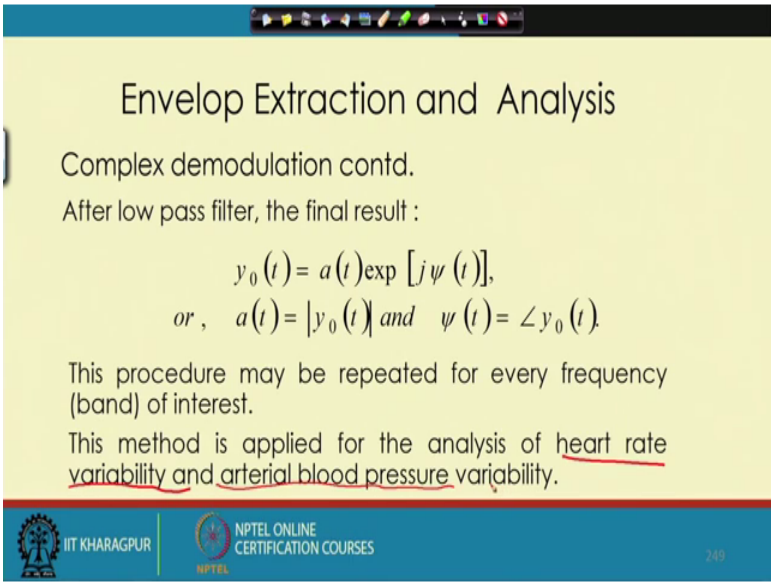

So, this process needs to be repeated till all the bands are actually cover. So, this method this has found actually a good application for the heart rate variability and the arterial blood pressure measurement variability and the heart rate and along with that that how the arterial blood pressure it changes. So, they are the signal is used and these signal also has been used in other places likes for speech synthesis and speech analysis there are a series of papers by Furu and Sandeep which actually makes use of these principles and they analyze and synthesize the speech with the help of complex a MFM signal.

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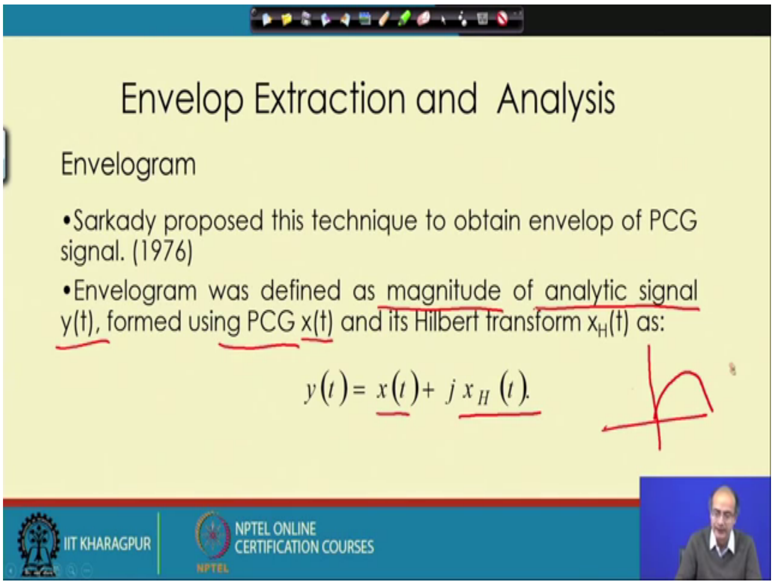

So, now let us proceed to get actually another technique that is called envelogram envelogram, this is actually having a unique idea that they are bring the same thing from a different point of view this is proposed by the scientist the Sarkady in 1976 and he has actually noticed that that we can use of the analytic signal, we can use of the that analytic signal for this purpose the magnitude of the analytic signal we can take and he proposed that the use of it for the PCG which we represented by x t and for that the corresponding analytic signal y t, it can be represented by x t as the real part and the Hilbert transform of the x t which is represented by the x h t as the that imaginary part now what is the easiest way to find out that the analytic signal for any signal which is real we know that spectrum would look like this spectrum would be symmetric say it would be like this.

Here we should assume that both sides has asymmetric and in case of analytic signal; however, these part actually, we can erase it for analytic signal the negative part of the that frequency, it does not have a any energy. So, the corresponding signal in time domain it would be a complex signal always and the magnitude of that actually we can make use of it as the envelop.

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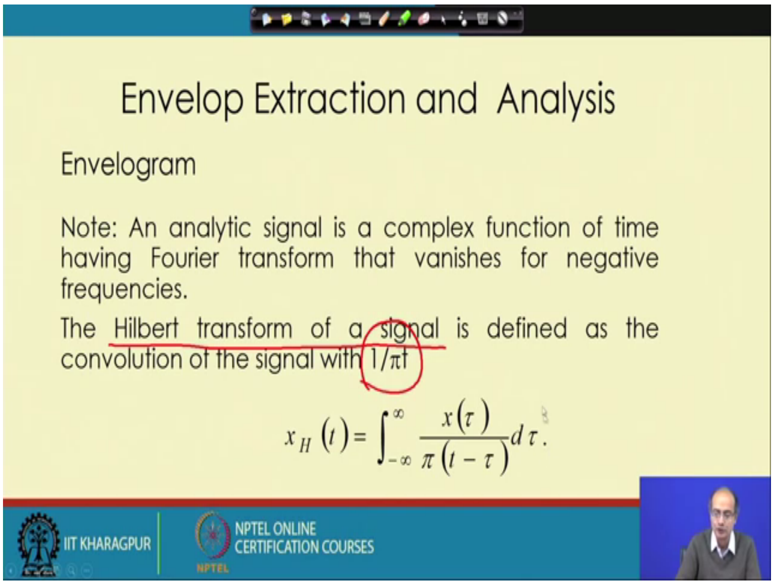

So that what we have to do for that to get that we need to find out the first the Hilbert transform. So, Hilbert transform we can get actually by convolving the signal with one by pi t that is the direct way of getting the Hilbert transform. So, we can actually get that that Hilbert transform in this way, but having a that integration like this specially when the x t is not known that it does not have any close form and numerically solving it or not also an easy task.

So, we look for a better way of doing that and for that.

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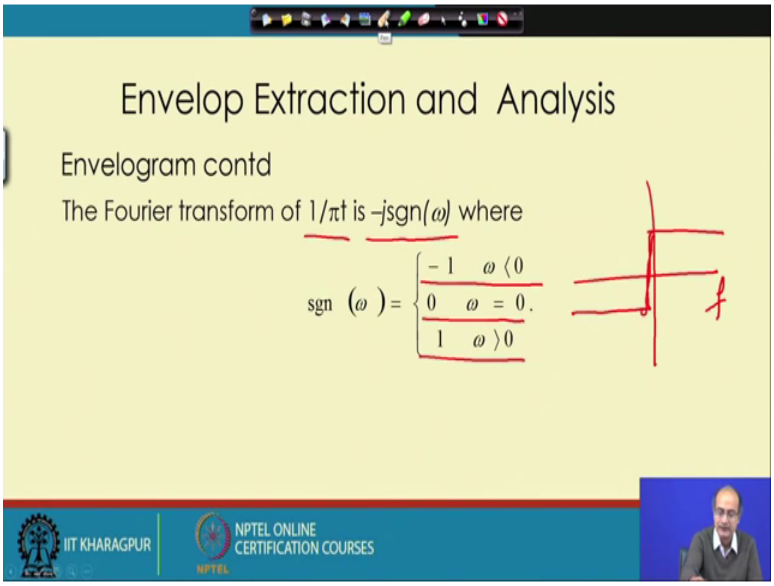

We look at the frequency domain transform that frequency transform of these signal that 1 by pi t, it comes in a nice form that it comes as minus j sign of omega and what a sign of omega, it would be negative for negative frequency positive for positive frequency and the both the sides are magnitude is one and it should be 0 at actually value 0.

So, if we take the axis. So, it is like this that is negative and then having a transition that 0 and then it is going to actually positive at that point at the positive value. So, that is a actually known form and we can make use of that.

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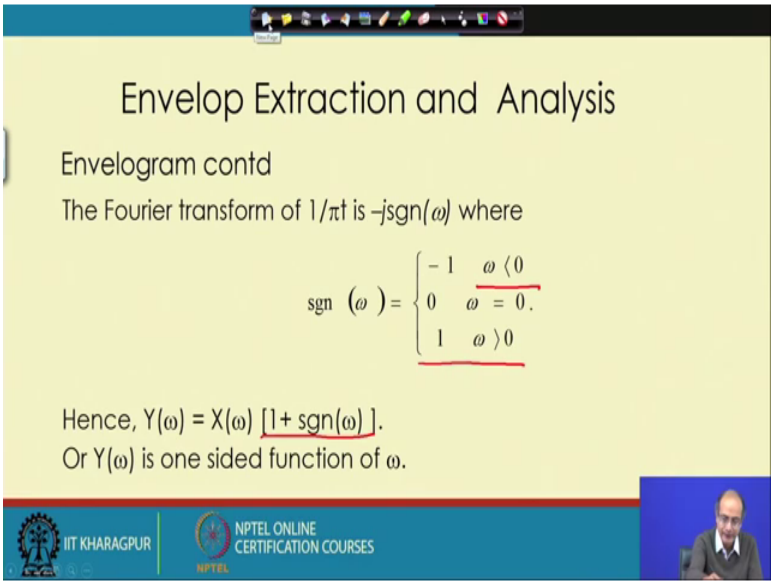

To get actually that affect of it, we know that we have to compute the Hilbert transform and if we look at the resultance signal of that analytical signal we can make use of these Fourier transform the convolution becomes multiplication in the frequency domain. So, the Hilbert transform part would be in the frequency would be x omega into sign of actually that j omega.

So, with that and another part real part will again contribute that x omega. So, we get that the analytic signal in this y omega in this way we get this form. So, using that we can very easily find out the value and here one thing we can note that if we take these term one plus sign omega the negative sign of the frequency that the values will be 0 for these term for the 0, it would be one and positive side it will become 2. So, it is a one sided function of omega irrespective of the spectrum of x omega. So, in this way we compute actually the analytical signal and here we go through the states of it.

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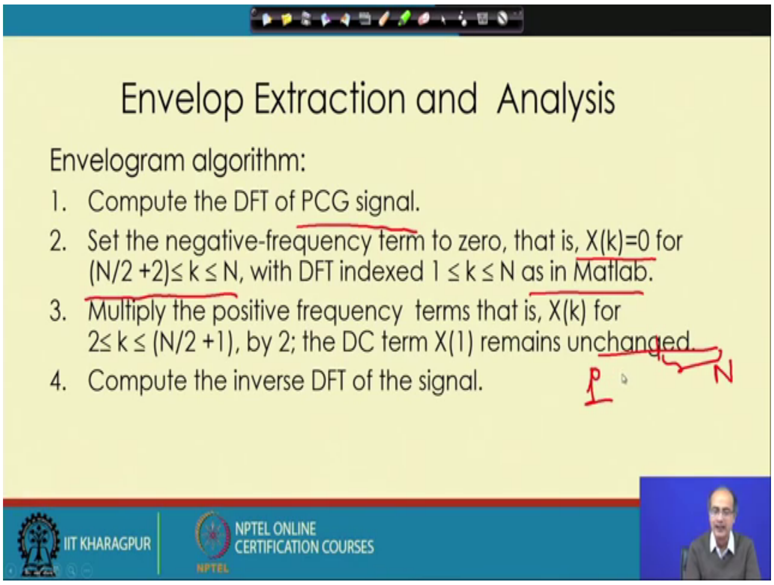

The first part or the first step is we compute the discrete Fourier transform of the PCG signal because PCG signal is the signal of our interest it can be any signal which we want to actually calculate the envelop next for the negative frequency part we to force them to 0. So, for that what we do that x k that is the d f t part we forced to 0 for the coefficients which are between n by 2 plus 2 to n; that means, if we have the terms starting from say 0 2 or the rather let us take 1 to n. So, half of it and these part we are taking forcing into 0 and that way and that is the way actually the Matlab negative sign comes here. So, that is the way it is done for that Matlab if we have a define routine that gives the negative actually indexes in a different way we should do the appropriate changes for that.

So, now, let us look at the next part for the positive frequency x k that from 2 to n by 2 plus 1. So, that part we should multiply it by 2; that means, the positive part of the spectrum we multiply by 2 and the DC term which is given by the first term x o one that will remain unchanged we would not like to change that because we want to compute the that the analytic signal and there is a definition remains in that way.

Now, whatever is remaining in the spectrum that gives us the spectrum corresponding to the analytic signal and if we take the inverse DFT, then we get actually that analytic signal and from there we can easily.

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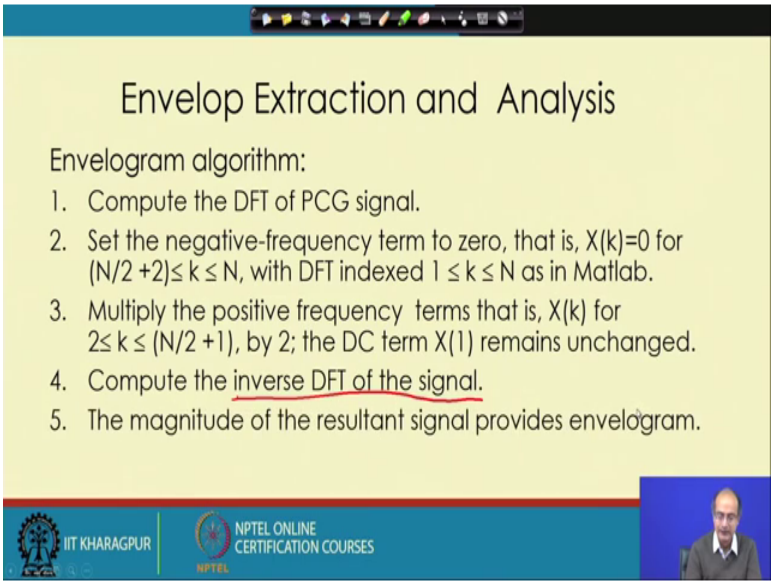

Get actually the resultant signal the magnitude of that resultant signal gives us the envelop. So, that is what we were looking at and this is the way that we get that our job done that is the last part we have covered for the envelop detection. So, we stop here.

Thank you.