Integrated Circuits and Applications

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Active Filters II
Lecture - 21
All Pass Filter

Okay, In the last lecture we have discussed about the narrow band reject filter which is called notch filter. So, the second type of band reject filter is wide band reject filter. So, how to design the wide band reject filter? Wide band pass filter we have designed by cascading the low pass filter and high pass filter. Whereas, to obtain this wide band reject filter instead of cascading this low pass and high pass filter we have to add the outputs of the high pass and low pass filter. So, that can be explained with the help of the ideal frequency response. This is the ideal frequency response of high pass filter.

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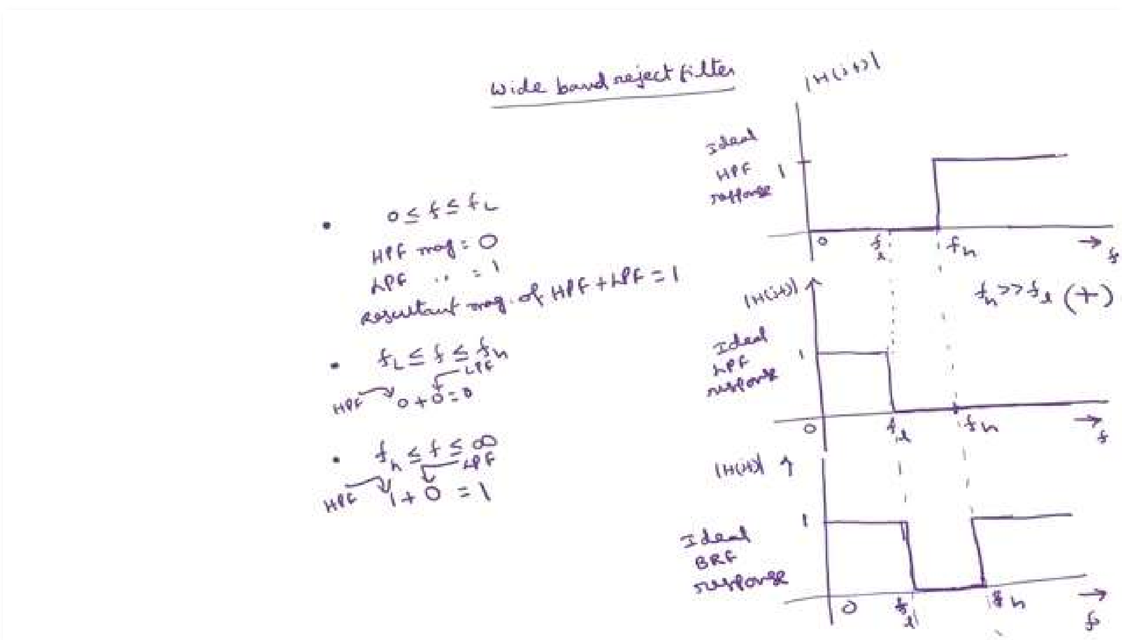

Say f_h is the cutoff frequency from 0 to f_h gain is 0 and f_h to infinity gain is unity. This is modulus of $H(jf)$ as a function of f. Say ideal HPF response and what will be ideal low pass filter response? It allows up to 0 to a particular frequency.

Say f_l is the cutoff frequency of ideal low pass filter and from f_l to infinite this will reject. So, here you have to choose that the cutoff frequency of high pass filter which is f_h should be greater than f_l. Then only you can obtain the response of band reject filter. So, if I add these two, if you perform the addition of these two. Then if I take these responses, this is f_l, this is f_h, this is 0. Between 0 to f_l, what is the value of high pass filter? is 0, between 0 to f_l. HPF magnitude is equal to 0, LPF magnitude is unity. If I add these two resultant magnitude is 1. So, here we will be having unity up to f_l. From f_l to f_h, f_l to f_h this is f_l, this high pass is 0, low pass is also 0. So, 0 plus 0 is 0.

So, this response will be having 0 response between f_l to f_h. f_h to infinity, what is low pass? This is high pass, this is low pass. f_h to infinity this is unity, whereas this one is f_h to infinity, this is f_h, this is 0. So, $1 + 0$, 1 is the response of high pass filter, 0 is the response of low pass filter, the resultant is unity. So, this will reject the frequencies from f_l to f_h, whereas it allows all the frequencies outside this range.

So, this response is nothing but the ideal response of band reject filter response. So, in order to construct a wide band reject filter, we have to add the outputs of the high pass filter and low pass filter with a condition that the cutoff frequency of high pass filter should be greater than that of the low pass filter.

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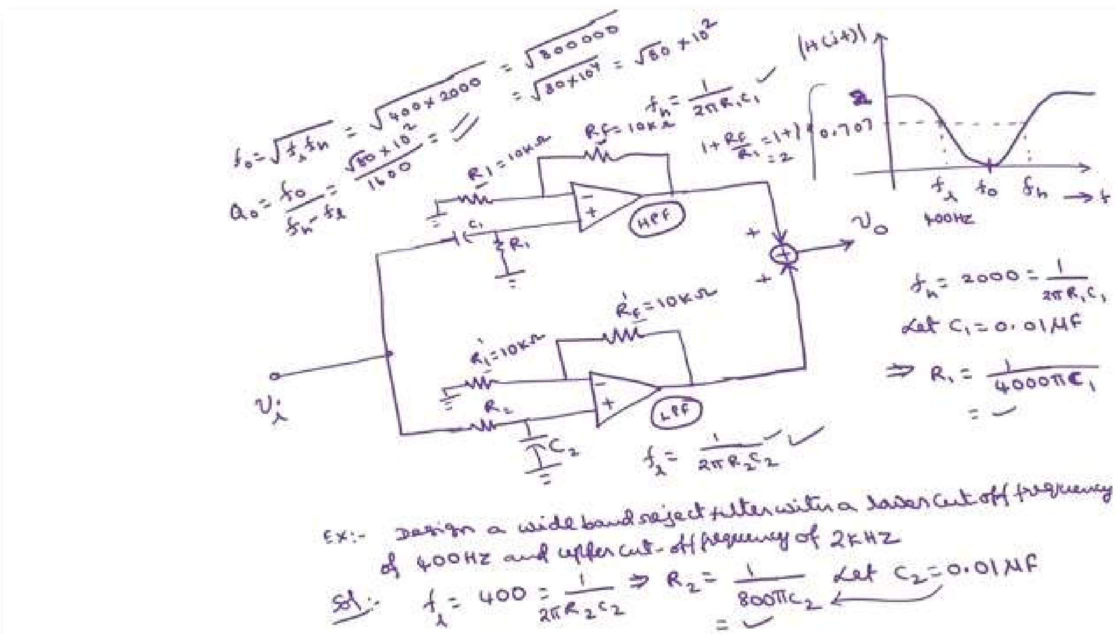

Now, if you consider the circuit diagram of wide band reject filter, this is high pass filter, this is R_F R_1, this is C R. And the low pass filter is you have to exchange the positions of capacitance and resistor. Now, these two will be having common input, which is the input v_i. So, this will give some response, this will give some response, finally you have to add these two responses.

This adder also you can implement by using op amp that we already discussed. Here I am showing this adder symbol, but this adder also can be implemented by using the operational amplifier. This is the final output v_o, this we call as R_1', R_F', this we are taking same values of R and C.

This is the circuit diagram of wide band reject filter and what are the frequency response let us assume that this is $C_1\ R_1$, this is $R_2\ C_2$. So, what is the f_h? the cutoff frequency of high pass filter which we have derived in the earlier lectures.

This is $\frac{1}{2\pi R_1 C_1}$ and what is f_l? the cutoff frequency of low pass filter, $\frac{1}{2\pi R_2 C_2}$. So, you have to choose the f_h value much greater than f_l value. Let us take an example, design a wide band reject filter with a lower cutoff frequency of $400\ Hz$ and upper cutoff frequency of $2\ kHz$. So, lower cutoff frequency means that is of low pass filter. So, f_l is given as 400 this is equal to 1 by $R_2\ C_2$ implies R_2 is equal to 1 by 400 $C_2\ 2\pi$ times this is $800\pi C_2$.

Let C_2 is equal to $0.01\ \mu F$ then if you substitute this we will get R value. This is about the R_2 and C_2 and to find out R_1 and C_1 f_h is given as 2000 this is equal to $\frac{1}{2\pi R_1 C_1}$. Here also let $R_1\ C_1$ is equal to 0.01 or $0.1\ \mu F$ it is up to you, now take some available value then implies R_1 is equal to $\frac{1}{4000\pi C_1}$. If you substitute this C_1 value we will get some R_1 value and R_1' and $R_1\ \ R_F$ and R_F' this you can choose a value of $10k$ which we have already discussed in the earlier lectures.

This is pass band gain of $1 + \frac{R_F}{R_1}$ which is 2. So, this will give pass band gain of 2 this is if it is $10k\ 10k$. So, then the response will be output versus input which is $H(jf)$ this is unity which is 0.707 this is f_l this is f_h this is $400Hz$ and this will be 2 pass band gain is 2 because $1 + \frac{R_F}{R_1}$ is equal to $1 + \frac{1}{1}$ which is equal to 2.

So, you can take any value of this pass band gain by choosing the proper values of $R_1\ R_F\ R_1'$ and R_F' and this is center frequency f_o is given by $\sqrt{f_l f_h}$ and what will be the Q factor? $\frac{f_o}{f_h - f_l}$. So, what is f_o? square root of one is 400 and other is 2000, $\sqrt{800000}$. So, this will be $\sqrt{80} \times 10^2$. $f_h - f_l$ will be $2000 - 400$ is 1600. So, if we compute we will get because this is wide band reject filter. So, you will get Q less than 10 you can compute this I am not computing this one.

So, you see about the design of wide band reject filter. So, by simply adding the outputs of the high pass filter and low pass filter we can obtain the response of a wide band reject filter. Then the last one is called as all pass filter. Before going to this I will discuss one example of the notch filter also.

Design a notch filter with a notch frequency of 50 Hertz means we want a notch filter which generates only 50 Hertz signal this is 50 Hertz signal it has to pass the remaining all frequencies. As you have seen in the previous lecture that you see the circuit diagram of this one. So, you have to find out the values of R, C this is $\frac{R}{2}$, $\frac{C}{2}$ you have to find out the values of R and C basically. So, to find out the R and C this is the value that we have derived $\omega_o = \frac{G}{C}$. So, this we call as f_o. ω_o we have derived as G by C.

So, this is equal to $\frac{1}{RC}$ or what is f_o? which is equal to $\frac{1}{2\pi RC}$. So, if f_o if we want 50 Hertz is equal to $\frac{1}{2\pi RC}$. So, now, let C is equal to 0.01 μF implies you can find out the R using $\frac{1}{100\pi C}$. So, you will get some value.

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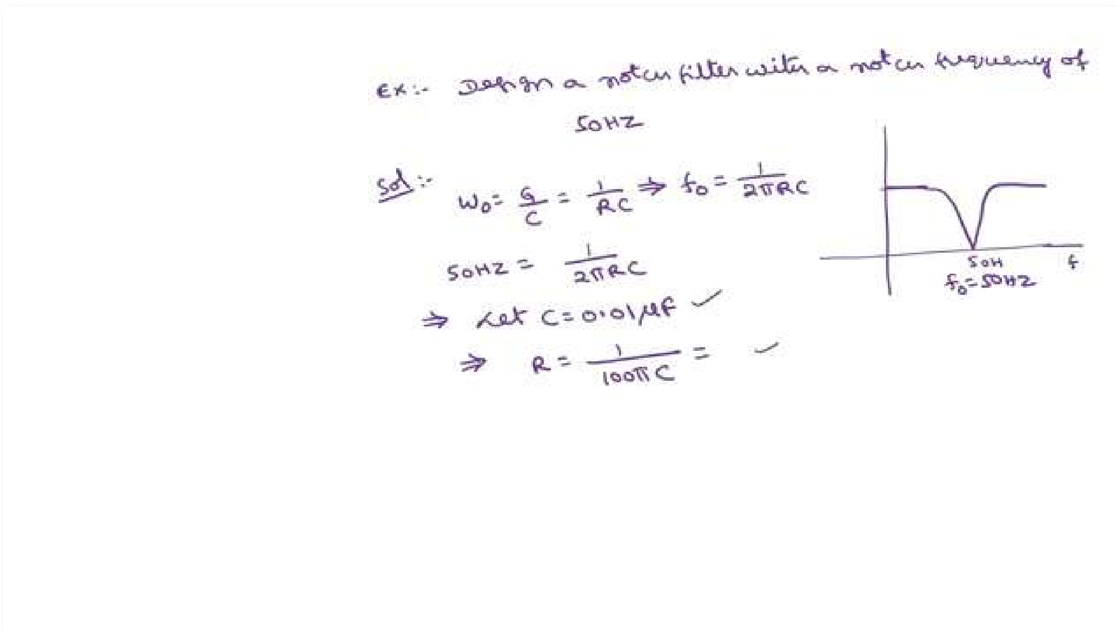

So, this is the C value this is the R value you have to set for this twin T network. You have to find set this C value as 0.01 and the R value that we are going to get using the relation that have given and there is no other component here also this is $\frac{R}{2}$. So, to get this $\frac{R}{2}$ we can connect a two such $R's$ in parallel R into R by R plus R two R which is equal to $\frac{R}{2}$ and to get 2 C we can connect two capacitors in parallel, okay. This is about this high pass filter, low pass filter, band pass filter, band reject filter.

So, these are most commonly used filters and there is one other type of the filter which is slightly different from the, these four filters which is called as all pass filter. As the name implies it passes all the frequencies. If I take the frequency response of this filter modulus of $H(j\omega)$ or jf as a function of f all the frequencies from DC to infinity this will be having unity gain. Then what is the use of this all pass filter? For this all pass filter if I apply say three input frequencies say f_1 is equal to 1 kilo Hertz, f_2 is equal 2 kilo Hertz, f_3 is equal to 3 kilo Hertz output will be having all the three frequencies.

Then what is the need of all pass filter? So, this all pass filter the frequency response will be constant throughout the frequency band, but if I take the phase response $\phi(f)$ this varies depends upon the type of the filter it may vary from 180 degrees to 0 or 0 to 180 degree depends upon the type of the circuit that we are going to use.

Means at different frequencies will be having different phase angles this is say f_1 this much is the phase angle, at f_2 this much is the phase angle, at f_3 this much is the phase angle this is ϕ_1, ϕ_2, ϕ_3. So, here this input if it is having phase of ϕ_1, ϕ_2, ϕ_3 then the output phases will be different this is $\phi_1', \phi_2', \phi_3'$. And if I take the input all with 0 phase then you will get output with some phase.

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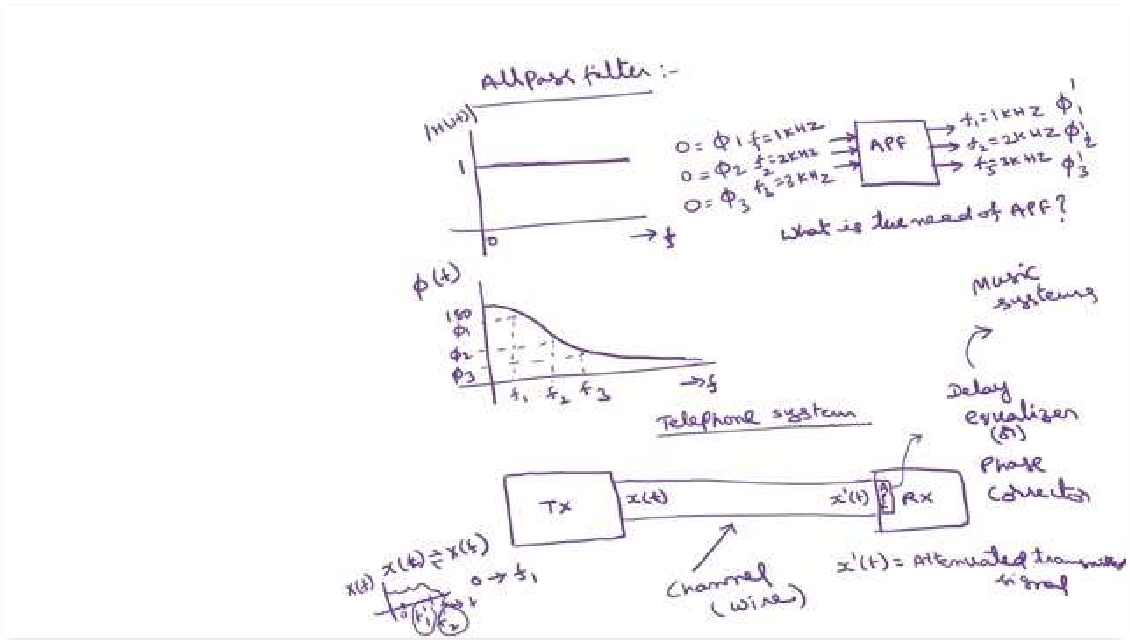

So, where this type of all pass filters are useful? So, if I take in telephone communications this is the transmitter, this is the receiver. So, this will be connected by wires this is wired communication this will acts as channel, channel is here wire.

In mobile communication and all we will be having wireless communication whereas, a telephone system is example of wired communication, okay. So, if I transmit this signal from here say $x(t)$, $x(t)$ consisting of different frequencies say $x(t)$ is having a range of frequencies say from if I take the Fourier transform this one $X(f)$ this will be having range of frequencies say from example 0 to some f_1. This is continuous $X(f)$ is continuous this is at frequency $X(f)$ because if it varies something like this this is say f_1 in between we have lot of frequencies. So, the receiver the received signal will be $x'(t)$ because the channel will attenuate the $x(t)$. So, $x'(t)$ is the attenuated of attenuated transmitted signal which will be received by the receiver.

Now here this $x'(t)$ will be having different attenuations for the different frequencies. So, all the frequencies will not be having the same attenuation okay. Suppose if I call this frequency as f_1', f_2' and so on. So, here this f_1' will be having some attenuation, f_1' will be having some attenuation and so on. So, at the receiver what you have to do is here we have to design all pass filter which provides the different phase angles or different delays for the different frequency signals.

As a result of that they will compensate the the channel effects okay. Channel will provide different attenuations to different signals okay. The signal which is passing through the more attenuation at the all pass filter will design with less phase delay. So, that will after this all pass filter all the frequencies will be having same delay. So, in order to equalize this we can use this all pass filter.

So, that is why the other name of the all pass filter is delay equalizer or phase corrector which is going to correct the phase. Instead of having different signals with different phase this will correct such that all the signals will be having same phase okay that is required in communication systems phase corrector. So, this will be sometimes used in the music systems also to have the sound effects. So, we can equalize there is a equalizer in the sound system also you can see. So, using that it will equalize the delays thereby will be having different sound effects.

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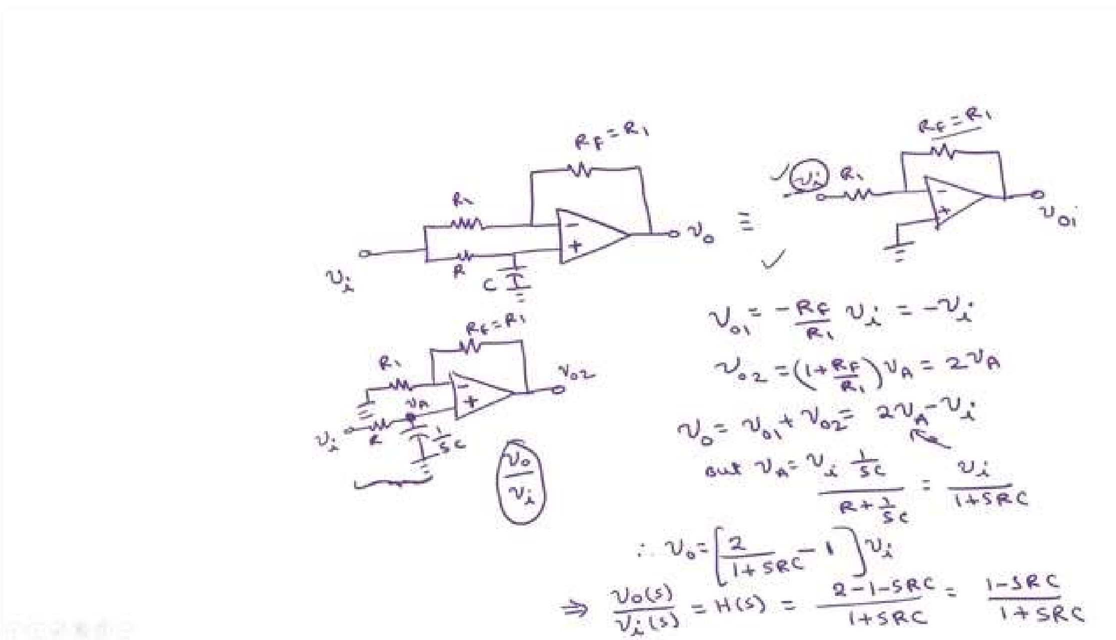

So, now, it will be circuit diagram of this all pass filter. There are two types of all pass filter one is phase lag and another is phase lead. So, this will be connected to the common input signal v_i and here the output v_o is taken. R_F normally we will take as R_1 this is some R and C. So, what is the transfer function of this circuit? So, we will derive the transfer function and we are going to show that this will acts as a all pass filter. This actually this input is applied to the two terminals inverting as well as non-inverting this is actually equivalent to two input circuit, but the inputs are same.

This is v_i this is also v_i. So, what is the output? Output will be sum of the voltages or the sum of the responses due to this v_i alone plus this v_i alone okay. So, if you want to find out the output due to this v_i this v_i this you have to short circuit. If I short circuit this, this is ground and this is ground point between these two ground points R and C will be eliminated. So, this will act as simply ground. Now, what is this circuit? This is nothing but inverting amplifier whose gain is $-\frac{R_F}{R_1}$.

If I set R_F is equal to R_1, -1. So, v_o if we call this one as v_{o1} due to this input. So, v_{o1} is given by $-\frac{R_F}{R_1}v_i$ which is equal to $-v_i$ because $R_F = R_1$. Now, due to the second input if I call as v_{o2} what will be the equivalent circuit? This will be grounded. Now, this will be grounded. This is v_{o2} here are they here this input will be there.

Let us assume that this value is say v_A. I will derive the relation between the v_i and v_A later. So, what is v_A? In terms of v_A what is v_{o2}? So, if I forget about this circuit if this is v_A this is non-inverting amplifier whose gain is $1+\frac{R_F}{R_1}$. This is $\left(1+\frac{R_F}{R_1}\right)v_A$. But R_F is equal to R_1 this is equal $2v_A$. Therefore, what is v_o according to superposition theorem? $v_{01}+v_{02}$ this is equal to $2v_A - v_i$.

But in order to get the overall transfer function $\frac{v_o}{v_i}$ you have to express this v_a as a function of v_i. So, that is clear from this circuit. What is v_a? voltage divider this is $\frac{1}{sC}$ this is R. So, but $v_a = \frac{v_i\frac{1}{sC}}{R+\frac{1}{sC}}$. sC sC will get cancelled this is equal to $\frac{v_i}{1+sRC}$. If I substitute this here therefore, v_o is equal to $\left[\frac{2}{1+sRC}-1\right]$ this overall into v_i implies what is $\frac{v_o}{v_i}$? which is the transfer function which of course, function of s is $H(s)$. The transfer function of the all pass filter is equal to the $1+sRC$ is LCM so $2-1-sRC$ this is equal to $\frac{1-sRC}{1+sRC}$.

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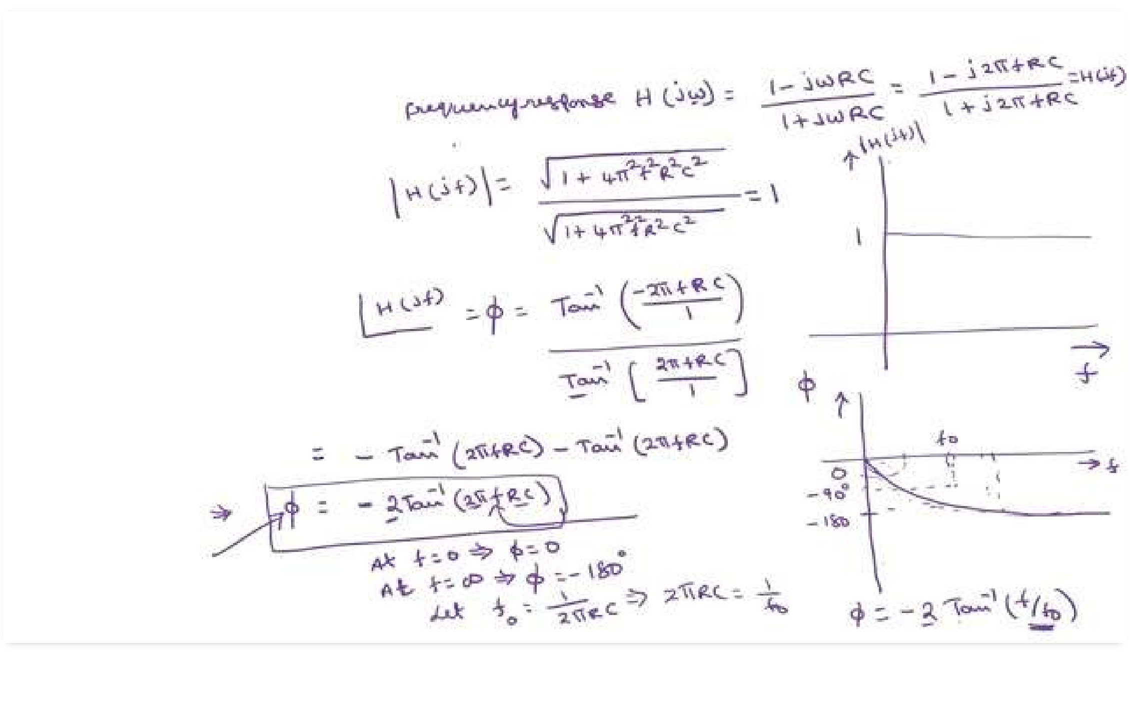

This is the transfer function of all pass filter and what is the frequency response? $H(j\omega)$ or jf also you can write is nothing, but this is you have to replace with $j\omega\ \frac{1-j\omega RC}{1+j}$. Now, if you want you can write in terms of f also it is up to you. If you want to write in terms of f this will be $\frac{1-j2\pi fRC}{1+j2\pi fRC}$. So, what is the magnitude of this you can call as $H(jf)$ because you are using f here you are using ω.

So, what is magnitude of $H(jf)$? So, the magnitude of numerator is square root of real part plus imaginary part $\sqrt{1+4\pi^2f^2R^2C^2}$. What about denominator? Denominator also same thing $\sqrt{1+4\pi^2f^2R^2C^2}$ means if I plot the magnitude response this is plot this is unity means it passes all the frequencies with same gain whereas, if I want to plot the phase response phase angle of $H(jf)$ if I call this one as ϕ what will be this ϕ?

So, the numerator phase angle is $tan^{-1}\frac{b}{a}$, $a+b$ phase angle of $a+b$ is $\angle a+jb$ which you have might have studied in your circuit theory or mathematics. This is $tan^{-1}\frac{b}{a}$. If I want the phase angle of $\angle\frac{A}{B}$ this is equal to $\angle A-\angle B$. So, the numerator phase angle is tan inverse of imaginary part is tan inverse of imaginary part is minus $2f\ \ 2\pi fRC$ divided by 1 divided by tan inverse of $2\pi fRC$ divided by 1. So, tan of minus theta is minus tan theta. So, this is minus if I take this numerator denominator tan to the numerator it will be minus. This is equal to $=$ $-tan^{-1}(2\pi fRC)$ again $=-tan^{-1}(2\pi fRC)$ this is equal to $-2tan^{-1}(2\pi fRC)$.

$$\angle H(jf)=\phi=\frac{tan^{-1}\left(\frac{-2\pi fRC}{1}\right)}{tan^{-1}\left(\frac{2\pi fRC}{1}\right)}=-tan^{-1}(2\pi fRC)-tan^{-1}(2\pi fRC)$$

So, this will be having different phase for different frequencies as f varies ϕ also varies. If you want to plot this ϕ as a function of frequency. At f is equal to 0, what happens $tan^{-1}(0)$ which is equal to ϕ is 0 okay. At f is equal to infinity, $tan^{-1}(\infty)$ is 90 degrees -2×90 is -180.

So, the response will be 0 this will be 0 phase and it will reach it at infinity -180 degrees. Of course, -180 will be in fact, this we have to write in the downwards this 0 you have take above you take this as 0 axis this as f at f is equal to 0 0 and it is -180. And, if I define this f_o as a some cutoff frequency here there is no cutoff frequency in the sense.

So, if I define let f_o is some frequency which is defined as $\frac{1}{2\pi R}$. So, $2\pi RC$ will be $\frac{1}{f_o}$. So, from this what will be the ϕ? ϕ is equal to, this $2\pi RC$ becomes $f_o-2tan^{-1}\left(\frac{f}{f_o}\right)$. At f is equal to f_o, $tan^{-1}(1)$ which is equal to 45 degrees into 2 means 90 degrees. So, at f is equal to f_o somewhere here we will get this is 90 degrees. At f is equal to f_o this is unity $tan^{-1}(1)$ is 45 into 2 is 90 so, -90. At f is equal to f_o, we will get 90 degrees.

This is this particular network is called as phase lag network because this is going to provide the negative phase delay. This circuit is called phase lag network. But, different frequencies will be having different this f_o will be having 90 degrees at this frequency this is the phase angle at this frequency this is the phase angle like that different frequencies will be having different phase angles.

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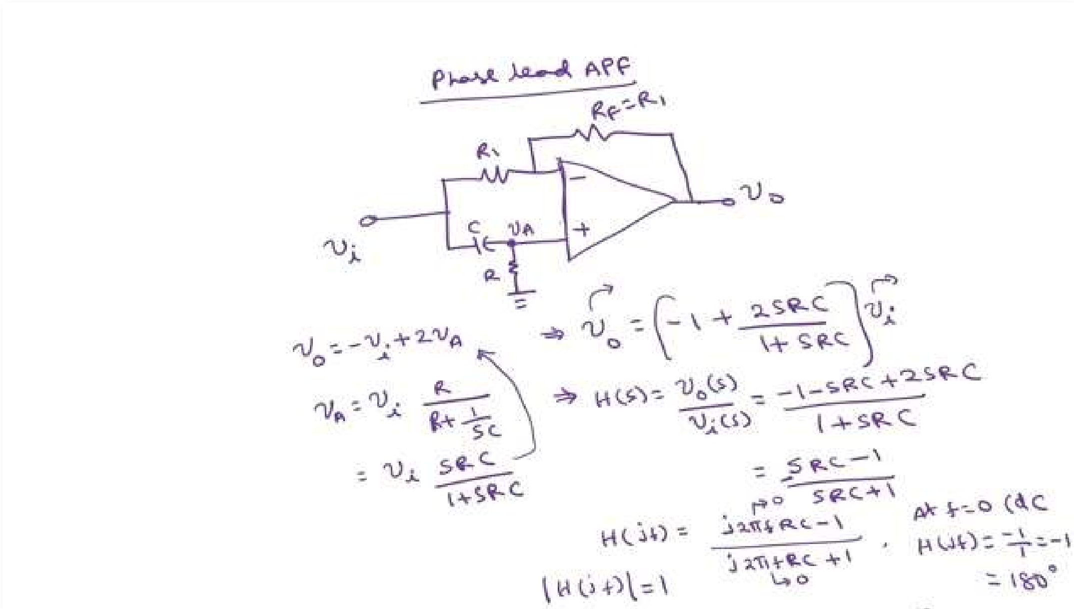

So, in some applications we require the positive phase delay instead of negative phase delay then we can design the phase lead network which is just by reversing this R and C you will get phase lead network. Phase lead all pass filter same circuit, but we have to change R and C. This you have to say C you have to take R this R_F also you have to take as R_1. If I call this as v_A, we will get similar to the previous procedure v_o is equal to $-v_i + 2v_A$, but here v_A will be different is equal to $v_i \frac{R}{R+\frac{1}{sC}}$. This is equal to $v_i \frac{sRC}{1+sRC}$.

So, if I substitute this here we get v_o is equal to $\left[-1 + \frac{2sRC}{1+sR}\right] v_i$ or what is transfer function $H(s)$ is equal to $\frac{v_o(s)}{v_i(s)}$. These are actually functions of s I am not writing here s for the sake of simplicity. $1 + sRC$ is LCM this becomes $-1 - sRC + 2sRC$ this is equal to $sRC \;\; 2sRC - sRC$ is $sRC - 1$, by $sRC + 1$ or what is H of $j\omega$ jf? $j\omega$, s is equal to $j\omega$ or $j2\pi f$ this is equal to $\frac{j2\pi fRC-1}{j2\pi fRC+1}$.

So, you can easily see that magnitude of this one is again unity it passes all the frequencies with equal magnitude, but what happens to the phase? So, tan inverse of this one imaginary component by real component is minus this minus minus becomes plus, but here one important point that you have to observe is at f is equal to 0 that is DC. So, this becomes $H(jf)$ becomes this is 0 this is 0, but this is $\frac{-1}{1}$ which is equal to -1 which is 180 degrees.

So, that is why you will get 180 degrees term extra this is equal to 180 minus so the numerator is $tan^{-1}(2\pi fRC)$ by minus 1 which is $-tan^{-1}(2\pi fRC)$ denominator is plus, but if I take into the numerator we will get minus. This is equal to $180 - 2tan^{-1}(2\pi fRC)$. So, if you plot this phase angle at f is equal to 0 at f is equal to 0 this is phase angle is ϕ is 180 degrees. At f is equal to infinity $tan^{-1}(\infty)$ is 90 degrees 2 into 90 180, 180 180 get cancelled ϕ becomes 0 and this will be something like this.

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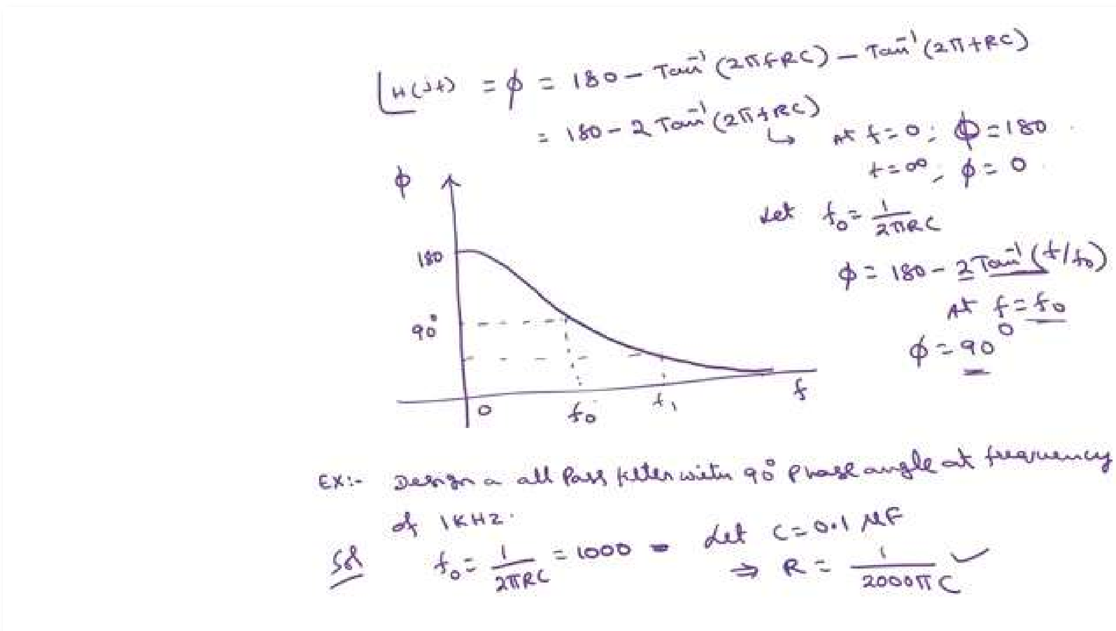

This is positive angle so this is called as phase lead network. And here also if I define let define f_o there is no cutoff frequency concept here, but I am just using f_o as a some frequency $\frac{1}{2\pi RC}$ then this ϕ becomes $180 - 2tan^{-1}(\frac{f}{f_o})$. So, at f is equal to f_o this ϕ becomes this is 45 twice 45, 90, $180 - 90$, 90 degrees. This is f_o. So, like that different frequencies will be having different phase angle. This is f_1, this is the phase angle like that. This is phase lead network. So, this is about this all pass filter which passes the all the frequencies, but it provides the different phase angles.

If I take one example. Yes, you have seen 90 degrees will be occurs at f is equal to f_o what is f_o? is equal to $\frac{1}{2\pi RC}$ this should be equal to 1000. So, let C is equal to 0.1 μF in case you can calculate $R = \frac{1}{2000\pi C}$ you can find out this R value. So, you can set this R and C values here and this $R_1 = R_F$ you can take some say 10 kilo Ohms. This is the design of this all pass filter which will give 90 degrees phase angle at 1 kilo Hertz frequency.

This is all about the design of filters. So, in the next lecture we will take another type of special application which is called oscillators. It will generate the sinusoidal signals using operational amplifiers which are called sinusoidal oscillators. Thank you.