**NPTEL**

**NPTEL ONLINE CERTIFICATION COURSE**

**Course**
**On**
**Analog Communication**

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**Lecture 22: VSB – SC (Contd.)**

Okay all right so we have four in case of VSB- SC we have already seen the input-output filtering characteristics and their relationships right, so let us just assume one particular filter.

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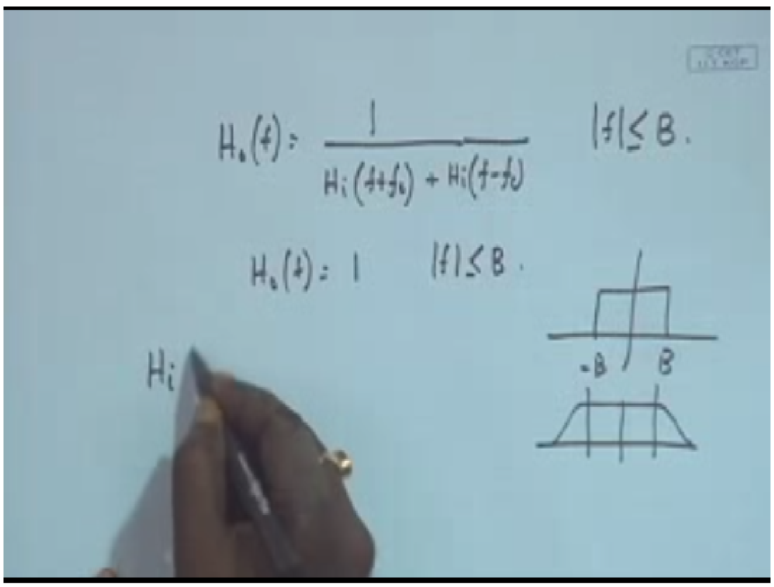


So we have already seen this relationship $H_0(f)$ is $1/ Hi(f+f_c) + Hi (f – f_c)$ right this is something we have seen okay but also remember this particular criteria is required only when I am within my band because what I was doing I was trying to detect my signal which is m(f) which is valid up to B so this filter characteristics must be valid up to B after that whatever happens I am not bothered as long as it is a low-pass and it is rejecting that higher frequency term so as long as this is happening I know that things will be all fine okay.

So because I have to only concentrate on that band rest of the portion are not that important okay because anyway nothing will be there okay so that is what we are trying to see now if we just employ this method that within this band this $H_0(f)$ is a very simple filter it is a simple low-pass filter okay ideal low-pass filter let us think about that so I can say that $H_0(f)$ must be one within this band for low pass filter it must be flat otherwise the signal will be distorted my m(f) will be distorted right.

So $H_0(f)$ must have unity gain within that band of interest outside that band it might be anything so I am not that much bothered only thing is that it must be having a low-pass characteristic so it might be something like this or it might be also something where the roll-off is there so within the band as long as it is unity I have no problem so that must be ensured rest other things are not that much important to me okay because it is all defined within that band only my actual signal was within that band.

So I all I have to do is within that band it should be alright okay all the other places what is happening it is not very important but one thing you need to keep in mind that it should not be in means separated in such a way that that 2fc term is coming back, so it must be having that low-pass characteristics and it is not taking that very high frequency table which we have rejected while doing this calculation right.

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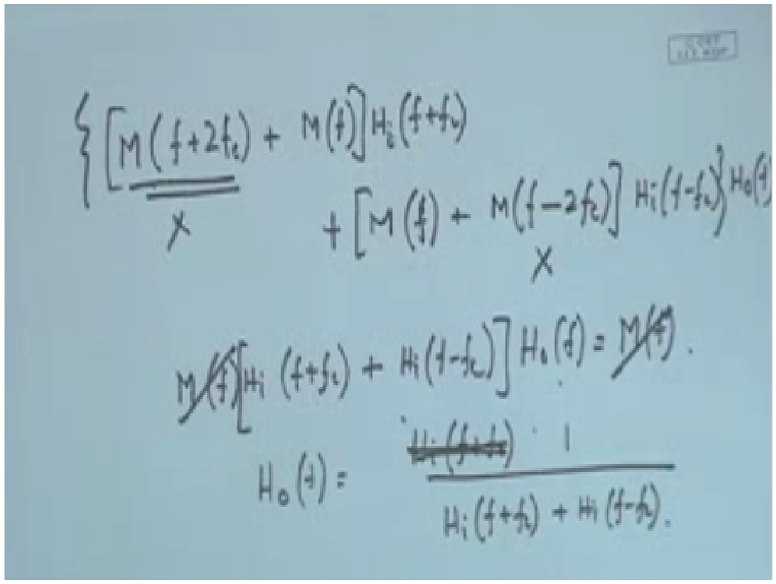


That relationship we have got by saying that these two terms we can actually cancel so it should not be taking these two terms, so whatever it is it must be means satisfying that criteria as long as it is doing that I have no problem okay.

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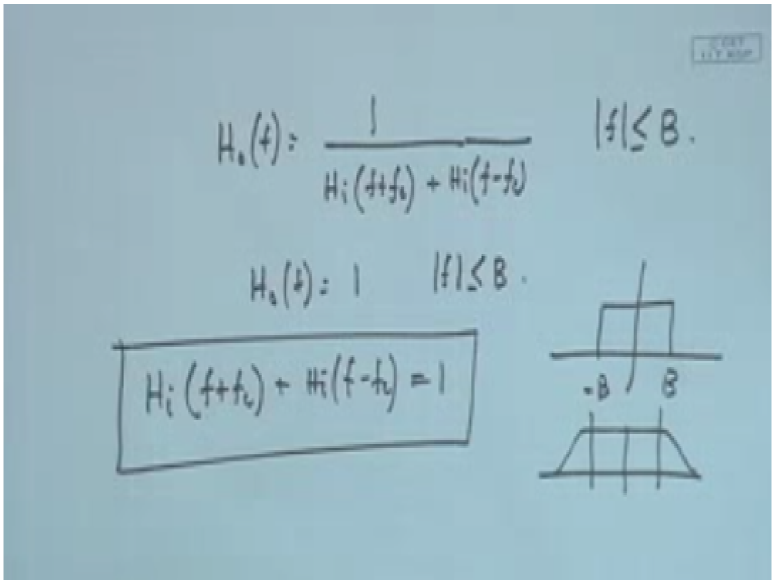


So if I assume this so within the band I can write that immediately Hi(f + $f_c$) + Hi (f – $f_c$) that must be 1 so this happens to be one of the desired criteria of designing my input filter okay output filter now is becoming very simple it is just a low-pass filter as long as it is flat within that band of interest it is alright so if I just take that then I can see that input filter has to be properly designed whatever it is it must satisfy this thing so let us try to see first what is this input filter this is how that should look like, okay.

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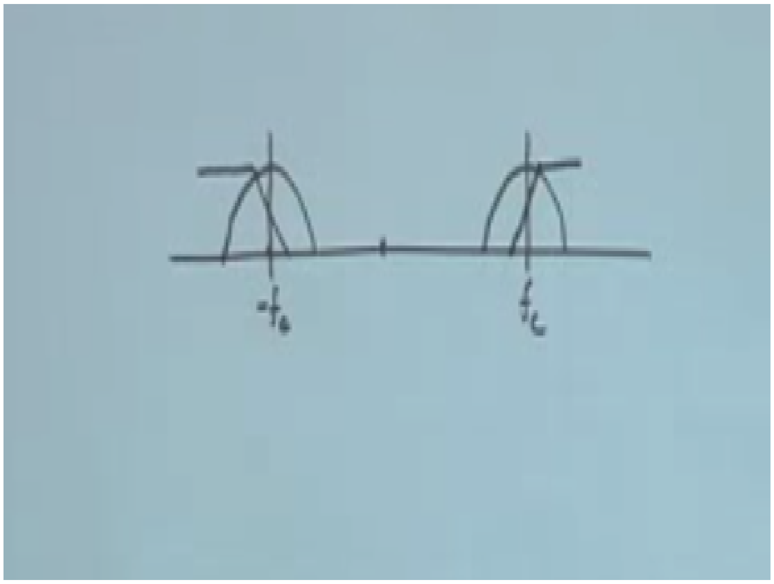


So what you are saying that this input filter is a band pass filter right so suppose I have a BSB – SC modulated signal okay now I want to put a input filter so what do I do I put an input filter which has a characteristic something like this okay and again it should be symmetric because it is a real circuit suppose I have a characteristic like this now here I can see the slope is not very sharp and it will of course distort the signals but what will happen let us say this is $f_c$ and this is -$f_c$.

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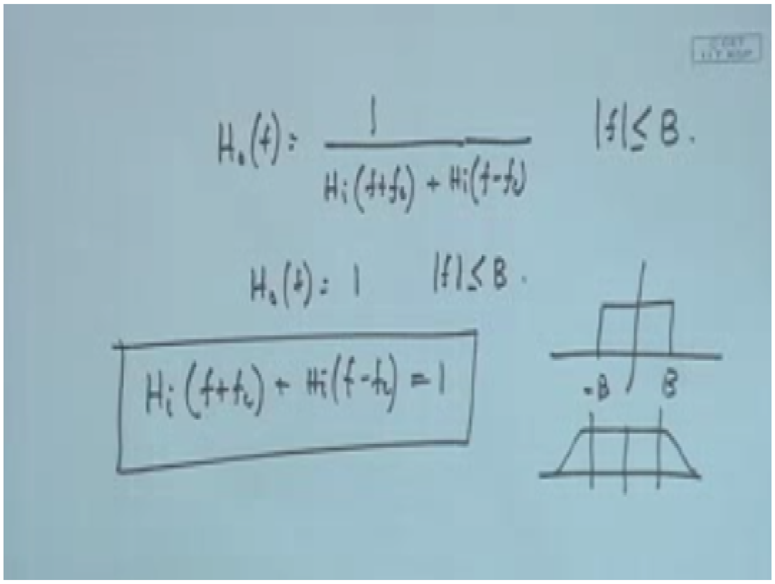


Now you shift you put this thing this criteria right.

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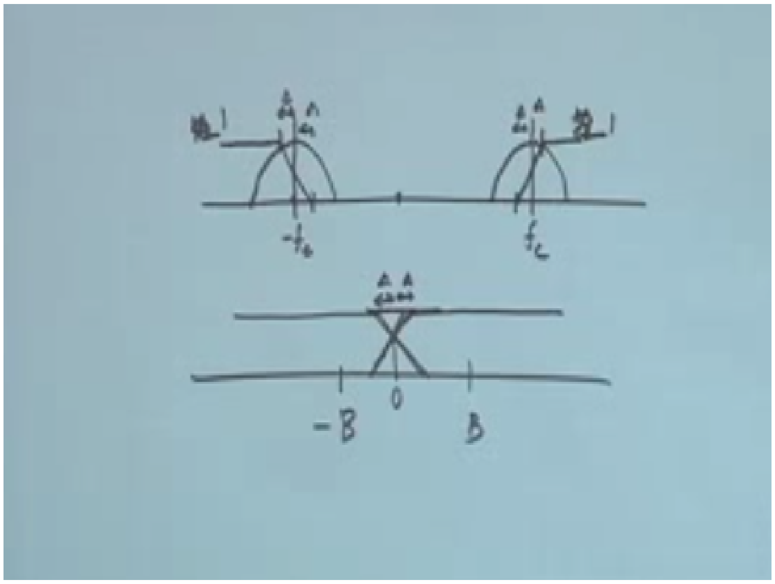


Let us say this is 1/2 and this is also 1/2 or I should not say 1/2 let us say this is 1 okay so what will happen now I have to shift this to so what all I have to do is this H whatever H I have I have to shift it to $+f_c$ and shift it to- $f_c$ and add these two and then I have to see whether it is giving me 1then only I can say that this is a valid filter for me okay the way I have adjusted it so this frequency and this frequency will be all similar right.

So if this is $f_c$ +some Δ this must be also Δ if this is also Δ this must be also Δ okay so this thing has to be all properly fixed okay so this is the case now shifted by $f_c$ so what will happen this will be completely shifted by $f_c$ and you shift on the other side by $f_c$ so they will actually sorry slope should be al same I will actually roll up like this that is the 0 this these are Δs now because it is a constant slope if you add this two they will all give 1.

So if you just add these two things will just give one so it becomes flat and you can now see from – B to +B it remains flat, so that means this criteria is being fulfilled if I design my filter this way right, so there now you can see that whatever relationship that we have got over here I can actually have a realistic filter okay where there might be some roll-off okay of course it needs to be linear or maybe some mere linear part I can just select and that will still suffice okay. So I will be suffering probably the filter will love some this thing so that will create a little bit of distortion but as long as the slope is almost tracking whatever I wish to do it will almost give me similar result so if I can do that I can immediately see that if I put that Hi(f) + fc and Hi(f) – fc I can always get back what whatever relation I have hence to get my actual SSB signal back from BSC signal actual signal back whatever criteria I put I can actually apply that so now I have seen that there is a realistic filter realization that can be employed over here and which can still give me due to the filter designing still give me back my signal back okay.

So once we have done that let us try to see whether VSD -SC signal also has similar representation as SSB signal that will be our next target okay.

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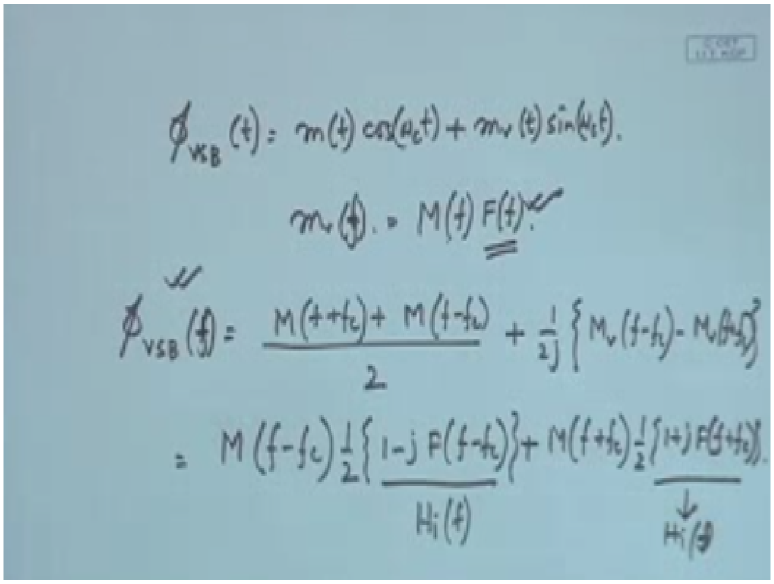


So for that what will try to see that will assume that maybe that is the case okay so we will say V SB( t) is nothing but some m(t) cos $\omega_c$t which is almost similar to a SSB signal and some mv(t) sin $\omega_c$t okay so we just try to prove that there is a representation like this okay so we are just it is almost like we are assuming this and then try to see whether we get something which are not contradictory okay.

So that is what we are targeting so we are assuming that okay there is a relationship representation which is like this where this mv(t) is nothing but my message signal pass through our filter okay or I can write mv(f) this is actually my M(f) followed by a filter called F(f) okay let us fill the transfer function so some filter I will be requiring through which if I pass this M(f) and of course we will later on see that this F(f) must have a low pass characteristics okay.

So it is a filter which I am aiming I still do not know I will be proving what this filter is so which gives me this mv so far we are just assuming okay that this mv(t) can be represented as this almost similar to our SSB in SSB what was happening the signal was passed through a Hilbert transform, so inverse transom has a transfer function like this filter and we were just passing it through it same thing we are doing.

Because we have said that we want our representation almost similar like SSP that is what we are doing okay so immediately what we can write this ϕVSB(f) we can now represent it in frequency domain right so if MV(f) is written as this I can now write this as M(f) + fc+ M(f) - fc of course because it is cos $\omega_c t$ so divided by 2 + because it is sin so it should be 1 / 2j and I will have this MV(f – fc) Mv (f+fc) right I will have this now Mv I can write as M(f) and F(f) I can write that I can take these two in the same part.

So I will get finally m(f) – fc if I just take f- fc there will be a 1/2 term and I will have 1 –j F(f-fc) if you just do a algebraic ordering and similarly if I just do M(f+ fc) I will get again another 1/2 1 +j F(f+fc) I will get this right, so this is what we are getting so what is happening now this M(f-fc) and M(f + fc) see this is my VSB(f) what was my earlier assumption it must be M(f –fc) +M(f +fc) x Hi(f) that will my our assumption okay or I should not say assumption that was the actual generation.

So this must be similar to that if that has to be the case then this and this both should be Hi(f) right so this should be Hi(f) and this should be also Hi(f) so let try to put that okay, that okay if this has to be equal then this should be both of them should be equal to Hi(f) if I put that do I get a contradictory result or do I get something which is what we have proven already.

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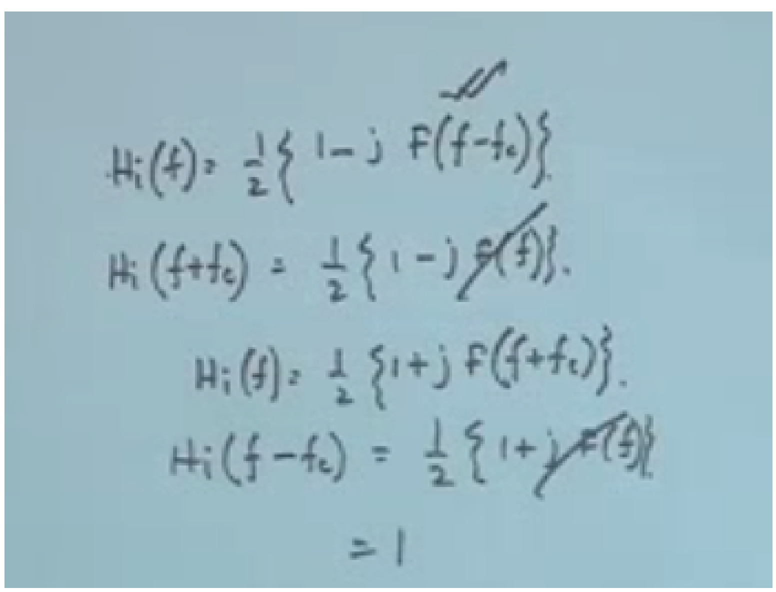

So if I just say Hi(f) is1/2 1 - j F(f-fc) right so from here I can calculate Hi(f) + fc right which is nothing but 1/2 1 – j that should be F(f) right and from the other relationship I can get Hi(f) should be1/2 1 +j if f+fc  from here I can get Hi(f – fc) okay so immediately that will be 1/2 1 +j F(f) now what was my relationship Hi(f) + fc and Hi(f)  - fc if I add these two that must be one add this two you can immediately see this will get cancelled and I will get 1/2 +1/2 that is 1.

So it is not giving any contradictory result okay so the filter that I have designed it is actually being consistent with that so therefore I can say my filter F is having this relationship okay, so it is just that filter if I just pass it pass my signal through this filter I will be getting the representation which is already being proven for SSB- SC I get almost similar representation for VSB- SC.

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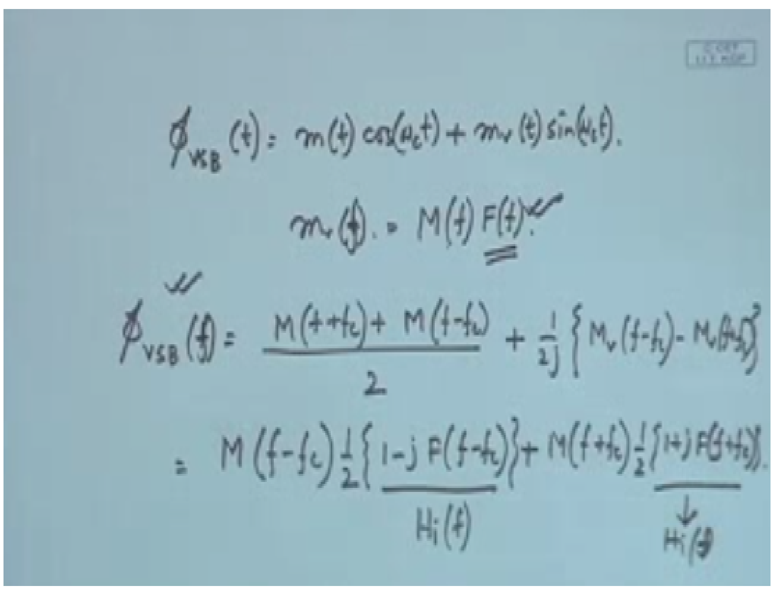


So that is why I can also see now things are all consistent I can also see if I just multiply this by a 2 cos $\omega_c t$ immediately I get my signal back because I am multiplied by 2 cos $\omega_c t$ put a low pass filter I will get my empty back that is consistent earlier also we have proposed the same thing so we can see that but this particular thing will be always getting back this thing right.

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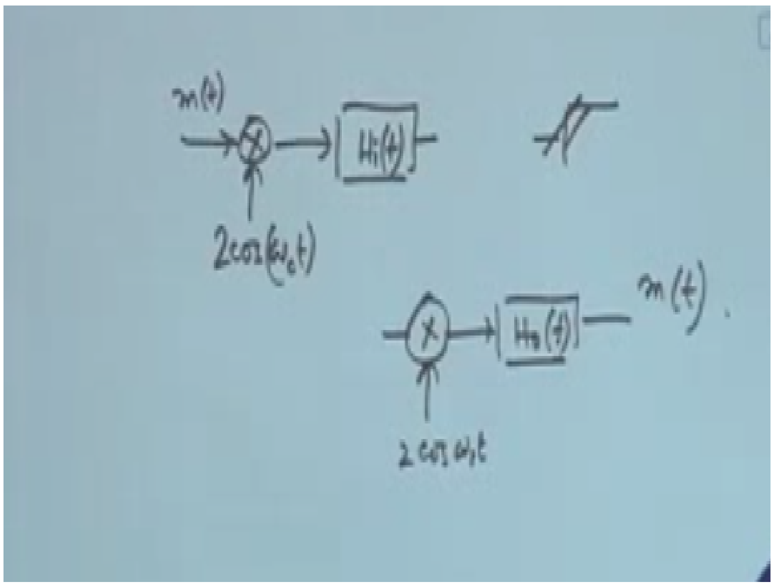


So basically I can now put my modulator as this so I will be having m(t) sorry means for the modulation what I can do that m(t) I can multiply by a cross first so let us say - cos $\omega_c t$ followed byHi(f) the filter I have designed so the demodulator will be take this sorry take this one multiply with because we have now seen that it is represented as this right it is represented as this so if I just multiplied by cos and followed by a low-pass filter I should be getting so multiplied by 2 cos $\omega_c t$ followed by whatever that Hof which is a low-pass filter.

I get my m(t) back okay so this is actually the VSB- SC modulation and demodulation it is pretty simple that only thing is that this filter has to be carefully designed the way we have demonstrated in the earlier case so that has to be very carefully designed so that at Hi(f) +fc and Hi(f) -fc gives me 1 in all the places only thing is that beyond that it is not that much required.

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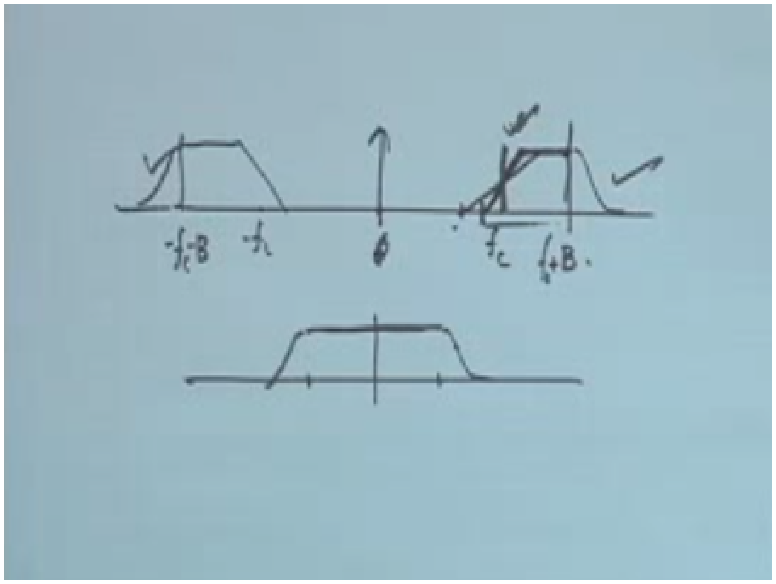


So whenever you are designing that filter you can always say that I need to have this after that it can have any roll-off so let us say this is my fc this is B up to be I mean fc+B and let us say this is my fc –fc - fc - B after that it might have any roll-off that is not a problem because if you just do F +so this is the spectrum if I just – Hi (f) + fc) and –fc what will happen these two will add up to 1 so up to B and - B it will be flat after that it will have this and this roll-off so no problem in that that can be anything only thing is that I have to ensure that whatever that filter that is flat up to be beyond that what I put it is not a problem okay.

So that is how generally as VSD signals are being generated so basically you have to all you have to do is you have to apply this filter and you have to characterize this roll off properly whatever rule of you wish to do now regarding the role of it depends on what kind of circuit you can choose and according there will be a trade-off of how much bandwidth you take suppose you want to have a little lesser roll-off okay.

Then the filter characterization accordingly will be simpler because you do not need to go for very high higher order filters, so what will happen it will take some extra band if you start going towards higher and higher order filter the roll-up will be sharper and he will be saving more and more bandwidth and you can see that at the end when you can make this roll off almost sudden then it goes back to SSP so BSB is nothing but from DSB slowly you can move up to SSP and it all depends on what kind of roll off you wish to give to your filter okay.

So depending on that your filter designing will be more challenging or less challenging but whenever you have more challenging filter design you will be saving on bandwidth whenever you have a less challenging filter design probably you will be will not be saving bandwidth that much okay so in TV spectrum probably they go up to the 50% of this band so they take over all banned if it is to be they take 75% of that so that is what they do they save 25% of spectrum okay.

And also they put carrier you will see that why that carrier has to be put that we will discuss later on whenever we talk about carrier recovery will be discussing more about that carrier recovery means why carrier has to be put in SSB signal okay.

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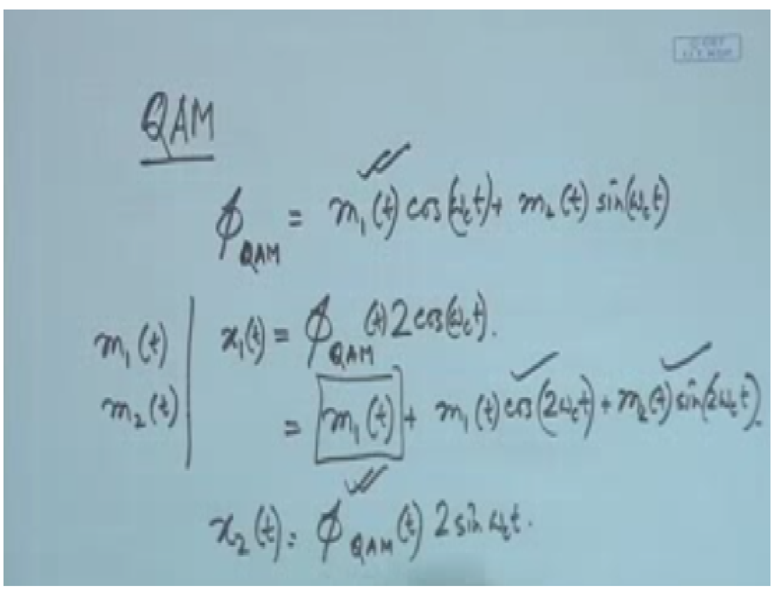


So after this now we have discussed about two band width efficient one is SSB one is DCB – SC sorry one is SSB- BC right now we will discuss another bandwidth efficient scheme which is called clam QAM quarter amplitude modulation, so what is this is something like that let us say I wish to generate QAM so what I do actually instead of modulating one signal now I take two message signals one is m1(t) and the other one is m2(t) it is almost like multiplexing I am taking this two message signal and within the band off 2B I wish to transmit both of them as long as both of them are actually demonstrating similar band so they are similar signals they might be having different spectra different time domain representation but the overall band that is being considered for them are similar as long as that is happening I can actually represent them as this composite signal.

SD it should be m1(t) cos $\omega_c t$ + m2(t) sin $\omega_c t$ so that is exactly what we are trying to achieve okay so through this motivation so QAM what we are targeting is we wish to basically transcript two signals simultaneously okay if you just try to see the spectrum that it will be occupying so if I multiply by cos $\omega_c t$ it will still have similar spectrum shift okay it will still be occupying that fc +B 2 fc – d and the negative house and if m2(t) is also having B bandwidth if I multiply by sin it will also be occupying same band okay.

So both of them will occupy the similar band and they will superimpose but just we have to see because the way we are producing them we are multiplexing them in the same band so they are getting superimposed is there a possibility that we can separate them out that is what we will have to appreciate can we separate them out after this so modulation was fine I know that they occupy if they are having similar baseband representation there will be occupying similar band but can I really demodulate them and separate them out.

That something which will be will be discussing now so what we have to do is suppose I want to modulate demodulate back m1(t) that is my target so what I do I take this QAM sorry QAM signal okay quarter amplitude modulated signal ϕ QAM put a band pass filter around fc of 2B So I will be getting the entire this composite signal okay I will now multiply this with two cos $\omega_c t$ okay.

So I will just multiply with cos Sinusoidal so what do I get so this is I am calling this as x1(t) okay so what do I get now this will be multiplied by cos so that should be $\cos^2$ so I will have 1 m1(t) +cos 2 $\omega_c t$ so m1(t) x cos 2 $\omega_c t$ and this will be 2 sin $\omega_c t$ cos $\omega_c t$ so that should be m2(t) sin 2 $\omega_c t$ right.

Now a very nice thing has happened both of these things has gone to a higher frequency only thing that is left which is at the base band is m1(t) I put up band pass filter I get this signal back so basically I have option of demodulating only m1t from this similarly if I just multiply this by sin you will see that I have an option of the modulating m2(t) back so if I produce x2(t) which is $\phi$ QAM of course that should be T into 2 sin $\omega_c t$.

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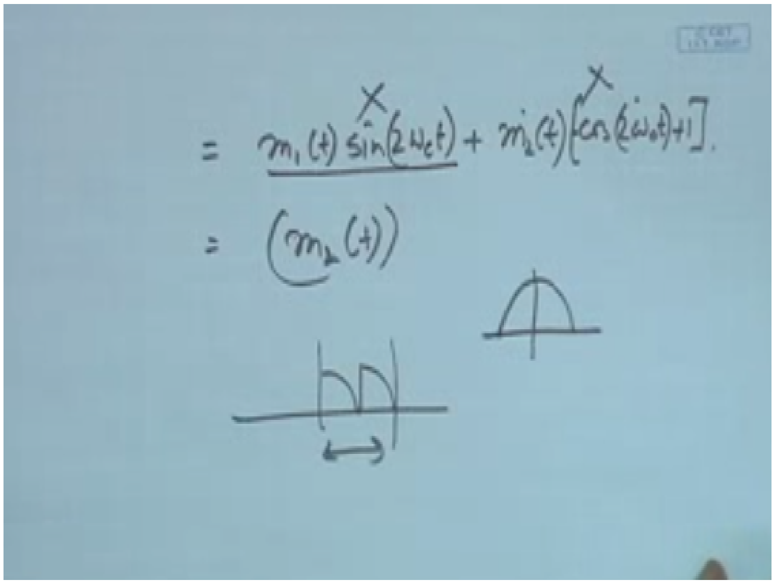


Now what will happen this is the first term is now becoming m1(t) sin $\omega_c t$ and then it should be m2(t) x 2 $\sin^2$ that actually cos 2 $\omega_c t$ - 1 or 1 – right so 1 -this right, so what I get again this term that is a modulated signal again going at 2 $\omega_c t$ so I can neglect this into this that can be neglected as well so I get back m2(t) after I do a band pass filtering, so basically I have now option of demodulating both the signals okay simultaneously but the good part is both of them are occupying B band.

So it is actually sorry 2B band so within 2V band I am able to transmit to such signals it is almost similar means bandwidth efficient as SSB signal because in SSB also if I had this 2B band I was putting 2 facility signal whereas here I am probably occupying a composite signal and it is having both of them right so m1(t) as well as m2t and I know now know that I can actually demodulate back both of them.

So equivalent I am able to transmit almost similar and similar bandwidth efficient modulated signal as SSB so what will be the corresponding modulator and demodulator.

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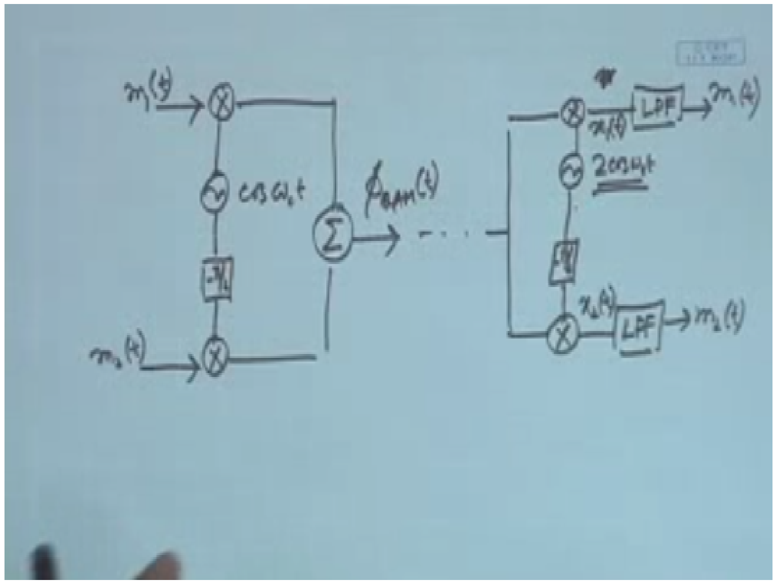


So I just have to generate a local cos $\omega_c t$ right multiply that with our message signal which is m1(t) and then from cost I have to produce sin so put a Π /2 phase shift multiplied with another signal m2(t) right so whatever you get you add them together and put it in the channel right so that is the ϕ QAM (t) once you receive that you put in two arm and then again locally you generate two cos $\omega_c t$ you directly multiply and then followed by a low-pass filter will be getting m1(t) back if you just give a Π /2 phase shift.

So this will become sin multiply and then followed by a low-pass filter will be getting $m_2(t)$ so that is actually the x1 sorry that is the $x_1(t)$ and this is the $x_2(t)$ just after or before the modulation right so this is what which will be happening if I do or the modulation right so modulation demodulation is pretty simple right now can we now say that this is probably the best modulation scheme where right now we cannot give that answer but bandwidth efficient wise probably yes it is equivalent to DSV.

And we can say that the transmitter part is not as hard as sorry bandwidth efficient voice it is equivalent to SSP but transmission part is not as hard as SSP because we do not have to employ any filtering like our SSP okay so either beat wave a circuit or any other way do not have to do that okay no Hilbert transform is informed means involved in this right so that is also a good part and we are able to modulate similar things but later on you will see that if this there is a carrier recovery process because I am doing this demodulation coherently right.

We are means there is non-coherent and coherent demodulation probably I have not talked about this terms but I have been keep on mentioning about it so non-coherent is like that envelope detector means I do not need a local carrier I do not have to generate local carrier which is in synchronous in phase and frequency to the incoming carrier I do not have to do that that is called non-coherent that means the receiver part is pretty simple.

And I am not having any influence on carrier generation okay whereas coherent is means I am subject to my receiver performance will be subject to how good I am in generating the local carrier at the receiver something is coming already with carriers because it is modulated so that same carrier I have to somehow extract so that the phase and frequency are completely in synchronous with the incoming carrier.

And then with that local oscillator which is generating synchronous phase and frequency carrier I have to demodulate it okay so whatever we have seen so far amplitude modulation was probably the non-coherent means there I can employ non-coherent demodulation DSB- SC definitely requires a coherent demodulation because I have to multiply by carrier SSB also if I do not put carrier it requires a coherent demodulation and for VSB also we have seen that it requires a coherent demodulation.

This one definitely I can see already that it requires a coherent demodulation okay so this coherent demodulation if I have there are all possibility that whatever carrier I will be generating that might have some phase drift or frequency drift compared to the incoming carrier and now what we will try to do we will try to look into that part if there are drift what happens to all this coherent demodulation part okay.

So if there are some drift in frequency or phase how I will be suffering what kind of suffering I will get for DSP what kind of suffering I will get for quarter amplitude modulation what kind of suffering I will get SSB and deals these similar things would be happening but specifically I will be targeting these two and there you will see probably this is not as good as DSP- SC both of them will have detrimental effect but probably this one will have more detrimental effect.

And will also be able to prove that whoever is bandwidth efficient like SSB or VSB will be also able to prove that carrier recovery is almost impossible over there okay similar thing over here also so carrier recovery will be impossible but the first thing second thing is even if we somehow get the carrier if you wish to demodulate if you have some phase or frequency offset you will be having a detrimental effect over it so in the next class probably we will be discussing that part in detail thank you.