Millimeter Wave Technology.
Professor Mrinal Kanti Mandal.
Department of Electronics and Electrical Communication Engineering.
Indian Institute of Technology, Kharagpur.
Lecture-22.
Passive Components (Contd.)

So next topic is filter at micro wave and millimetre wave frequencies we use couple theory to design filters and the method we use it is called insertion loss method.

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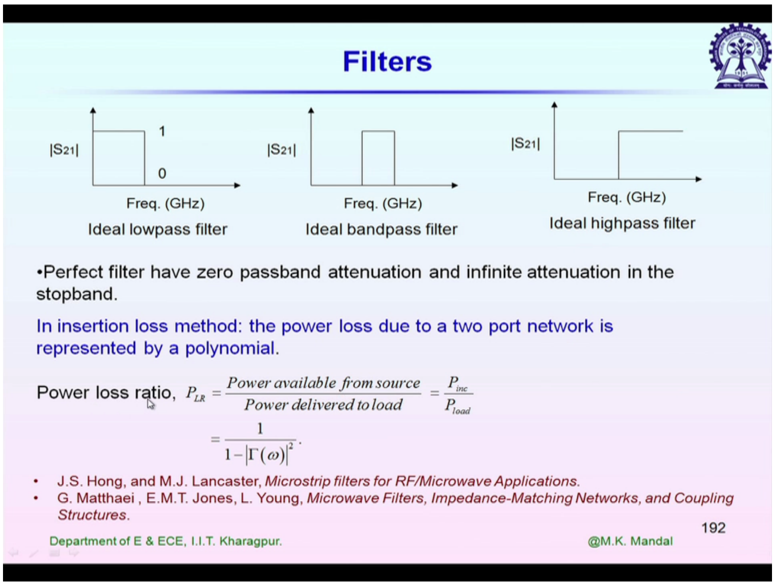

So in this method what we do power loss ratio we define it by using a umm a polynomials so here power loss ratio it is equal to power available from source divided by power deliver to load. So that is equal to P incident divided by P load. If I consider normalised power values in that case we can consider P incident that is equal to 1 and divided by this is 1 minus magnitude of gama omega square.

So gama it is the reflection coefficient and if I consider 1 minus gama square it will represent the power deliver to load now what we do in this method we actually change the input impedance or input impedance it becomes a function of frequency and that will be vary umm according to our requirement. For example if we have to design 1 bandpass filter then in band or in pass band it will have good impedance matching with the source and load.

And out of band the impedance mismatch it will be as high as possible and it is done by changing the (())(1:59) parameter of this component whatever we used for filter design.

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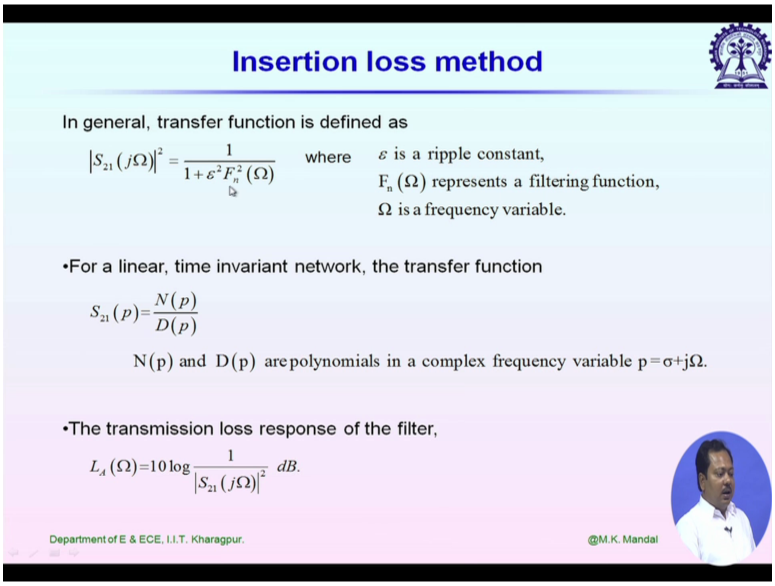

So then above the S parameters transmission parameter is 21 which is a function of frequency we can write down the transfer function in terms of this S 21 as magnitude of S 21 square this is equal to 1by 1 plus epsilon square into FN square here capital omega. It is a frequency variable so for now you can consider capital omega is equivalent to small omega and epsilon it is repel constant now we define different function for this Fn accordingly we name the filter in different names like butter hot filters. (())(2:54) filter, electric filter.

It can be shown that for a linear time in variant network the transfer function it 21 it also can be represted by ratio 2 polynomials N function of P divided by D function of p so where this NP and DP they are polynomials in a complex frequency variable P is equal to sigma plus J capital omega. Then we can define transmission loss response of the filter if we have a S 21 from that we can calculate what is the loss?

Loss its a positive quantity so this is equal to 10 log of 1 by magnitude of S 21 square in decibel so once we have transmission loss which represents the loss power loss from port 1 to port 2 for a 2 port network.

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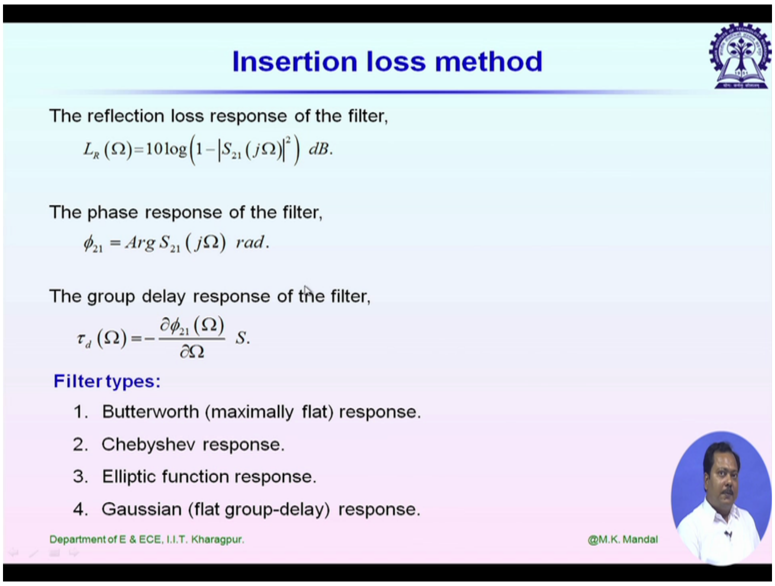

Then we can easily calculate reflection loss for a loss less network so LR reflection loss this is equal to 10 log of 1 minus S 21 square so for loss less network is 11 square plus S 21 square that is equal to 1 so from using that relationship we are calculating reflection loss here. Phase response of the filter phi from port 1 to 2 that is equal to the angle value of S 21 again its a function of frequency and the phase response is very important in particularly white band applications. Why?

Because the delay of phase represents sorry the slope of phase represents group delay it can be represented as tou D this is equal to minus del del omega of Phi 21. Now this is the time taken by your signal to reach port 2 its a function of frequency that means this time it varies with frequency so if it varies more in that case we will be facing pulse distortion and the medium it becomes dispersive so there should be a minimum allowable value for this time delay variation over the band to keep that distortion minimum.

Now we define different types of filter by choosing different types of polynomials. They have their own properties the most popular one are butterworth or maximally flat response. Second one is chebyshev response then elliptic function response and Gaussian or flat group delay response.

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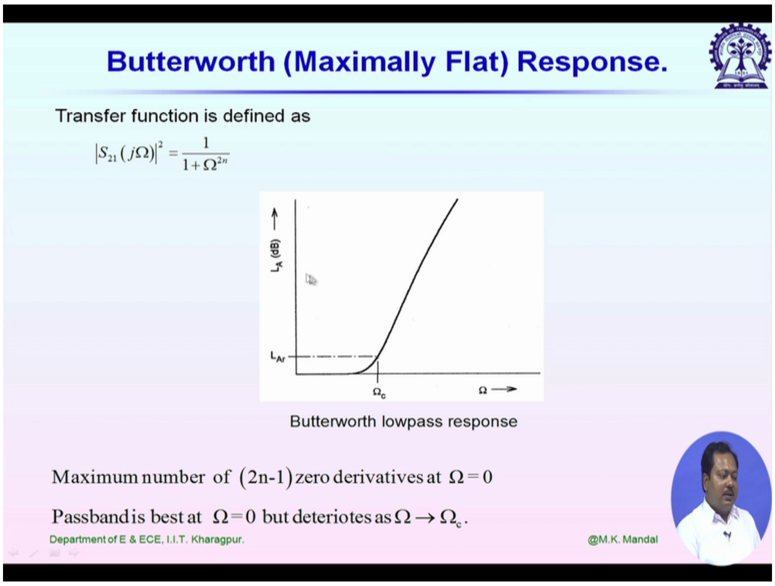

So let me discuss about one of this so whatever response for this one you can see S 21 square transfer function it is represented by 1 by 1 plus capital omega to the power twice N. And if I plot loss which was defined by this relationship LA this is equal to 10 log of 1 by S 21 square. So if I plot that we will be having a low pass type response you can see here so at capital omega equal to 0 loss is 0 and after that it is increasing.

It has the property that maximum number of twice N minus 0 derivatives at gama at omega equal to 0. So that means here for a loss less system we have magnitude of S 21 equal to 1. And from this point if I increase capital omega then loss its slowly increases and after omega C let us call it the cut off frequency omega is (())(7:09) is defined where we have a specified loss value L AR after this loss increases rapidly so this is for a low pass filter we can design band pass filter by using low pass to band pass transformation.

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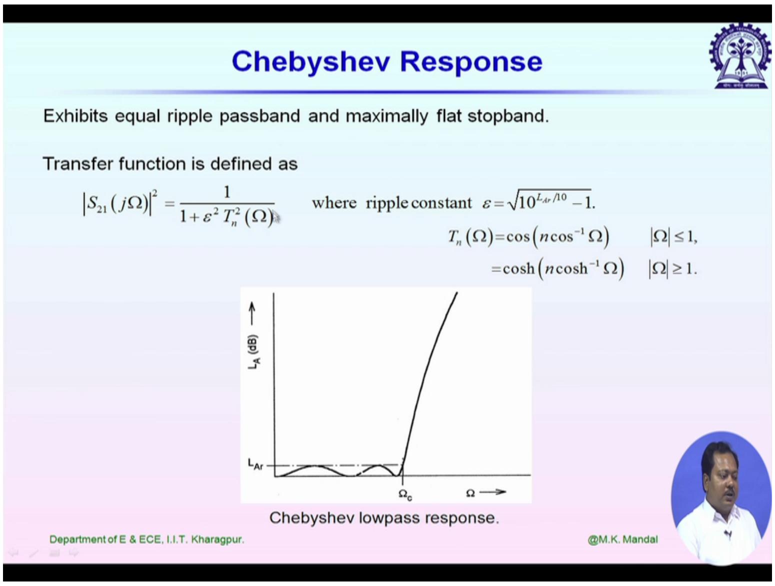

Next another example this is for cheby chebyshev response you can look at the plot we have ripple in passband and in stop band we don't have any ripple. So it exhibits equal ripple passband and maximally flat stop band. And the transfer function is defined as 1 by 1 plus epsilon square into TN square where epsilon this is called the ripple constant it can be given by square root of 10 to the power LAR by 10 minus 1.

LAR this is the specified loss value for example you can define your cut off frequency omega C by 3 DB in that case LAR that is equal to 3 DB you can define your cut off frequency by lets say point 1 DB in that case LAR that is equal to point 1 DB so TN this is cosinocydal function or cos hyperbolic function depending on the omega value. Now if I compare this chebyshev press forms with the previous one.

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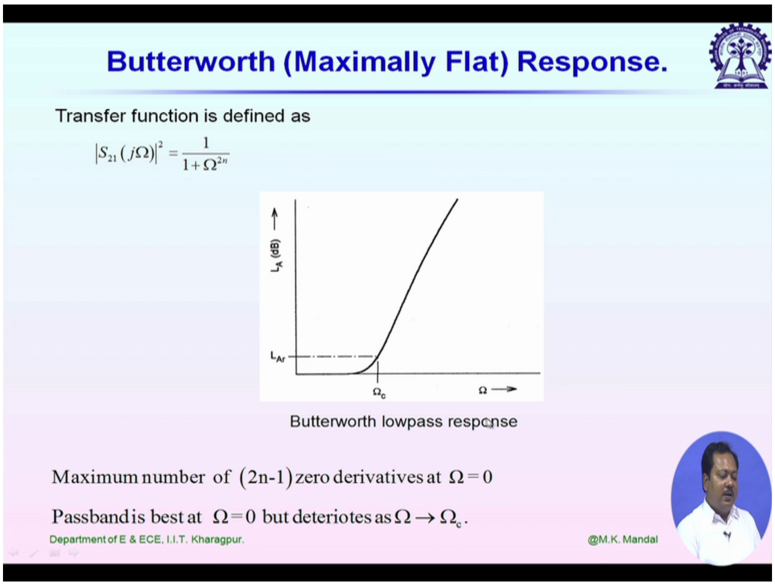

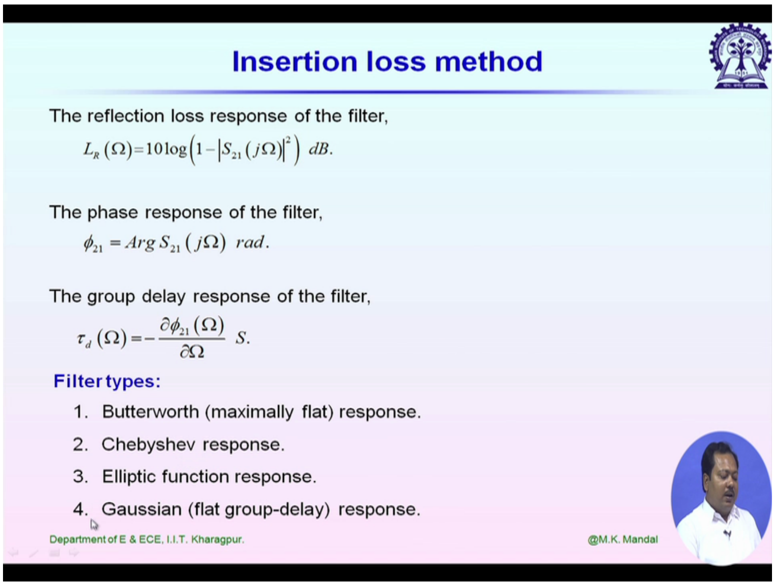

Butterworth lowpass filter response so you see in pass band for butterworth we almost don't have any variation where as for the chebyshev we have ripples in the passband and for the same order we will discuss later we have higher attenuation slope for chebyshev response. So if attenuation criteria are more prominent than the passband criteria in that case we can choose chebyshev lowpass prototype response.

But for butterworth polynomial we have one advantage if we plot the group delay variation it will have lower variation compare to chebyshev one. So for wide band applications then butterworth should be preferred if we want to avoid any dispersion so in this (())(9:43) you can use Gaussian response which will have a flat group delay.

Theoretically no group delay variation over the whole bandwidth but its attenuation responses poorest among all this 4 and this third one umm electric function response it shows ripples both in passband as well as in stopband and it is having maximum group delay variation inside the bandwidth.

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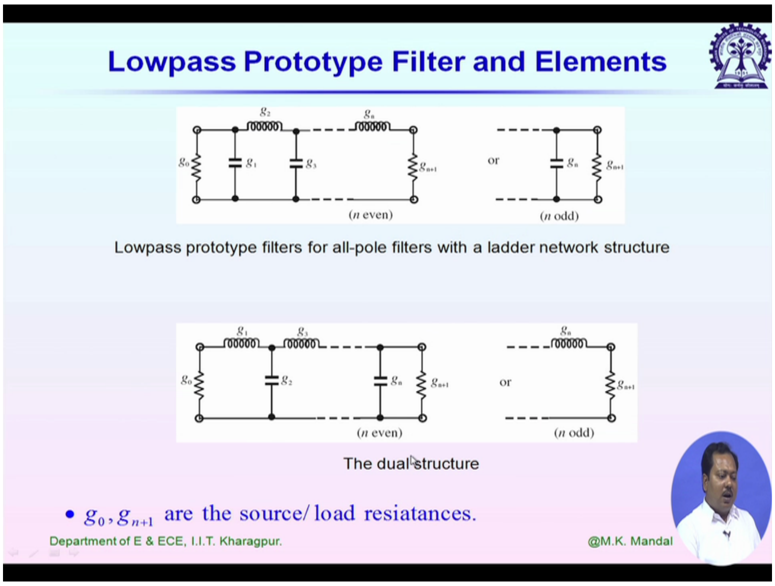

So next how to realize this functions? So we start with lowpass prototype so considering you see if I asked to design a lowpass filter using (())(10:31) like capacitor or inductors how will be connecting so capacitor usually we use in shunt configuration so high frequency component will pass through capacitor and inductors will be using in series configuration so it will attenuate frequency components so the basic lowpass filter configuration it can be a pi type or a T type configuration so in this case we are showing both of this 2 different types.

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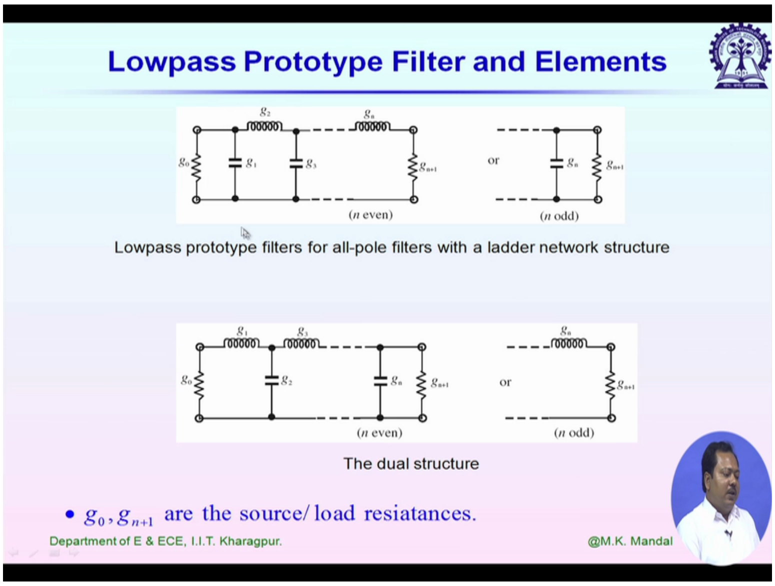

In the first type where we are using a basic pi type you can see2 capacitors separated by 1 series inductor now this elements usually they are represented by their normalised values we call simply G parameters G1, G2, G3 if N is given then last element will be one inductor and if N is odd in that case the last element will be one capacitor.

And you can see G nought and G N plus 1 they represents shorts impedance and the load impedance represented by 2 resistors we have develop this in this case in place of pi we are using a T network 2 inductorG1, G3 and one shunt capacitor given by G2. Now if N is even in that case last element is a capacitor and if N is odd in that case last element is one inductor.

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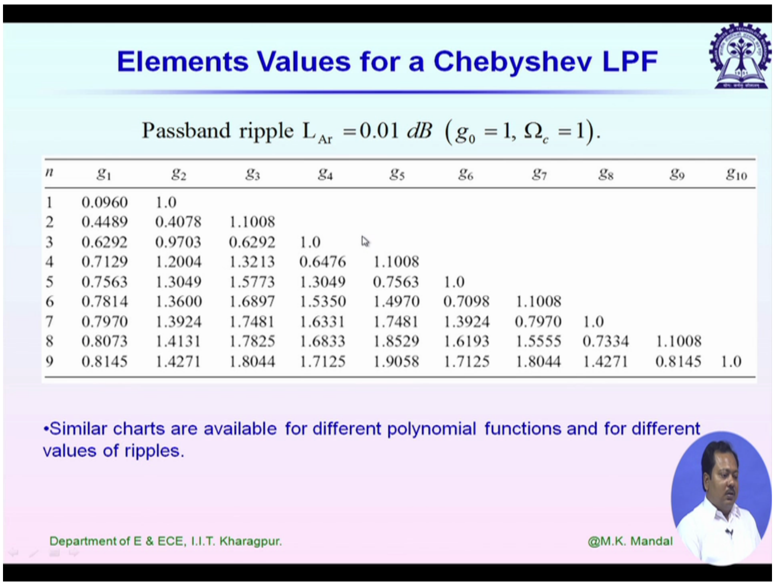

n	g_1	g_2	g_3	g_4	g_5	g_6	g_7	g_8	g_9	g_{10}
1	0.0960	1.0								
2	0.4489	0.4078	1.1008							
3	0.6292	0.9703	0.6292	1.0						
4	0.7129	1.2004	1.3213	0.6476	1.1008					
5	0.7563	1.3049	1.5773	1.3049	0.7563	1.0				
6	0.7814	1.3600	1.6897	1.5350	1.4970	0.7098	1.1008			
7	0.7970	1.3924	1.7481	1.6331	1.7481	1.3924	0.7970	1.0		
8	0.8073	1.4131	1.7825	1.6833	1.8529	1.6193	1.5555	0.7334	1.1008	
9	0.8145	1.4271	1.8044	1.7125	1.9058	1.7125	1.8044	1.4271	0.8145	1.0

So we have actually different tables available for this G values for different LAR specified LAR for different types of polynomials here for example we are showing the element values for a chebyshev lowpass filter where it consider that LAR equal to point 01 Db.

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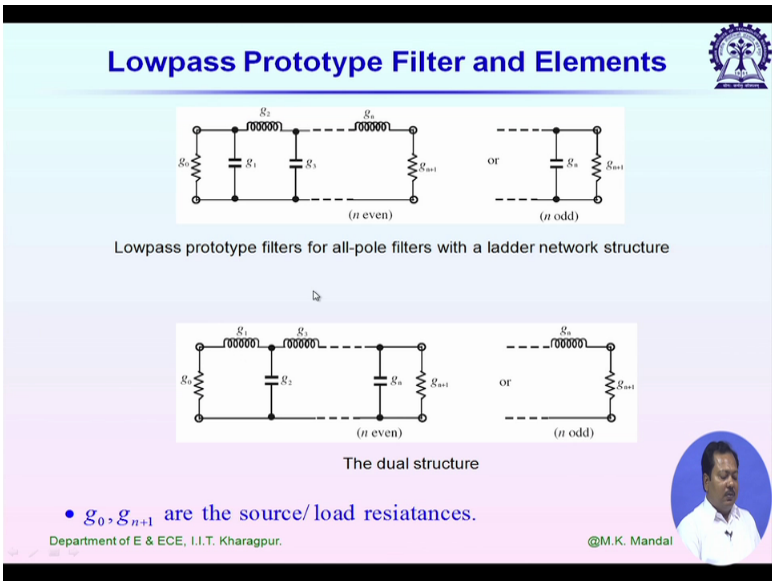

Okay let me discuss one point how to define this LAR better let me draw it.

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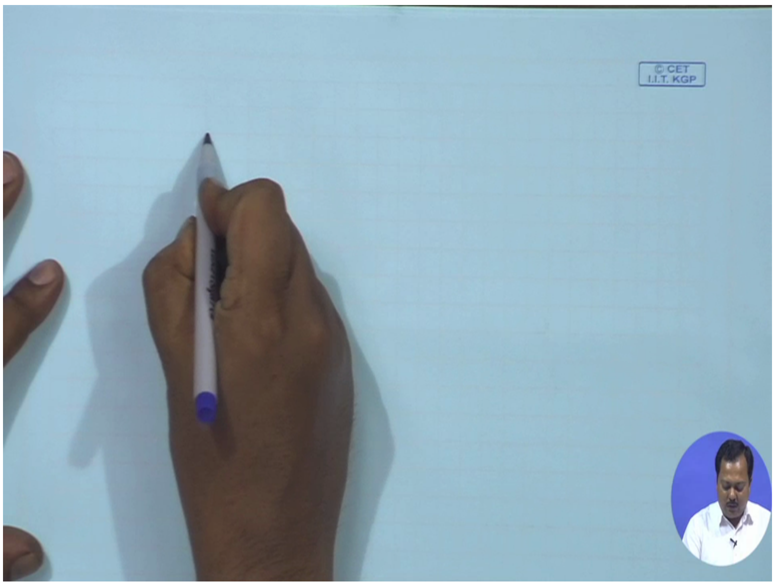

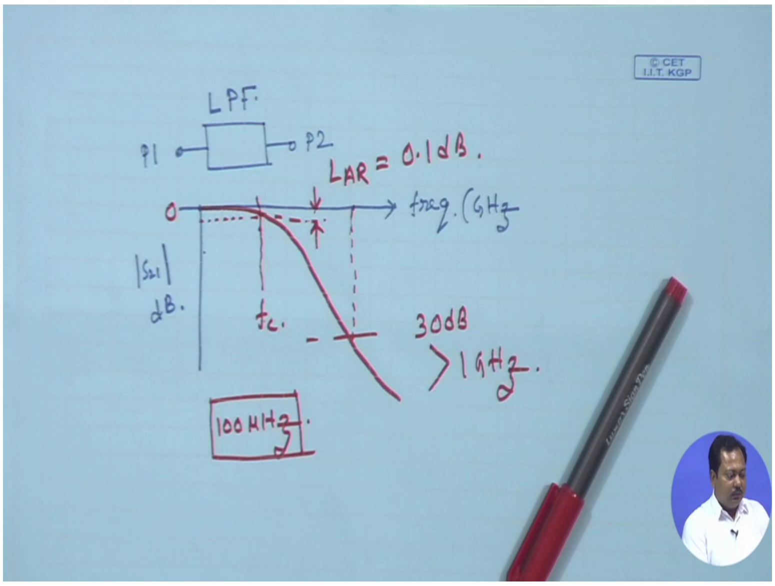

So lets say we have a 2 port network this port 1 this port P2 and it is behaving as a lowpass filter now if I plot S 21 versus frequency so along X axis I am representing frequency in gigahertz and along Y I am representing magnitude of S 21in DB. Now if we have perfect transmission from port 1 to port 2 so that means S 21 in Db it is 0 DB and since we are considering lowpass filter as the frequency increases in that case attenuation increases or S21 in DB it becomes negative goes to minus infinity.

Now how to define the cut off frequency lets say I am using this lowpass filter to separate the IF and my IF requirement is 100 megahertz so IF bandwidth minimum we need 100 megahertz so whatever signal we have it is in this 100 megahertz we don't want any attenuation for this 100 megahertz signal so we can define LAR lets say point 1 DB or point 01Db and it should have at least a minimum value FC of 100 megahertz so LAR then we define LAR this is equal to lets say point 1 DB.

Now in filter applications many times we simply define 3 DB cut off frequencies but not here if I define my 3 DB cut off frequency at FC equal to 100 megahertz that means already we I am assuming that my 50 percent power near the cut off frequency it is being attenuated by the lowpass filter which I don't want so when I define LAR for this cut off frequency it should be much lower.

Now to have some specified attenuation at some given frequency you can define 1 more cut off frequency lets say for this application we need at least 30 DB attenuation above FC equal to 1 gigahertz. So in that case you can define another cut off frequency. But you remember whenever we are going to use this charts this table we first consider what is the signal bandwidth and accordingly you can choose LAR equal to lets say point 1 DB or point 01 DB or point 01 001 DB anything.

Now after fabricating this filter so whatever bandwidth I am defining lets say point 1 or point 01 it can be measure. Because of fabrication tolerance and also umm accuracy problem of VNA this S21 it will vary inside the fast band and this variation will be more than lets say point 01 DB. So thats why it is really difficult to measure that point 1 DB or point 01 DB bandwidth of a filter and practically we measure 3 DB cut off frequencies it is easier to measure.

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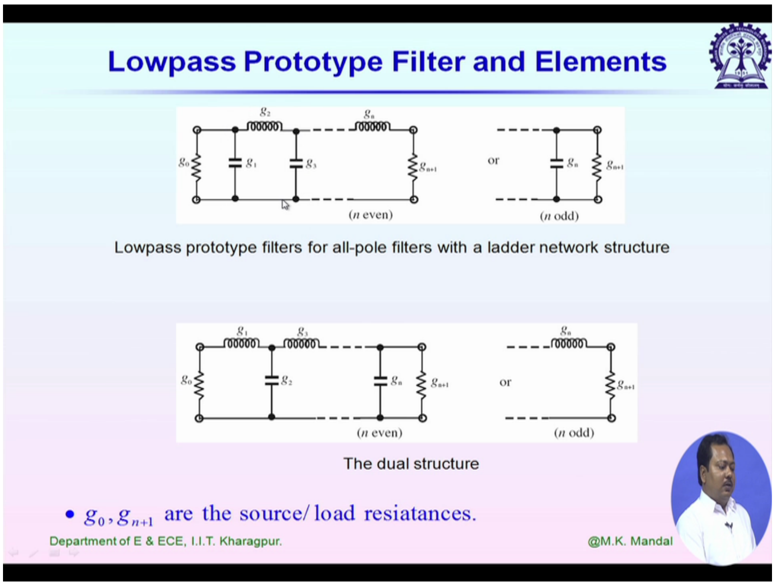

So now lets go back now you see in this lowpass prototype filters if I use 1 single pi section obviously attenuation slope attenuation will vary slowly so if I want to improve this curve selectivity that means the attenuation roll of. In that case we have to increase number of section or we have to increase the order of the filter. So we have to use more number of elements.

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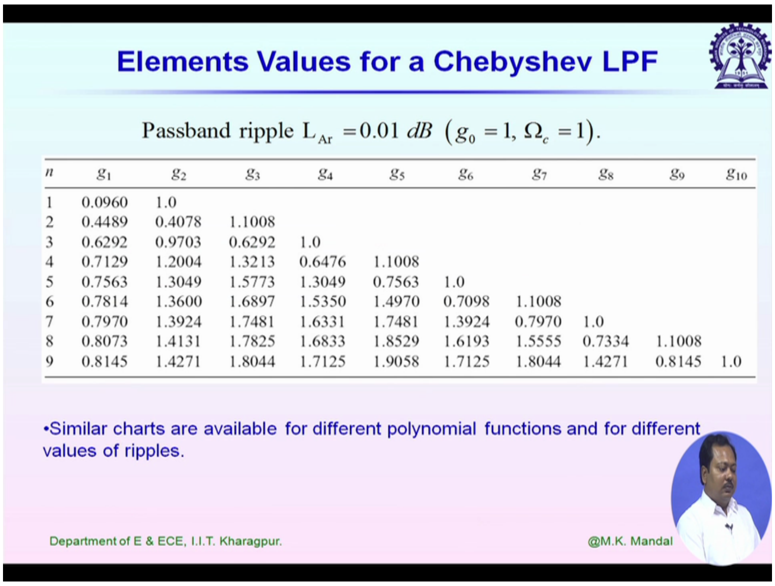

This table it shows till G equal to 10 G nought it is they are all normalised values G nought this is equal to 1 and omega C its considered as 1. Then for example lets say if I consider N equal to 2 we have G1 equal to point 4489. G 2 equal to point 4078 and N plus 1 that load point 1.1008 similarly depending on your attenuation requirement you can choose the order of the filter for example lets say you decided your order of the filter at least it should be N equal to 7 then you can choose this G parameters.

This particular row now we have to realize this G parameters or L and C values so similar types of table available for different LAR values and also for different polynomials. Now practical filter design, so already lets say we have the required G parameter we have to now find out then what is the actual L and C values. Because the G parameters those are normalised values given in the chart. So for that we have to use some sort of impedance transformation.

Next step we have used omega C equal to 1 that is again normalised value so we have to use 1 transformation for our required FC so the example which I was showing FC equal to 100 megahertz.

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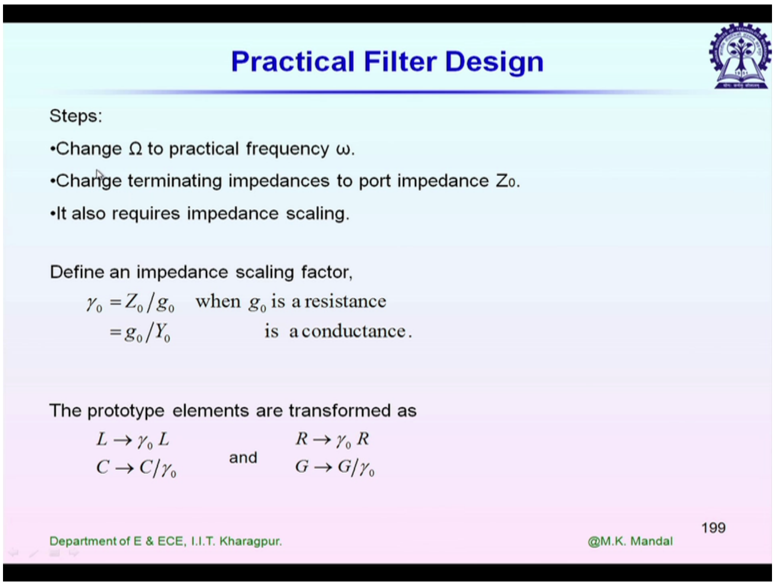

So then the practice steps you first change capital omega to practical frequency small omega. Then the change terminating impedance to port impedance Z nought previously it was shown by G nought and G N plus 1. It requires obviously impedance scaling. So how we define impedance scaling factor gama nought this is equal to Z nought by G nought when G nought is a resistance and it is equal to G nought by Y nought when its a conductance.

So typical Z nought values for any millimetre wave system or micro wave system 50 ohm. Next we have to find out the real LC values. The prototype elements are transformed as L this is gama nought equal into small L so it is actually that G parameter then capital C equal to small C by gama nought again this right hand side C this ih the normalised G parameters. Similarly we can find out the real capital R and capital G all the right side they are the G parameters.

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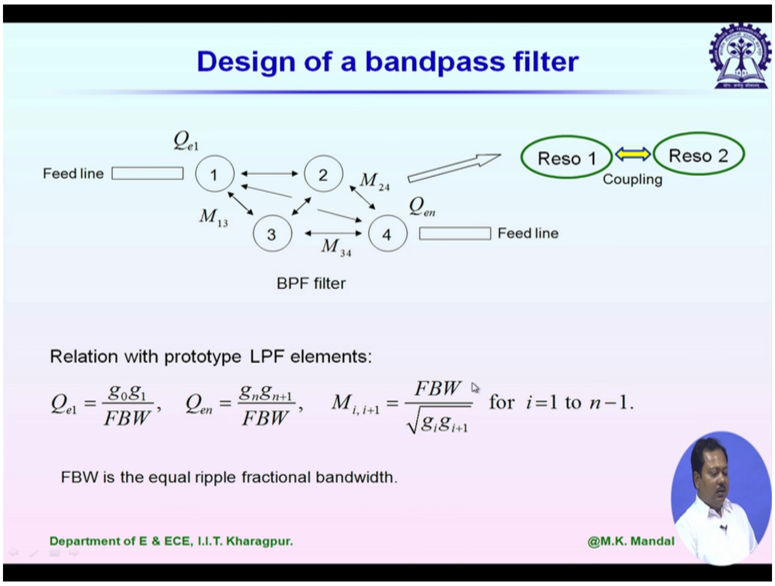

Let us consider then how to design a band pass filter directly so for band pass filter design we use resonator coupled resonator systems so obviously it will be a 2 port network port 1 and port 2 they are connected to my millimetre wave systems which is a (())(20:44) system so in simulation we can represent then port 1 and port 2 by the equivalent resistance 50 ohm and then power should be coupled from port 1 to resonator so we have to umm study this coupling mechanism in details and how the filter design steps it can be represented in terms of this coupling.

So what we do? The coupling from port 1 to fast resonator or umm port 2 to last resonator we represent it by quality factors. We call it external quality factor so let me show you in diagram.

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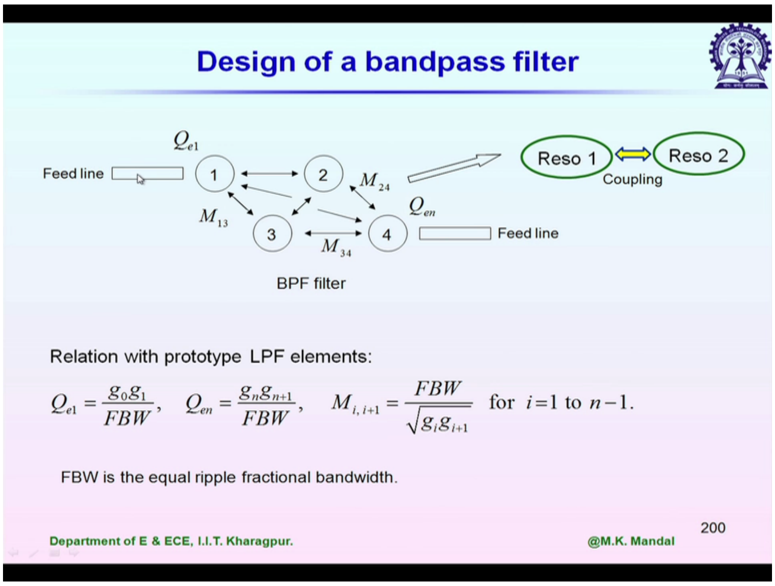

So here you can see the input port input feed line and the output feed line we are considering a 4 resonator systems. Also these are conceptual diagram then the power transfer from feed line to resonator 1 it is being represented by external quality factor QE1 and for the (()) (21:57) resonator to feed line it is being represented by QEN and then the coupling between 2 resonators. It is represented by coupling coefficient.

I will discuss later what is coupling coefficient and how we can extract the values by using any full wave simulator. This coupling it can be electric coupling due to the electric field it can be magnetic coupling or even it can mixed coupling. So these are complex situations and the relationship with the prototype LPF element. I am showing you directly the values. QE1 we can relate it to that G parameter this equal to G nought G1 divided by FBW.

FBW represents the fractional bandwidth so if you have the bandwidth then simply bandwidth divided by this mid band frequency it will give you fractional bandwidth. QEN this is the external quality factor for the second port this GN into GN plus 1 divided by FBW. And the coupling between I and I plus 1 resonator that is a function of fractional bandwidth and the G parameters so FBW divided by square root of GI multiplied by GI plus 1where I it varies from 1 to N minus 1.

So you see then if we need higher fractional bandwidth or for white band design we have to obtain higher coupling.

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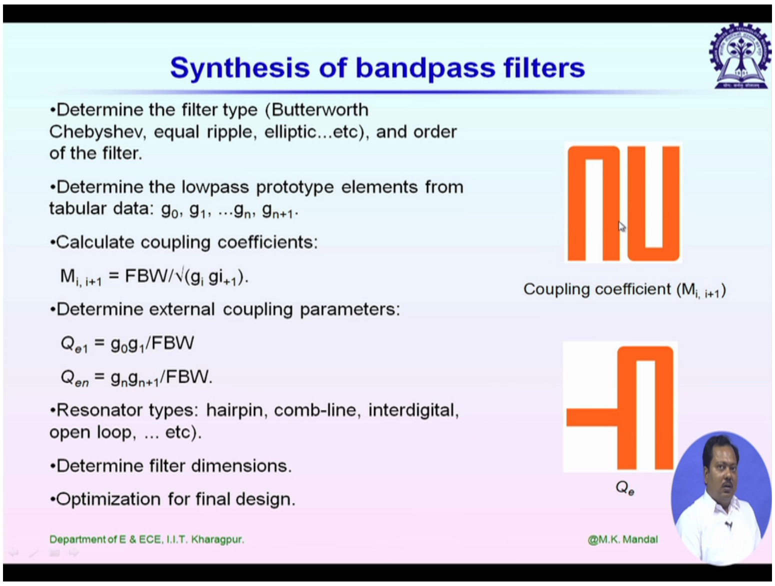

So for any given specification then how to implement any filter? So this is we call synthesis tapes first we have to choose the type of polynomials depending on the requirements fastband lets say group delay variation. Fastband is 221 variation then stop band attenuation requirement curve selectivity depending on all this criteria we can choose the type of polynomial it can Be anything butterworth chebyshev, electric or a short group delay or Gaussians response.

So if nothing is specified you can select chebyshev response most probably. Because chebyshev it is somewhat in between butterworth and electric. Then determined the low pass prototype elements from that tabular data that means the G parameters then calculate the coupling coefficient so you have to determine number of resonators from umm order of the filters then the required coupling coefficients that means the coupling matrix basically we have to synthesis so M I.

And I plus 1it is already defined we discussed previously then determine the external coupling parameters choose the type of resonators so it can be of different shapes and implementation also it can be in micro strip line it can be in lets say image guide, it can be in rectangular wave guide anything. So depending on again power handling capability, capacity depending on loss requirement we have to choose the proper umm implementation scheme.

Next determine filter dimensions so this filter dimensions we have to use some sort of full wave simulator to realize this filter and than the optimisation for final design.

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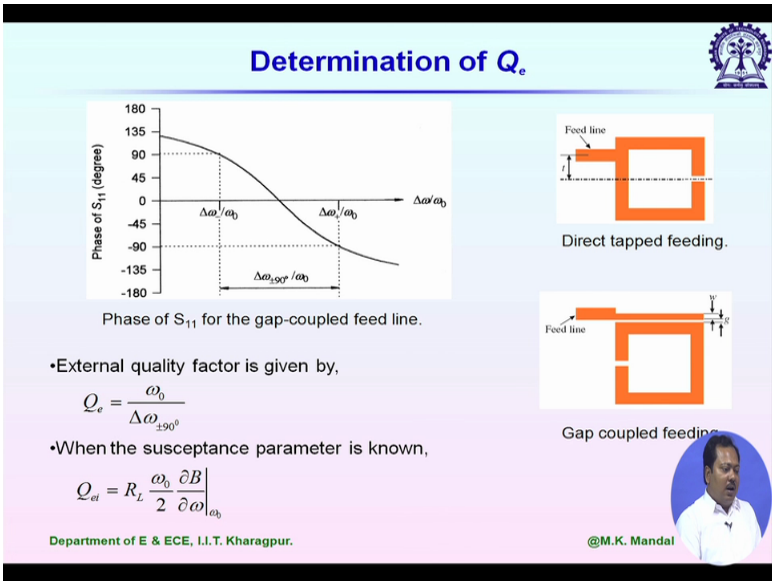

So its a lengthy procedure and it starts with the QE and K calculation. So here what we will do? We will assume that we have the required G parameters and from that already we have calculated the external quality factors and the coupling matrix. So next we will discuss then how to implement this filter after a short break. Thank you!