Biomedical Signal Processing
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Lecture – 20
Event Detection (Contd.)

So, in the session will take one more the very best algorithm, which is much more sophisticated than the previous algorithm that is called the Pan-Tompkin algorithm.

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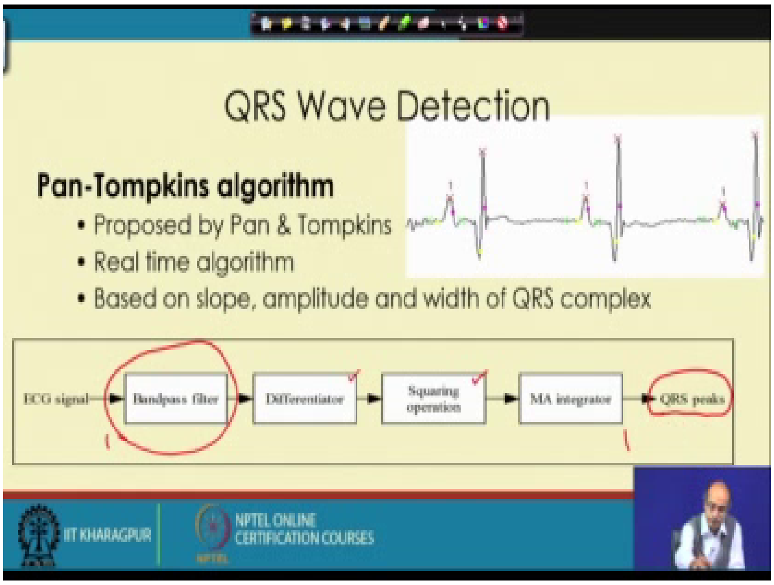

This Pan-Tompkin algorithm, it is actually that proposed by Pan and Tompkin this two scientist. And it is actually proposed for the real time operation, and again it is actually proposed based on the slope, amplitude, and width of QRS complex. So, this is the things keeping in mind that this technique was proposed. And the first part of it is that there is a band pass filter. To clean the signal we know that when we are using that differentiator or the derivative, so that it will accentuate the high frequency noise. So, we need to get rid of them.

So first, it is first way band pass filter, next that differentiator, after the differentiator there is a squaring operation. And, we know that if we take the square and if the amplitude is more than 1, actually it gives a non-linear enhancement to the signal magnitude and the high amplitude one it will be magnified and the difference between the low amplitude and the high amplitude the signals they will get separated. That is followed by again a MA filter, because we have seen in the previous two cases that many times output is jiggered, so to smooth it out. And then the last part is the QRS peaks. And for finding out the peaks that the Pan-Tompkin has suggested and elaborate procedure for that. And if you look at this part that is that band pass filter to MA integrator it would look like that it is just an extension of the previous techniques, but the real actually that impact comes that in terms of knowledge as well as that the sophistication that comes in the QRS peak search.

So, let us go through that algorithm step by step.

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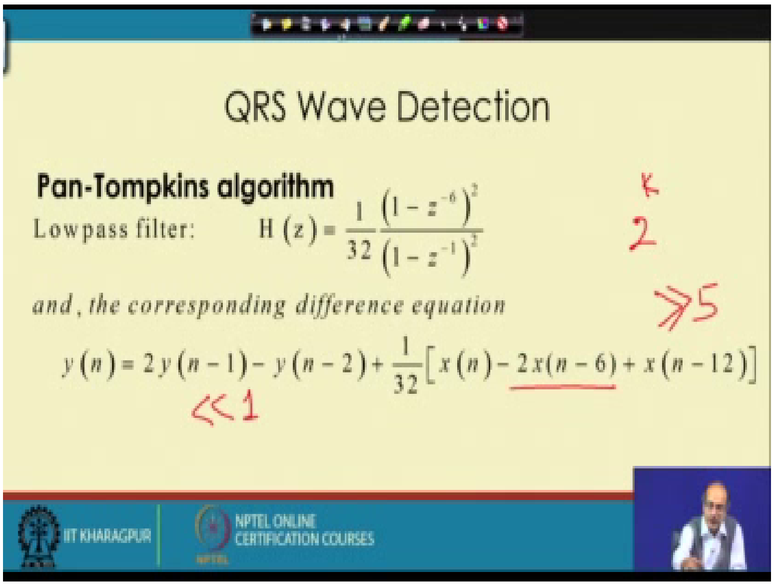

So first, the signal is passed through that we have to pass through a band pass filter for that in low pass filter is taken, that low pass filter the transfer function is given here and the corresponding that the difference equation is also given. And we have told that one thing that we have a real time, actually that algorithm we are looking for that the Pan-Tompkin has proposed. And for that purpose what they have taken? They have taken a actually IIR filter, they are using that both the poles and zeroes.

And for the IIR filter if you look at the weights, all these weights they are actually power of 2, either that power of 2 this k is positive or it is negative. So, when it is say 1 by 32; that means that in that case it is that 2 to the power minus say that 2 to the power 3 is 8, so 2 to the power 4 is 16, 2 to the power 5 is 32. So, that is 2 to the power minus 5 will give us 1 by 32. And that would be implemented by actually 5 right shift operation of the register, ok

So, keeping that in mind that while we are multiplying which power of 2 that for example, here we are multiplying with that, that would be left shift operation one left shift operation will give us multiplication with two, here also will do the same thing. And when we are taking that division by 32 that would be affected by that 5 right shift operations.

So, that is a way that they have implemented so that the operations can be done in a very fast and frugal manner.

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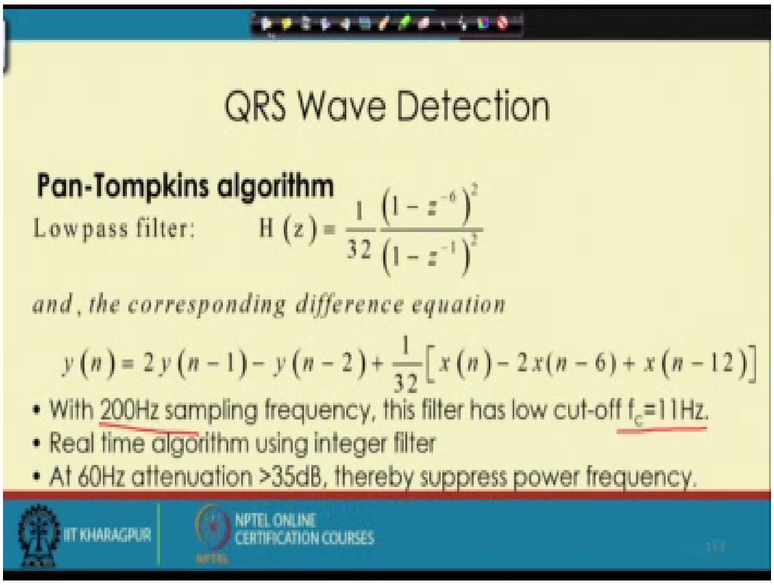

That what they have taken that they have taken the signal that is sample that 200 hertz sampling frequency. And for that this filter has a cut-off frequency of 11 hertz

So, what we get that it is a very I would say that low cut-off; that means the maximum part of the signal and the noise; of course, the high frequency anything above 11 hertz that will get eliminated. And that also take care of the power frequency that in North America that would be 60 hertz, in our country or some part of the Europe that would be 50 hertz that is well above this cut off frequency. So, that will be eliminated by this filter itself.

And they have used actually this integer filter for the real time operation. Attenuation at 60 hertz would be more than 35 dB, and that is why that this operation is good enough and we do not need any more notch filter in this case.

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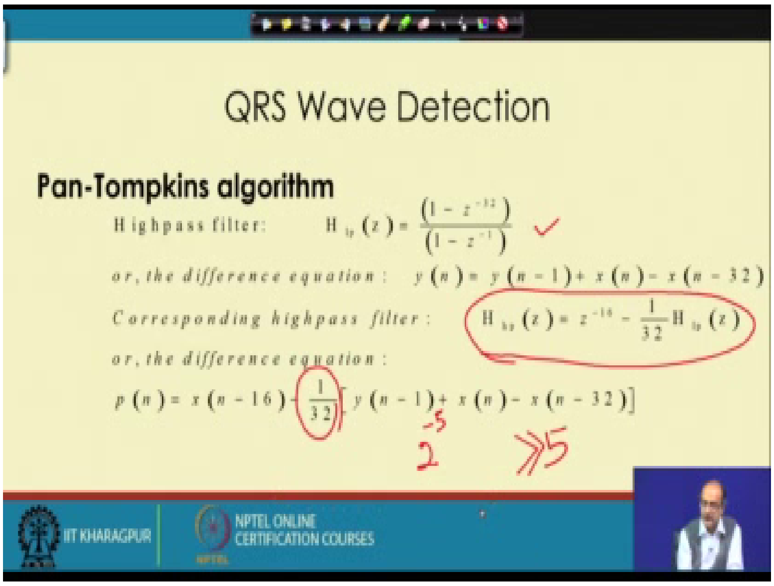

Now that, as we told that we take actually that a band pass filter, so we have to have a high pass filter also along with the low pass filter to do the job. And for that first will start with a low pass filter. So, this is the low pass filter, the corresponding that the difference equation is given. Again note that, all the coefficients are 1, so no multiplication is required only addition is required for that. And, from that low pass filter that the corresponding high pass filter is actually derived from this low pass filter. And this is the corresponding high pass filter, the corresponding the difference equation also is given. And here that the only term other than that one that is 1 by 32, we know that this means that it is power of 2 to the power minus 5 and we can actually incorporate that for 5 right shifts operation, ok.

So, again this high pass filter is also a real time integer filter and with that we can actually as we apply it after the low pass filter this high pass filter along with the low pass filter gives rise to a band pass filter.

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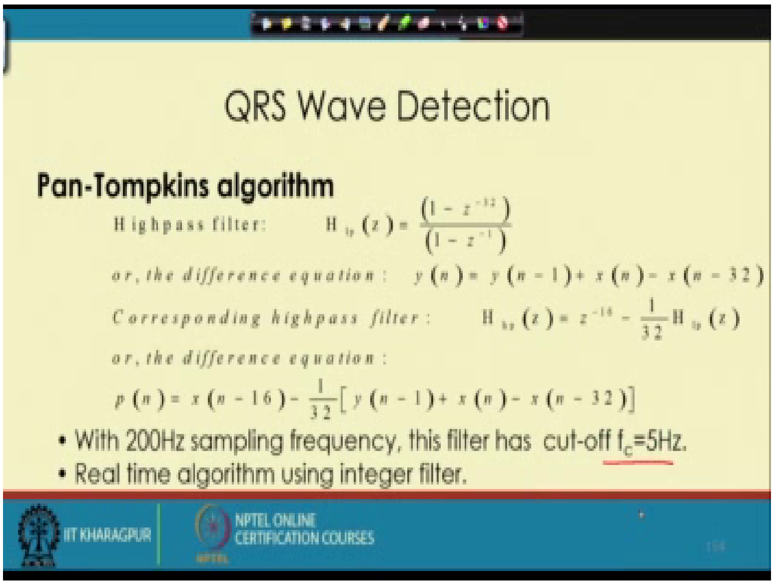

And that for these if we look at that cut off frequency, the cut off frequency is low that is 5 hertz. So now, the signal is actually restricted in between 5 hertz to 11 hertz; only that part of the signal is preserved. So, below the 5 hertz means that all the low frequency signals are also eliminated; and that means, Pan-Tompkin has found that most of the energy of the QRS complex is concentrated within this band: 5 to 11 hertz. And if we just keep that and throw away the other part that is good enough for the QRS complex detection.

And this real time algorithm that helps us to; actually do the fast operation and for that the integer filtered is used again.

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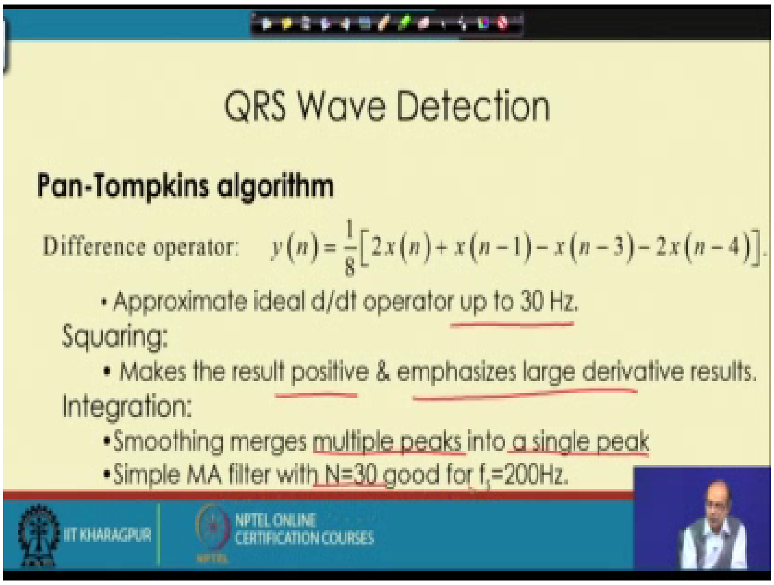

Now, let us look at the corresponding the difference operator. That, for the Pan-Tompkins algorithm they have a little more elaborate that difference operator. That, the difference operator is weighted actually difference operator is taken; some of the weights are 2 some of them they are 1 and that for averaging that 1 by 8 is taken. And again you note that this is an integer filter, because the coefficients are integer and the divisions are also power of 2. So, again that multiplication can be replaced by the shift operations here. And after that differentiator that will do the squaring.

Now, for in this case these difference operator the way it is chose it act as a actually ideal derivative filter up to 30 hertz. That means, within that band that 5 hertz to 11 hertz where the signal is present or preserved at that part it has a linear actually operation. And, for that when you pass it through that will get the high frequency terms would be emphasized and followed by squaring operation which will help us to make it positive and emphasize the large derivatives. And following to this squaring operation there would be an integration block which will help in smoothing and multiple peaks it will actually merge into a single peak. And for that purpose that MA filter is used and they found that N equal to 32 that gives actually good result for f s equal to 20 hertz.

Now, if you ask that why they have chosen N equal to 30 is difficult to answer. The only way we can think of that they emptily come across these numbers. If we think in terms of the that integer filter we should have chosen 32 instead of 30, but I think what they may have done that MA filter that when you are actually suppose to compute the average actually they may have taken as a sum, because following these there will be a threshold operation. So, that division by 30 if it is omitted instead of average will get the sum that will simply give a scaling. And so long that our computer that registers bandwidth that it can actually handle that number there is no problem with that scaling. That may be that the reason that they have settled it to 30 not gone for 32.

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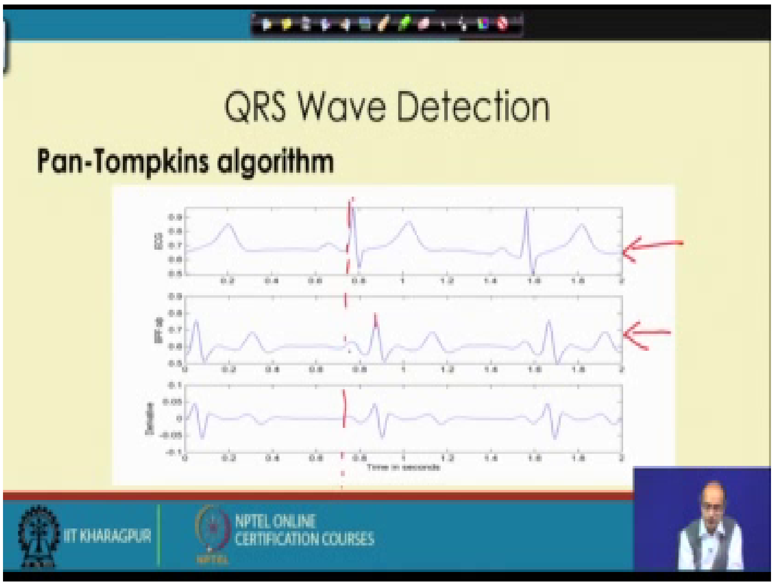

So now here, let us look at that how the signal changes by these operations. First we show at the top that this is the ECG signal, and after the band pass filter we see that the signal is actually distorted that we get the QRS complex that is the most prominent one. Also the T signal is also there, this not completely eliminated. And another thing we note that if you look at that the position of the r, because of these filtering it has added some delay. So, the position of the QRS complex or that R wave varies it has actually that it has been delayed and the delay is corresponding to the delay introduced by this band pass filter.

Next we look at that squaring operation after the squaring we get that the output of this Steve wave that is giving rise to small wave form, whereas QRS complex it is giving a much larger wave from. So, using that be actually we get that separation of the other waves an amplification of the QRS complex.

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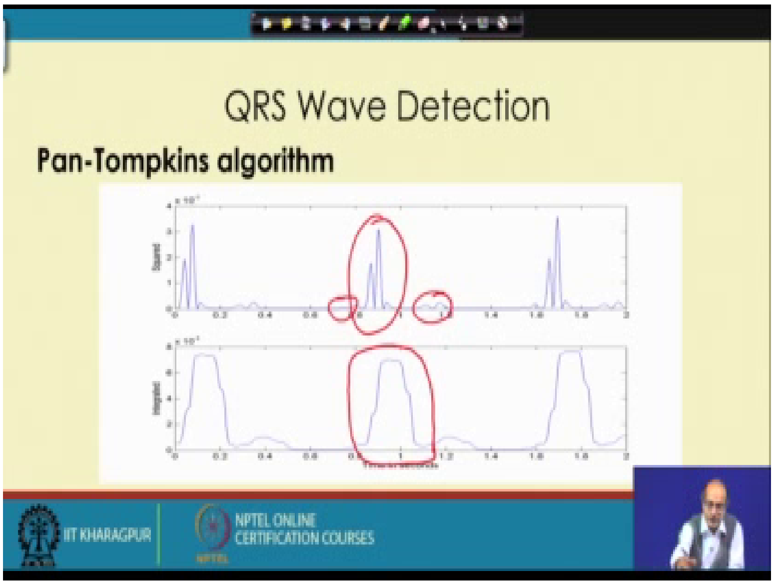

So once it is squared, now we can get that effect that the R wave that gives multiple peaks and that is much higher than the peaks created by the T wave. And you see that P wave hardly that is visible nothing is there almost. And, once it is pass through that integrator that is a MA filter of order 30 then we get that it becomes a solid actually stamp that all the peaks they get merged and give raise to a post kind of structure here which can be actually threshold to find out that there is a QRS complex is present at this position, and it is very difficult to (Refer Time: 16:29). So, up to this we get actually from the knowledge of that previous algorithm.

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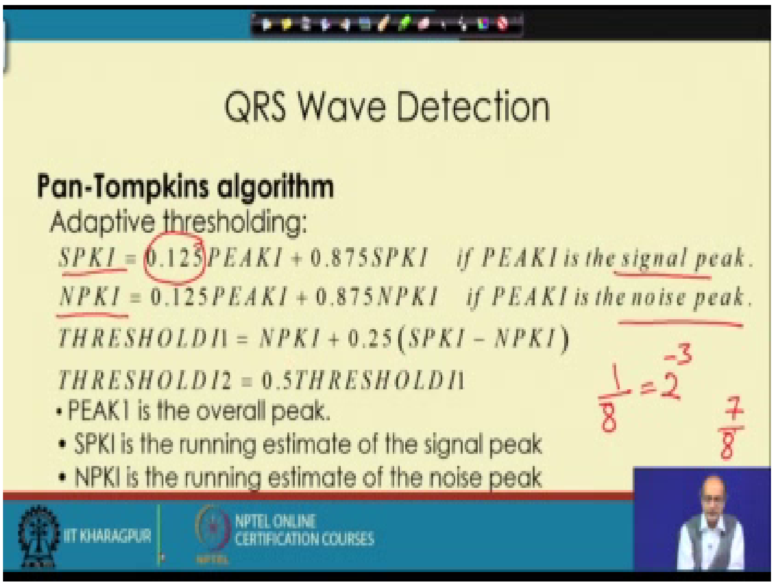

Now, comes the special contribution of Pan-Tompkin; that the first change they have made the thresholding should be adaptive. That is the beauty of the Pan-Tompkins algorithm. So for that what they have done that, they have introduce couple of terms that first one is SPK 1, second is NPK 1, and two threshold: threshold 1 and threshold 2 they are introduced. Out of that if we look at that here the inputs are the peak, peak means that the overall the peak whenever after we take a threshold that the peak what we get that we get anything above the threshold that is a peak. So, using that we are actually updating that SPK 1; that is the signal peak running estimate and NPK 1 is a running estimate of the noise peak, ok

So, if we take a threshold whatever we get above the threshold that could be a signal peak or a noise peak. If it is a signal peak that is used to update the estimate running estimate of the signal peak. If it is a noise peak then that peak is used to update the estimate of the noise peak. And in the both the cases what we get that we are taking actually 0.125; 0.125 this fraction this comes out of 1 by 8. Again, it is a power of 2 2 to the power of minus 3 has been taken. And these part what we are taking, this is actually if it one 8 the other part is 7 8. So, the previous value whatever was there it is multiplied by 7 and then 3 right shifts will give. So, one-eight part of the that previous estimate that is taken and it is actually used to modify both the signal peak and the noise peak.

And after that we go for the selection of the threshold. And the philosophy is very simple: that threshold should be above the noise peak so that no noise peak should be detected. And it should be much lower than the signal peak so that we are able catch actually all the signal peaks; there should not be any miss of signal peak. So, what they have proposed that the threshold 1 it should be above the noise peak so that is the base line. And then the difference between the signal peak and the noise peak; that running estimates the difference of that one-fourth of that difference is added with the noise peak. So, that would be the threshold.

That is the threshold 1. And 50 percent of that threshold 1 is taken as the threshold 2. So, there is a that number of actually that terms we get and this how we define the threshold 1 and threshold 2, ok.

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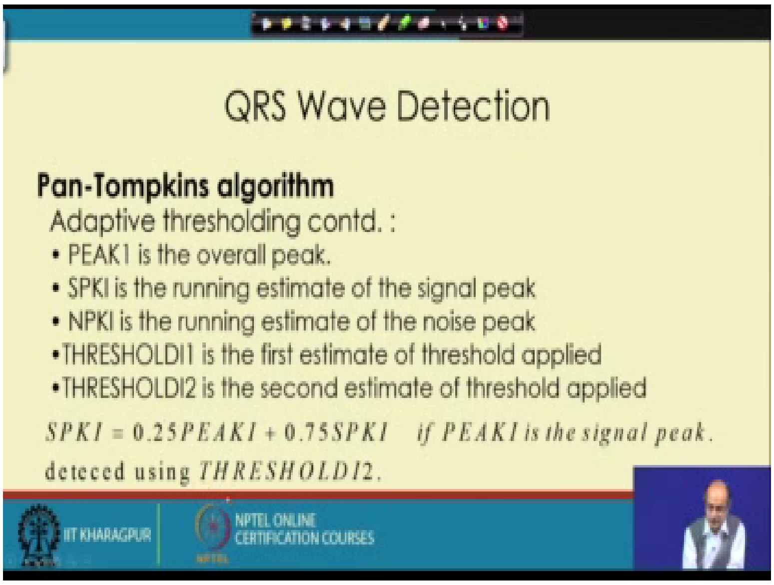

Now, let us go through the definitions once more so that we can give remember that, but then that we have the peak 1 that is the overall peak whenever we get one post that is above the threshold. The peak of it is taken as peak 1. That will have some more steps to find out whether it is a signal peak or a noise peak. If it is a signal peak then that peak is actually used to again update the signal peak estimate; running estimate. If it is a noise peak then that is use to update the running estimate of the noise peak. And we have used that using that this two running estimates of the signal and the noise peak that the threshold 1 that is a first estimate of the threshold first threshold that is actually calculated and half of the value of the first threshold that gives a lower threshold that is we call the second threshold or threshold 2.

So, with that we proceed. And here one more thing that if a signal or a peak we miss by this that threshold 1. That means, a time has passed we could not get any peak above the threshold 1, then we would fall back on the threshold 2 and try to find out whether we can get any peak. And if we get a peak by threshold 2 and if that peak is a signal peak then our speed of update of the signal peak becomes more. Instead of one actually eighth we take one-fourth part of this new peak and three-fourth part of the previous estimate of the signal peak and we update the running estimate. That means, what we realise that due to some reason may be contact resistance or so that signal amplitude has deduced. And because of that we could not get any peak above the threshold 1. Under that situation we need to update the thresholds so that we do not miss any peak. And for that we have used that second threshold to get that peak.

Now, after getting that peak if it is a signal peak we should update that estimates more quickly so that next time such thing does not occur. So, we increase the speed of adaptation in this case.

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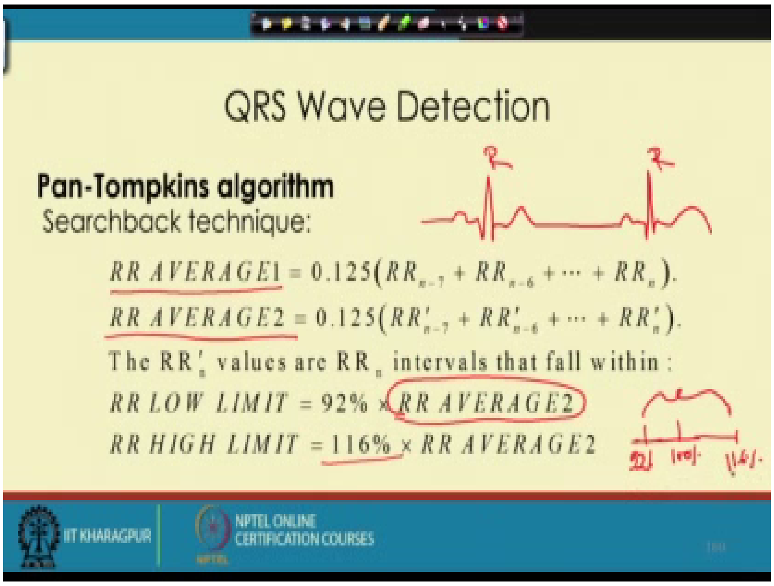

Now, we get the search back part of it. Here, we have two quantities of the RR average. If we have a ECG signal we know that we get the p q r s t then a gap again we get p q r s t. So, in between the two actually bits this difference between the two r's that is called RR interval. And for this RR interval they are having that two averages actually estimates of the RR interval. The first one: it is taking the row one that whatever the peaks are we get that we take those that corresponding the RR intervals and that gives raise to their RR average 1. And from these RR intervals row RR intervals we can say that we refine them and then the refined estimate that is taken as RR dashed that is used to calculate the second average.

Now, how that is refined? There is a limit lower limit and upper limit that is set that if the RR average 2 is taken as actually a more stable or the robust estimate of the RR interval. If the lower limit is actually set as 92 percent of it and the upper limit is taken as the 116 percent of it.

So, please note that they have not taken a symmetric window, if this is 100 percent that lower side if this is 100 percent lower side they are taking it 92; that means, in allowance of 8 and upper side 8 percent upper side they are taking it 116 percent means an allowance of 16 percent. If it is within that, then that RR interval we call as RR dashed; that means, that is an acceptable one. If it goes outside that then what could happen, RR interval if it is bigger, means bigger than the that within these limit if it above the 116 percent. That means that is an unusual kind of situation: for a healthy person the person that heart is beating so that the QRS complex would be there all the time. So, if it is absent, so what is actually concluded here that our algorithm has missed to peak actually that QRS that beat or the R actually peak. That means, we have missed one and that is why RR has become more.

The other option could be if it is very small; that is less than 92 percent of the RR average that such a sudden change in the beat does not happened even if there is change in the rhythm that is a gradual one. If the RR interval is smaller than that signifies there must be a noisy peak that has actually reduced the RR interval. So now, we need to take care of that noisy actually peak. So, if it is below that then we get that new peak as a noisy peak and that helps us to actually change that running estimate of the noise peak. And if it within these interval then it is a signal peak then that helps us to find out that from the signal peak that we can get that RR average 2 as well as we can estimate or update the estimate of the signal peak.

And if it is above, that means you have missed something. So, we need to look back and find out that missed peak and that is a part we call as sear back operation.

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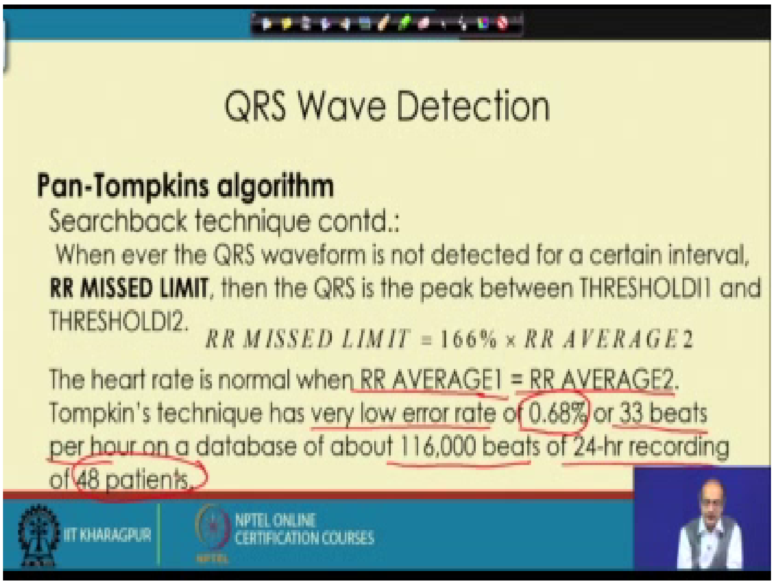

So, as a part of it what is done? Whenever the QRS wave form it is not detected within that is a started interval that is called that RR missed limit. Then the QRS peak between the threshold 1 and the threshold 2 is taken. In fact, the first part is imperative we call it missed, because there is no peak above the threshold 1. And in that situation we can get a peak only if it is above threshold 2. And that essentially means it has to be between this two threshold: threshold 1 and threshold 2. So, that becomes the new peak what we take here.

And that RR missed limit is actually set pretty high it is taken 116 percent of the RR average 2. So, that RR average 2 is the stable average of the robust average of the RR interval if it is 166 percent of that; that means, 166 percent more is considered as the limit if still we do not get that the R peak. That means, we assure that we have missed that R peak, because the signal cannot actually heart can stop beating that long or the frequency cannot go down that low. So, we should go for that the threshold 2 and take that peak above that threshold, ok

So, when the heart rate is normal that we do not have any problem then we get that RR average 1 and RR average 2 they should be similar. There is no reason that they would be different. However, if there is fluctuation in that the heart beat then this two can actually vary. And using this algorithm which is a real time algorithm the Pan-Tompkin they found that a they could achieve a very low error rate; that is 0.68 percent. And, that means, only 33 beats per hour for a huge database of 116000 beats of 24 hours recording of 48 patients. So, with a very exhaustive state of database of 48 patients and by collecting that 24 hours recording; that means, throughout the day whatever the changes has occurred they could actually get a very good actually estimate only 0.68 percent that the R beats are missed in this case. That means, less than 1 percent actually loss.

So, that is a way Pan-Tompkin has contributed in terms of a very sophisticated that QRS detection algorithm.

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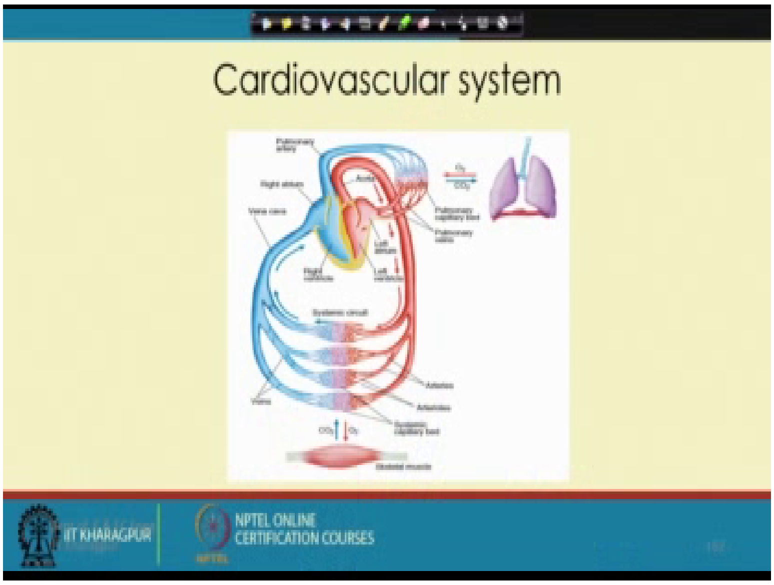

Now, will go through another a small part before concluding this session. We have seen that the cardiovascular system. And a part of it we told that the left ventricle which is pumping out the blood through the arteries and which reaches throughout the body; that the first the QRS complex signifies the start of the compression then between s and t that the contraction actually remains, and because of that the blood flows out of the ventricle because of that pressure.

And when the T wave comes which signifies the start of the relaxation of the ventricles. The moment the pressure of the blood inside the left ventricle it becomes lower than that of actually that means the blood pressure within the that aorta. So, then the valve that is helping for the unidirectional flow of the blood in the aorta that will get closed; and that gives rise to a small dip in the pressure and that will remain constant after that because the valve is closed even if the ventricles relaxed it will not allow for a back flow of the blood.

And that part we would like to capture.

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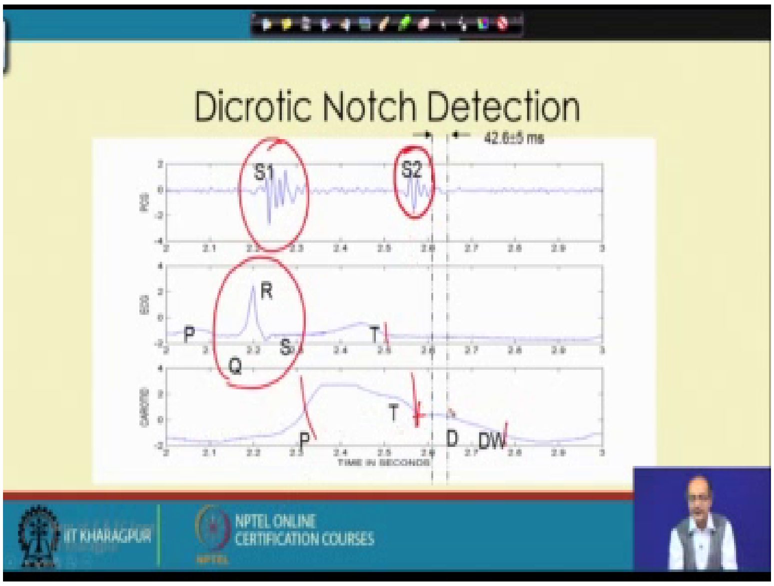

And that can be done by the carotid pulse signal. That, we know that the carotid means it is that is an artery that is going towards the head for supplying of the blood. So, when we have the QRS complex, here is the QRS complex of the ECG that is associated with some noise called actually S 1 signal. And S 1 signal we get in that PCG signal the phonocardiogram, ok.

Phonocardiogram is that audible sound we get near the heart. And how that is generated it is generated by the sound of the valves. And again that the blood is oozing out from the ventricles to the arteries or in the left ventricle to the aorta. Now it gives raise to turbulence and we get that noise as the S 1 signal.

So, S 1 signal can actually help us to find out that whether it is the normal sound or in case of some steroids in the artery or imperfection in the valve. Say at the time when the blood is flowing from the left ventricle to the aorta if our valves in between that aorta and the ventricles they should have closed before that. If they have not closed properly some blood can actually back flow and that will give raise to some sound. So, all such scenes will be actually depicted in the S 1 sound or S 1 vibration. And that the way we are getting here the PCG signal it is a very clean one or clean of it. It is actually super impose with lots of noise including the power frequency noise and a high frequency noise. So, we do not get such clean signal. So, we need to really have some idea that where the S 1 and S 2 sounds are occurring.

So, the first trigger to get the position of S 1 comes from the QRS complex. And already we have seen that how we can capture the QRS complex the three algorithms we have learnt. Now for the S 2 one that comes when the ventricles relax and the valves which was so far allowing the flow of the blood from the ventricles to the arteries they closes down. If the valves get calcified; so instead of a soft sound it will make a metallic sound and that will give raise to an increase in the S 2 sound.

However, to get that were that is occurring it becomes very difficult, because a T signal is a non-event its subdued signal we cannot get actually when actually it is finishing. So, instead of looking at ECG from where you cannot get really the boundary that of the end of T signal we go for the carotid pulse signal which speaks up after the QRS complex the pressure in increasing. And when the valve is actually closed there is sudden dip which gives raise to a deep in the pressure.

Again as the valve is closed that pressure remains constant and slowly that is going down. So, we can tell this part of the pressure is the systolic pressure and this part is the diastolic pressure actually, ok. So, in between this change this notch in this carotid pulse if we can catch this position it can help us to get the location of the S 2 signal in the PCG signal. So, that is the importance of the carotid pulse signal and the notch. This two are very close to each other.

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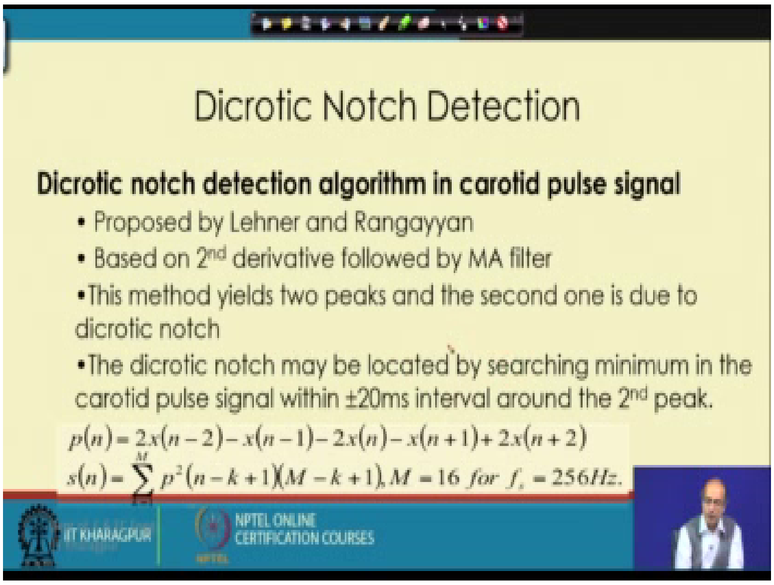

So, Lehner and Rangayyan they have used again a second derivative of the best technique. So, they have suggested that let us take the second derivative of the carotid pulse signal followed by a MA filter. And these will give raise to two peaks: one peak would be for the actually that if we take the second derivative that it will give the peak somewhere here where there is a increase in the signal amplitude. Again that will give a peak here. So, to get that is why they are talking about that you look for the second peak out of the two peaks the second one is taken that is corresponding to the dicrotic notch. And dicrotic notch may be located that were the peak comes they look for local minima around that peak by taking an interval of plus minus 20 millisecond around that second peak.

That is how they could actually get the location of the dicrotic notch. And we know from that we can get that time window where the S 2 is located.

Thank you.