**Machinery Fault Diagnosis and Signal Processing**
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Lecture - 27
Signal Conditioning and Filtering

In this class, we will continue our discussions on transducers and measurements and in particular, I will focus on signal filtering and also discuss something about signal conditioning.

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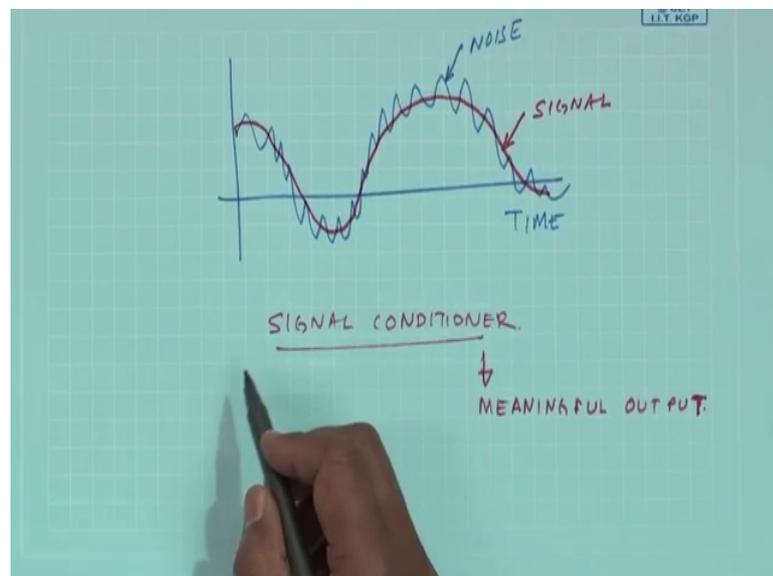
You will recall that I was mentioning in my earlier class that certain signals may have certain noise in it. For example, my actual signal is this red line, but the way I have measured it is this green; this blue line is riding on it. So, this is my actual signal and then I have this noise modulating it or riding over it. So, we have to use this is just one example, there could be many examples as well. So, we have to use signal conditioners first to condition the signal. So, that it contains or it provides meaningful output.

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So, what are the different signal conditioning elements you will see? Amplification, I will come to the amplification little bit later. Filteration; so, that I can remove certain noise or linearize it or provide a power supply. So, signal conditioning element can have all these quantities. Now if I talk about signal filtering.

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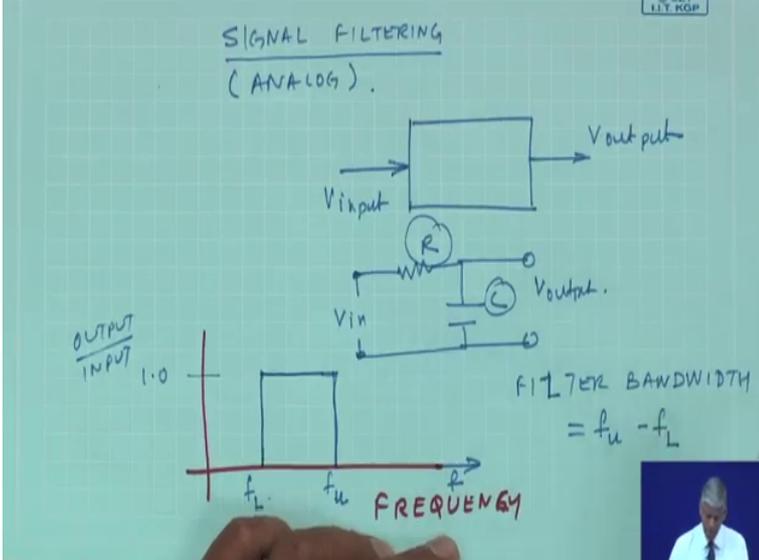
So, if you think about analog signal, we will talk about analog signal, if you think of a filter, this is filters are realized some input V output.

So, this in electrically, you know, this could be having a combination of R and C and get a V output and so on. So, we can design a realized filters depending on the combinations of R and C, whether they are in series and parallel and so on, we will not go into details of this in this course because this is not a class on filters filtering, but you will see that any device, if you think of it a signal has a frequency.

So, if I design a filter; a filter ideally means some f lower f upper and this would be one this means output by input; that means, here any signal which comes into this filter it will only produce an output between f u and f L. So, the filter band width is nothing, but f upper minus f lower I will draw it again here. So, such commercial filters are available in the market.

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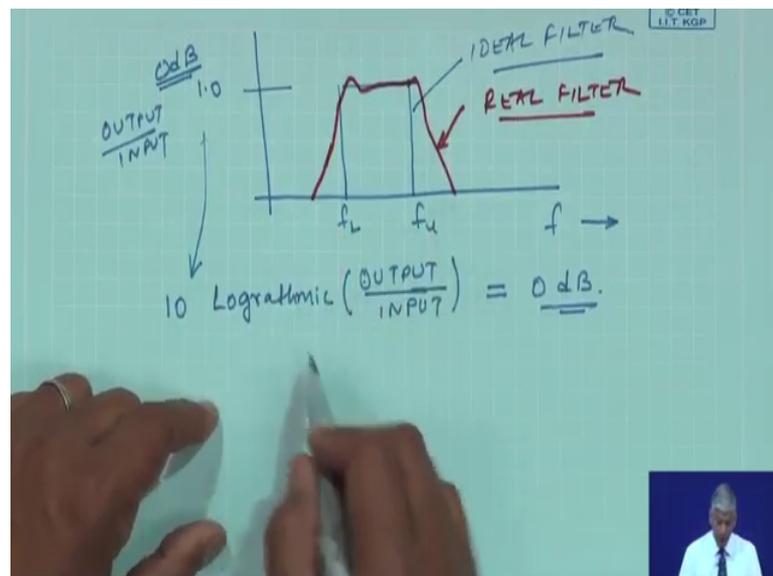
Sometimes this output by input is in the linear unit is given as one, but sometimes, they are also given in the logarithmic scale of output by input time is 10. So, this will be 0 dB. Now; obviously, this f L and f u are decided by the combination of R and C. So, the filters actually in an ideal filter actually it rolls off something like this is the real filter.

We will not go into the difference between real and ideal filter, but what it suffices us to say is filters are available of constant band width, I can decide on the range of the frequency of the filter. So, that it will allow only the signal to come in the output which fall in this filters bandwidth?

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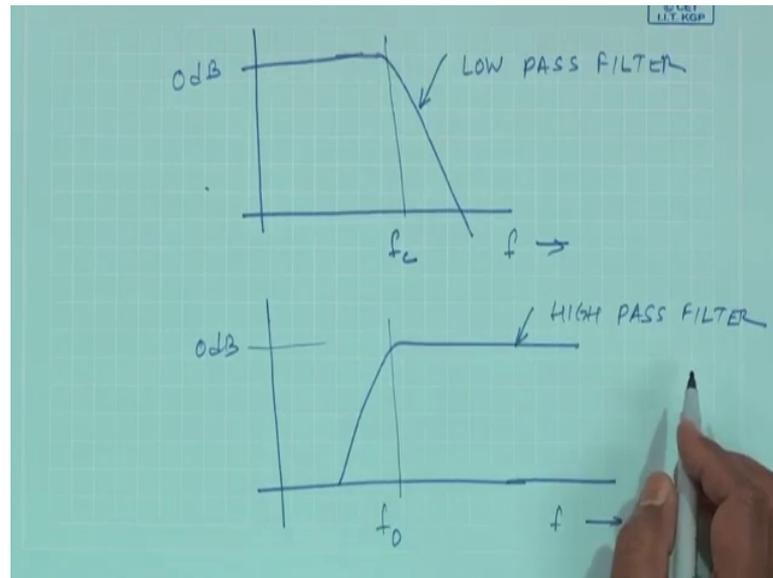
Now, these filters can be made as a combination of this kind of responses.

So, this is what is known as a low pass filter; that means, till the frequency f c, it allows every signal to pass through and after that it rolls it off and it truncates it down and this is known as f o, this is similarly a high pass filter. So, such analog filters are available for monitoring or for conditioning the signals and of course.

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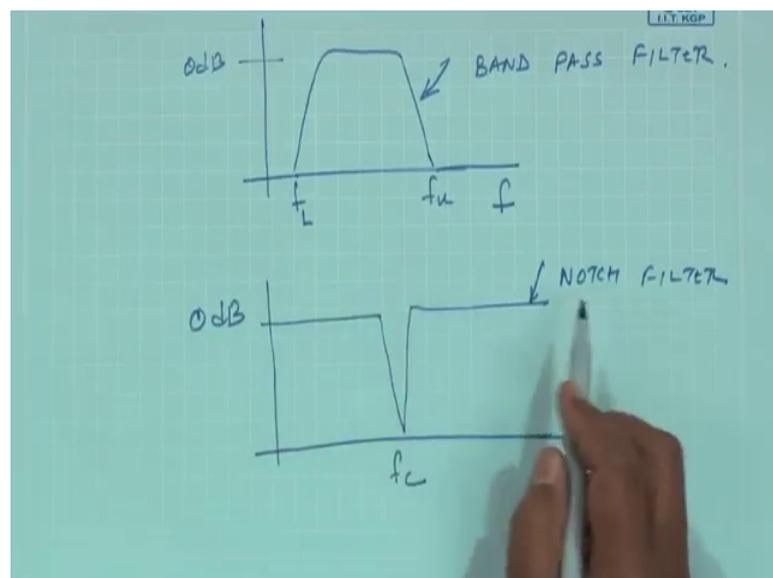
Now, we had the band pass filter and then we have a filter which removes a particular frequency is known as a notch filter.

I will talk about band pass filters litter later because I can have series of band pass filters and if you all realize to find out the frequency content in the signal, I could be passing them through a series of band pass filters where the band width is known to me. So, we will recall that the many a times, the analog filter sets are used to find out the frequency content in a signal, right.

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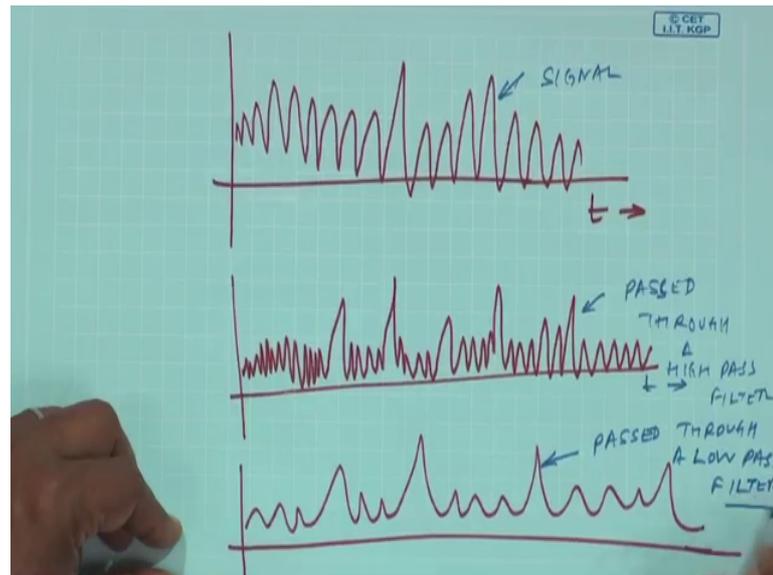
Now, I will give you a qualitative feel of what filtering does; for example, this is the signal. So, same signal; it will have some high frequency low frequency component, if I pass it through a high pass filter, what will look what it may look like is passed through a high pass filter, this my original signal and if it pass through a low pass signal, it may look like this low pass.

So, such analog filters are available of course, you know once we have a DSP dam.
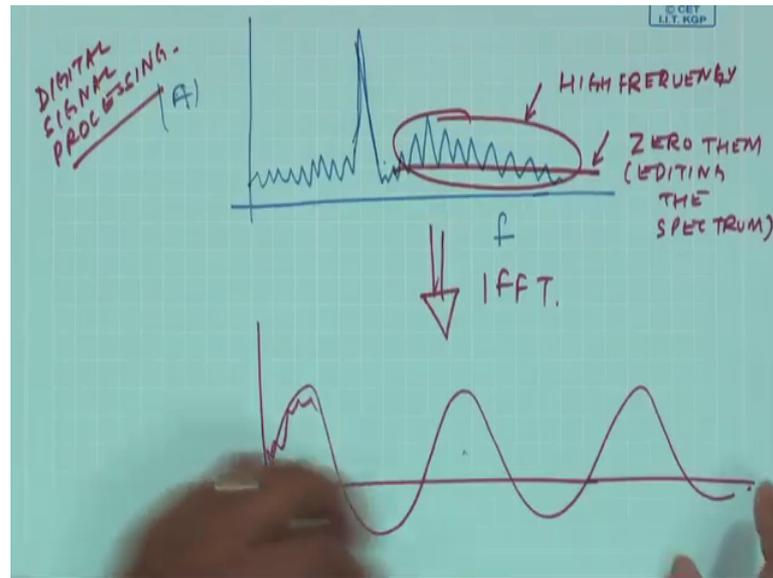
I can be using digital frequency techniques to do synthesis of the signal, for example, just to recollect in the FFT, suppose in a signal I have some amplitude, I have as you can see in the frequency spectrum these are all high frequency signal.

So, if I artificially 0 them by editing this spectrum 0 them, what is known as the editing the spectrum and then if I do an inverse Fourier transform which you all know by now, I will get back when this is the single frequency may be I will get back my signal, it be cannot be that neat, but there could be some variations. So, a noise has been removed from a signal the high frequency noise by editing this spectrum and doing an inverse for the transform..

So, this is in digital signal processing you can filter the noise out of a signal the reason we are studying this filtering is because in CBM, when we have transducers, when we have signals coming out of the transducers these effects do happen these issues do happen. So, by proper choice of the signal conditioning equipment by proper choice of setting the signal filters, we can eliminate some of this problem elementarily in real life many of the problems is this power supply frequency.
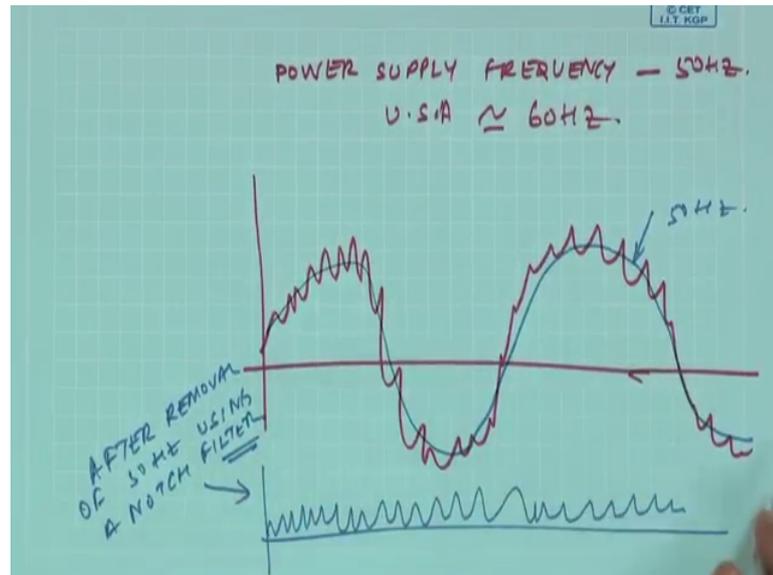
The electrical power supply frequency in India, it is 50 hertz, elsewhere abroad in US, etcetera, it is 60 hertz. So, these many of these electrical signals, you know, because of stray electromagnetic induction, there will be lot of high frequency, I mean the supply frequency modulating the actual signal. So, in such a case, I can use a notch filter, this could be my 50 hertz signal. So, I could use the notch filter to remove the 50 hertz signal and then I can get my actual signal.

After removal of 50 hertz using a notch filter and provided this can be done..
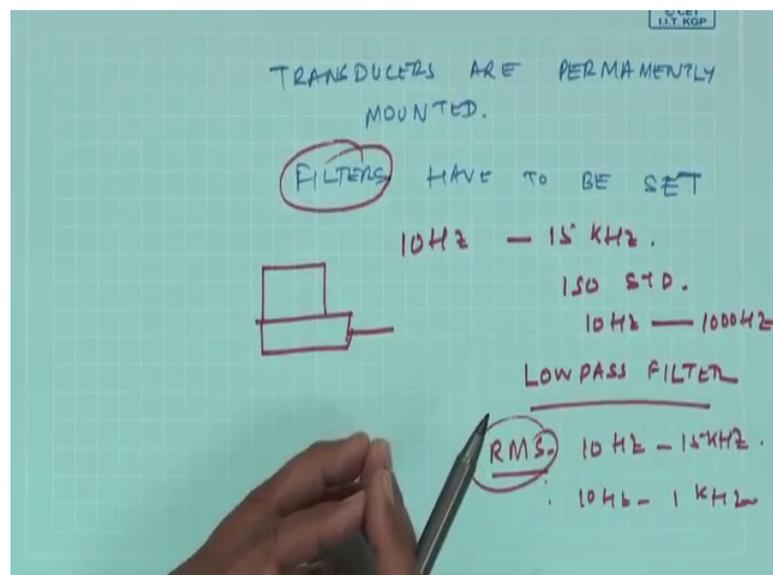
So, many a times when transducers are permanently mounted, the filters have to be set have to be set, for example, I will give you another example. For example, my transducer which I have measured which I have installed can measure all the way from 10 hertz to say 15 kilo hertz.

But if you recall the ISO standard on version measurement, it says an 10 hertz to 1000 hertz. So, I am not interested in anything beyond 1000 hertz. So, I could put a band pass filter or a low pass filter till 1000 hertz because if the low pass filter is not set and if you are measuring the RMS value in one case from 10 hertz to 15 kilo hertz whereas, it should have been 10 hertz to 1 kilo hertz, you will get a different value ok.

So, this one has to be very careful doing measurements then we take it for granted while we are doing measurements that you know whatever transducer is there we just get an output and see that it is this is what it is. It is exactly not because in CBM, we will be dealing with signals which are dynamic in nature signals with change with time. So, there is a strong frequency component in those signals. So, one has to be careful that what are the signal features we are measuring, we are in the frequency range of measurements which we are interested in that has to be done.

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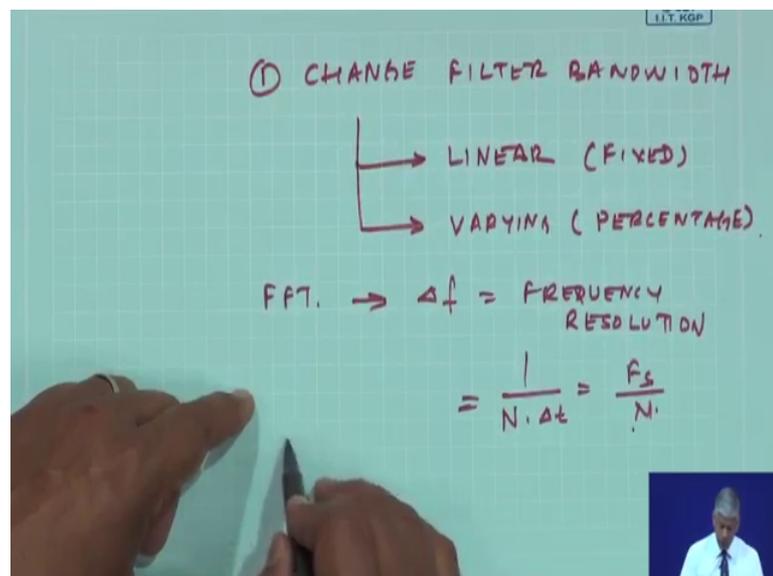
So, if you look at the filters, I can change the filter band width band width, they could be linear that is fixed or they could be varying or in other words fixed percentage or some constant percentage when we did FFT, we calculate delta f which is nothing that the

frequency of resolution and if we will recall this delta f is nothing, but 1 times N times delta t that is F s by N. So, to get a very narrow band I need to increase N if F s was same.

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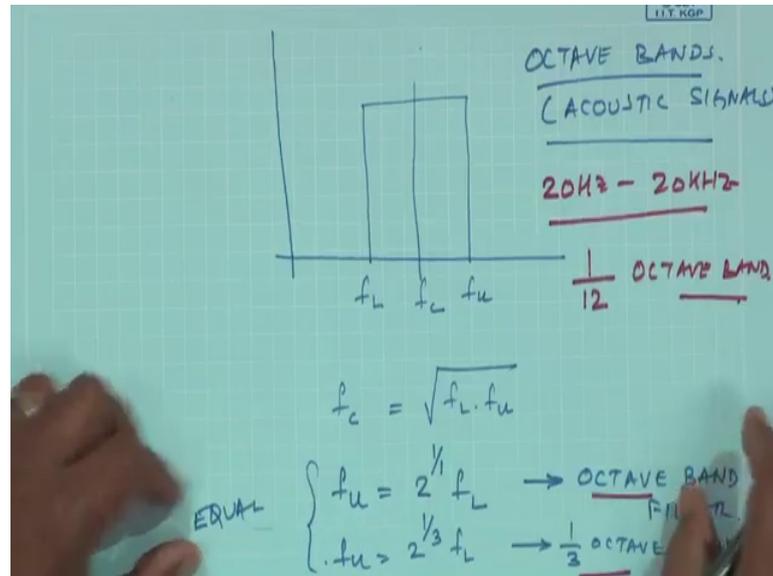
For signals there are cases in actually in acoustics wherein we will use band pass filters, but there is a strange relationship between the f lower and f upper and there is a center frequency f c. So, the f c is always the geometric mean of the f lower and f upper. So, this f upper is 2 to the power 1 by 1 a 2 times f lower, this is known as an octave band filter.

If f upper is 2 to the power 1 by 3 f L, this is known as one third octave and so on. So, if you will see the frequency band width in these cases is no longer equal and such octave band are used for analyzing acoustic signals, but we must have an idea about what this signal what this bands are and if you will go to any hand book you will see for the entire audio range from 20 hertz to 20 kilo hertz such band limits are there both for the octave band one third octave band. In fact, there is no end to it.

Nowadays, in a one by twelfth octave band etcetera are also available. So, you will realize the importance of signal filtering because see nevertheless whatever we measure.

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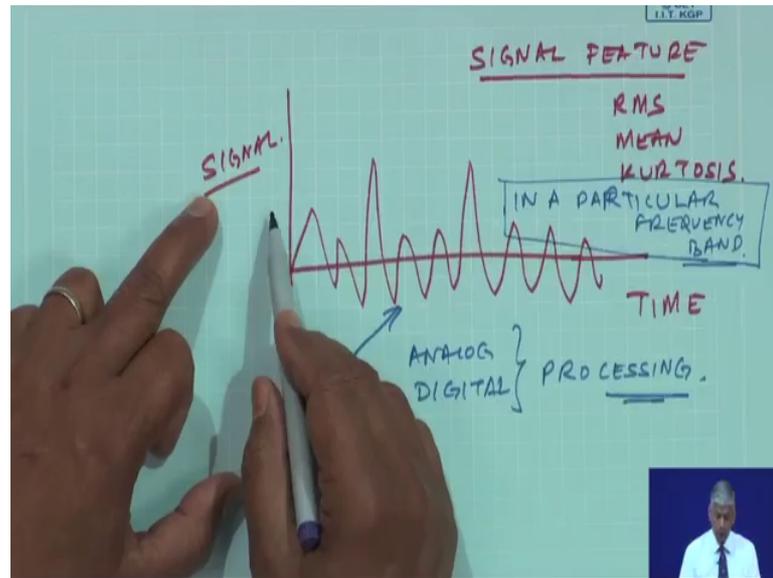
I need to measure the signal feature. Now if you will recollect, the signal feature could be; its RMS value could be its mean value could be its kurtosis value etcetera. But it all depends on what kind of signal I have. So, signal feature in particular frequency band is important sometimes you know as an end user we are not aware of it, but the signals which are been measured, they are actually band limited and then we are given this signal wherein we can find out the values of this signal depending on whether we do it an analog processing or digital processing the choice is ours.

But most important is whatever signal, I am measuring they are varying with time and so, we need to measure this RMS mean and kurtosis values in a particular frequency band and most important thing is when we talk about this measurement chain, what if I have measured in a frequency band and I am processing in different frequency band that can always happen right unknowingly.

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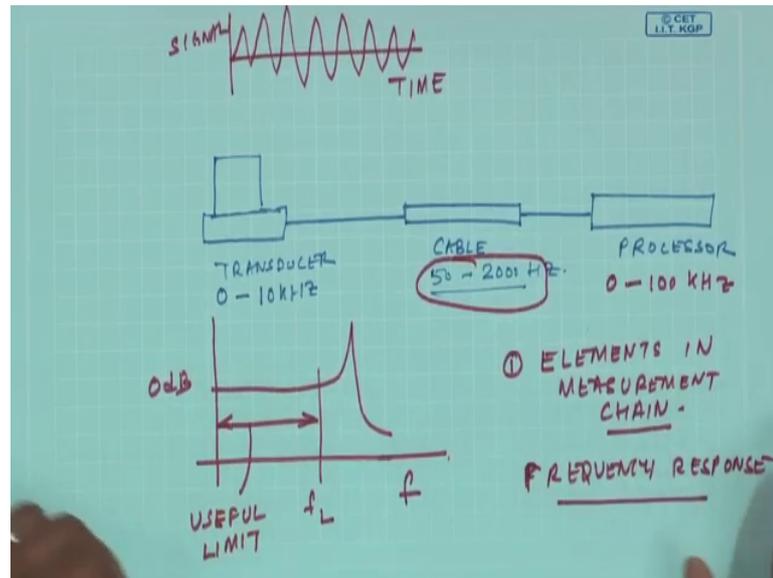
So, one has to be very very careful that when I have a transducer and I have a cable, I am drawing a cable like this because cable is also a system and then I have my processor see all of this systems and because I am dealing with dynamic signals are varying with time.

So, signals do have a frequency component a transducer has a certain mass certain stiffness transducer also has a dynamic response of the transducer. So, transducer has also a frequency limit up to till which the data which comes out of the transducer is of use same is to for the cable same is to for the processor. Now for the sake of discussion, suppose my transducer has a sensitive response or reasonable frequency range from 0 to 10 kilohertz and processor can measure anything from 0 to 100 kilohertz, but my cable can only measure only from 50 to 2000 hertz..

So, do you mean to say if I have bought a cable now if I bought a transducer which has a frequency response up till 10 kilohertz, all right and you say that my measurement is good till 10 kilohertz you are wrong because the cable is limiting your output because cable is only good from 50 to 2000 hertz. So, this is something one has to be very careful about when we have elements in measurement chain with different frequency response.

So, if you will see a signal conditioner we need to have good settings on the filters, there may be you know all this filters may not be available in a single signal conditioner, but definitely a low pass or high pass filter is there and sometimes a notch filter is there I mentioned about octave band filters that is just from the closed processing at some octave band analysis could be done.

But another very important aspects in signal conditioning is this amplification if the signal strength is less than the least count of the sensing element, I need to amplify it, I

had given you the example of a thermocouple signal. Now if I had talked about an 3 bit data efficient system and the amplitude resolution for a 10 volt input will be 10 divided by 2 to the power 3 that is 10 by 8; 1.25 volt.

So; that means, my thermocouple output suppose it is you know 0.125 volt. So, such an output cannot be sensed by the DAQ system, unless it is amplified by a factor of 10. So, this 10 x means a gain of 10. So, sometimes a signal amplification is required. So, that it is being sensed by an unit which least count is more than the signal strength, all right.

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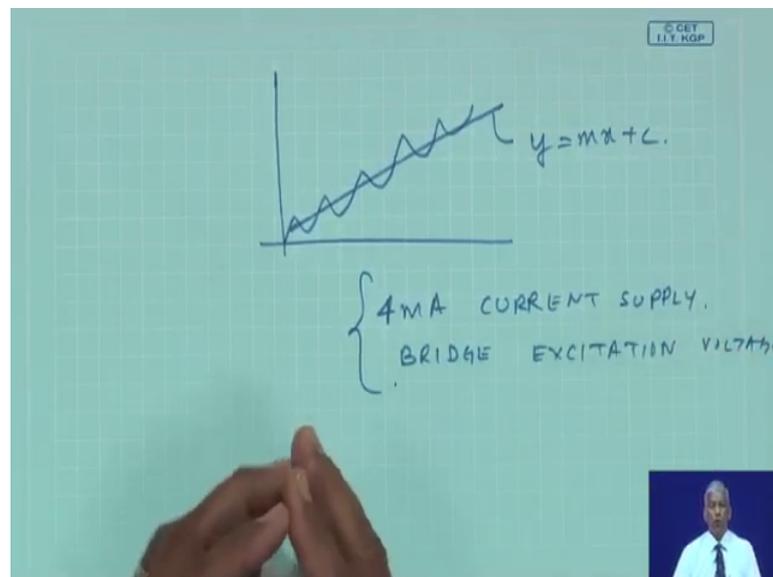
And another important aspect of amplification conditioning is signal linearization and this could be done through a DSP, I can fit a curve y is equal to m x plus c, in some of the signal conditionings, we require, I will when we talk about transducers you will see that some of these transducers require a 4 milli ampere current supply or like in the case of a strain gage bridge excitation voltage.

So, these also need to be provided by the signal conditioner which is connected to the transducer.

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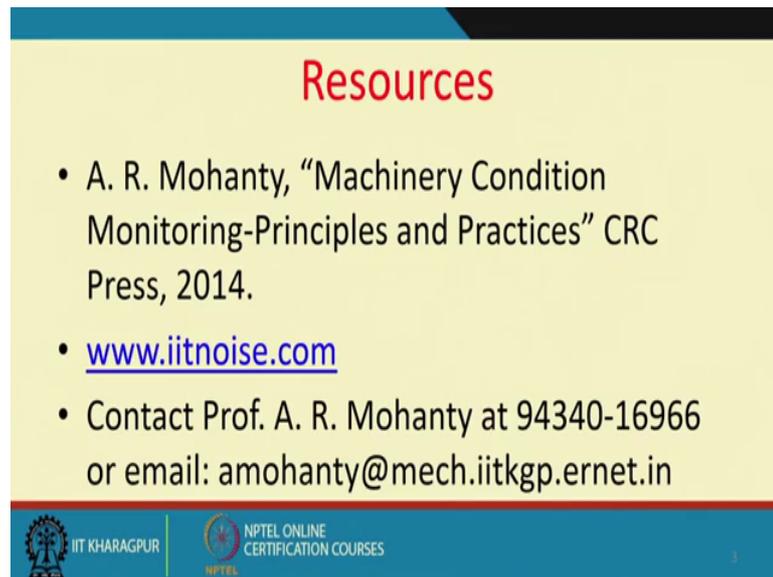
So, that we have a meaningful output from the transducer. So, more on this you can find in my book on machinery condition monitoring principles and practice.

Thank you.