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NPTEL ONLINE CERTIFICTAION COURSE

Course
On
Analog Communication

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Lecture 20: SSB-SC (Contd.)

Okay so we have already talked about single sideband modulation right in the last class so what we wish to do now we have already told that the single side band modulation we have already seen the frequency domain representation and through Hilbert transform we could finally get that the SSB signal if it is upper side band.

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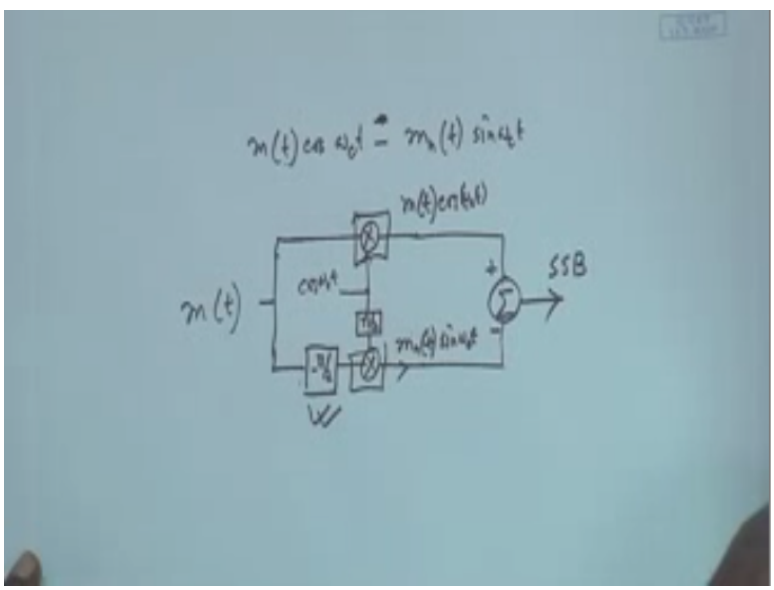

It looks like this Mt cos ω_CT -m HT sin ω_CT take that as a homework if we just do the lower side band this will just become plus nothing else okay so this can be proven so either upper sideband or lower side band it is just this separation of this plus sign or minus sign okay so let us say we are just taking this so that means the generation is pretty easy you take this empty okay take into two arms okay.

And then here it gives this Hilbert transform which is minus $\pi/2$ shift okay remember this minus $\pi/2$ shifter for each frequency component we have already have one understanding of that so you take that and then here you generate a cos $\omega_C T$ that if we put it over here multiply it so that will generate empty into cos $\omega_C T$ that you take in one arm here you give some $\pi/2$ phase shift okay so that will generate cos $\omega_C T$ sorry from cos $\omega_C T$ sin $\omega_C T$ and then you multiply so this will generate MHT into sin ωC so here you will get MHT sin $\omega_C T$ and here you will get MT cos $\omega_C T$.

So if you wish to do upper side band you just add with this Plus this minus that will be your SSB it okay so that is the easiest way we have understood so basically the modulator has been just defined by knowing that mathematical understanding so you wanted to generate that immediately understood that okay so it is nothing but this so I need to do this MHT so you devise a Hilbert transform circuit once you have that it is just nothing but two modulators one Hilbert transform right.

You can even tell that this is just the DSP SC modulator multiplication is nothing but the SJC modulated so basically this two DSP modulator and you only have 1 π /2 phase shifter which is the Hilbert transform and there is some things another adder is required that is all but we have already talked about that this particular thing that $\pi/2$ phase sheet for each frequency that is a very hard circuit to realize can we get some other circuitry which can do this.

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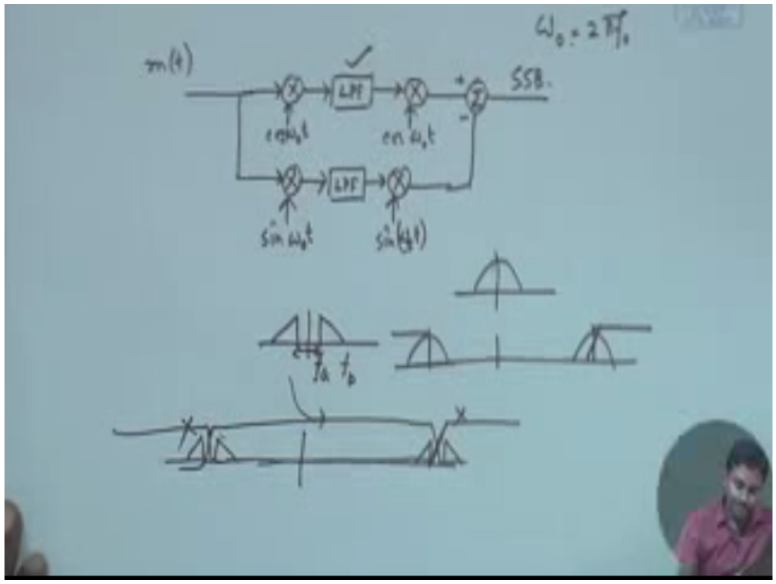

So I will just give the practical circuitry that was proposed by waiver that particular circuitry so the practical circuitry is looks like this so you have mt so this is one arm where you have a multiplier you multiply this by cos ω0T or 2 π F0T we will talk about this ω0 that is a or 2πF 0 what is that we will talk about that and this one in the other arm you multiply by sin ω0C okay after that you put a low pass filter I will also talk about the bandwidth of the loop means cut off frequency of that low pass filter once you have done that then you multiply with actual carrier so this is cos ω_CT and this is where you multiply with find ω_CT and you subtract them what I am saying I will be getting SSB signal.

As long as I turn this frequency properly and I put this ω0 properly see the technique we are applying that has a big historical significance and that is why probably SSB you cannot you can generate for voice but you cannot generate for video signal we will talk about that where you can generate SSB signal it is like this suppose I have a message signal which has no DC part or close to DC component.

I have a typical message signal like that so let us say it looks like this any voice looks like this voice has a minimum frequency so less than 300 hard there is nothing okay that will not significantly affect my voice signal okay so it has a minimum frequency let us call that as FA and it has a higher frequency FB beyond which I do not need that so for Y is typically three point four kilo Hertz right so 300 to 3.4 so FM is 300 FB is 3.14 kilo watt if the signal looks like this what I can do the basic functionality of this is something like this that I will take this signal to a small intermediate frequency okay.

And after taking it to small intermediate frequency probably I will put my filtering so it is something like this why I am doing this because I have to somehow employ a filtering technique to create SSB what I can do is something like this that I modulate it first let us say I get a modulation something like this and then I employ a low-pass filter which has a cutoff frequency exactly at this and I roll off like this then in reject this it will reject this or I employ a high-pass filter which other transfer frequency transfer function like this.

Then I will reject this part and I will get this very nice for suppose a signal even like this I can do that I do a modulation so it takes it over there and then I put our ideal high-pass filter so this is where the problem comes do you have something called ideal high-pass filter an ideal low-pass filter which has a sharp cut off you will never find that always any practical filter which are stabilizable will have a roll-off right and as you go to higher frequencies the amount of frequency it will take for roll-off will become higher and higher right so what is happening it will have our own of once it has a roll of fit is actually disturbing the SSB signal.

So I cannot allow that so basically I need to have a signal where I can still run this roll-off so I was talking about this free zone if I have that free zone probably I will be able to allow this roll off the float off of the filter I can allow this and I also know that at a very higher frequency I can do this that I immediately modulate to ωC whichever frequency I need let us say 900 mega or something like that and at that part I only have a separation of 300+, 300, 300 this side and 300 this side.

So 600 hard within 600 hard if I wish to put my roll-off that will be very high order filter and the corresponding filter transfer function will be almost non stabilizer they will not be able to and you will not be able to realize this whereas what I can do is something like this at a lower frequency I generate this filtering effect and then I translate it to a higher frequency I can do that this is what has been done over here so if you see now the circuit it is clear you are doing at a lower frequency modulation.

Then you are employing the low-pass filtering effect so that the roll-off is still realizable after that you are modulating with a carrier right that is all you are trying to do so that is the basic circuitry of wave a circuit okay but we will see he has done it in a little better way so we will try to employ that part so what he has done is something like this yes so this is the signal representation this is my FA this is my FB -FA and this is -FB.

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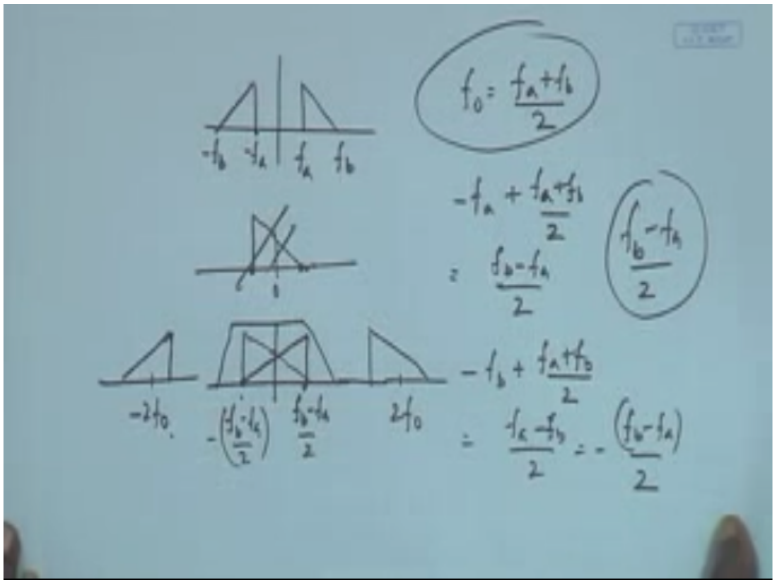

So the modulation he has employed is at a frequency this F0 is nothing but FA plus FB divided by 2 it will be very clear why he has chosen this frequency okay so immediately if you do that what will happen if I do modulate with this frequency so it will be translated by f0on this side and F0 on the other side right there will be a translation f0 on this side and f0 or the other side so if I just give pulse wave 0 translation let us say it shifts at the positive frequency side then what will happen his F minus a plus F0 will be the frequency where it gets translated.

So -FA+FB, FA+ FB/ 2which is nothing but FB -FA right so this frequency goes over here this FA- FB goes to -FB+ FA+FB/2 so that is actually FA- FB/2 or nothing but minus of FB- FA because they should FB is always greater than FA right so it just gets centered at zero and this will be your new to frequency the spectrum looks similar so only sorry I have done it wrong so this becomes FB- FA/2 this becomes FB –FA/2-1, 1 right and the separation immediately.

You can see if FP minus is which is the separation of this one so whenever I do give a translation this part shifts to the middle this part will go into F0 right so there will be one part centered around 2F0 and when I give negative shaped what will happen similar thing will be happening this will be centered around this and the other part will get shifted over here at centered around minus 2 at 0 right.

Now all I will be doing the filter he has designed is actually the filter cutoff frequency is this FB-FA /2 so that means he is trying to take this portion okay so that is the only meaningful information he will be required so he is just taking this portion after the filter once he takes that then he does rest of the things okay so this is what he is trying to apply so immediately we can see that F 0 is designed as this and the filter cut off frequency is FB-FA/2 that what has been employed so immediately.

You know for wise what should be our overall things right so that's something we already know okay fine and remember the big advantage is now it is no longer a even a band pass filter it is just a low-pass filter that I will have to put okay and that no pass filter is means it is the lowest frequency where you can roll off the low pass filter because any other low-pass filter if you just modulate it to a higher frequency the low-pass filter will still go to some higher frequency and then you have to put the roll-off you wish to put the role of because the role of region.

You have only 600 for voice right you want to keep it a slow as possible so that the roll-off can still be easily understandable or realizable so this is probably the best one you can do any other things you do your filter overall cutoff frequency will be little higher and the runoff of600 hard might not be sufficient okay so probably the waiver circuit is an intelligent design where is the best you can do you bring both the sideband overlapping in the middle but because he has those two arms you will see that he has manipulated it very well after doing the subtraction.

It will just generate with these two bands just generate the USB or LSB okay so we will see that part now so if I just go back to his circuit that was his circuit so basically what we do we take the MT in the upper arm we multiply it with cos ω0 T or 2 π F0 T right as zero value we already know now fA + SBπ/2 so MT is or equivalent MF can be written as M plus F plus M minus s that is the upper sideband and lower side better right.

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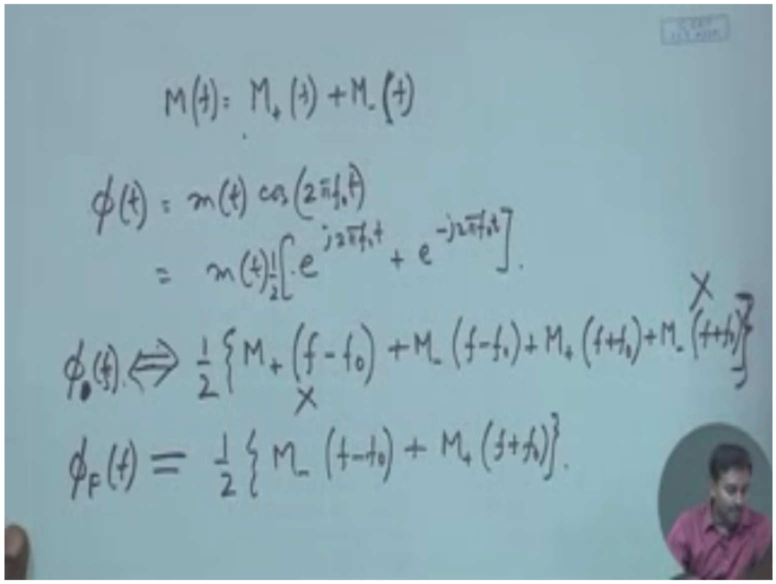

So what is happening I am if I represent this πT as my MT into cos 2π0T just after the first multiplication in the upper right so I can write this as M T into half e^{j} 2 π F 0 T plus e^{-j} 2 π F 0 T so I can write it this way now if I just do a Fourier transform which is π FT πF right if I just take a Fourier transform of this so what that should be half it is a Fourier transform of MP multiplied by this MP is already this one this one multiplied by this, this gives the frequency translation right.

It will give a frequency translation of F my it will take it to it is like F- F0 right and -J2 pi F0 will give produce F+F0 right so this will be if I just write M+ F minus F0 plus M-F- F0 because that is the whole MF right inverse multiplied by this that will produce these two terms and M F multiplied by this that will produce another two terms which will be at plus F0 so M plus 0 M minus 0 right.

So we get this π F alright now what we have employed after πF we have put a filter okay so in the filters which are the terms which will be vanished so this is the M plus one if you just go back to our filter representation so this is so this gets translated to this side that is actually multiplication by it is power J 2πF0T okay.

So there the negative 1 is getting survived see the negative 1/2 is I give a translation negative 1/2 is coming over here positive 1/2 is going over there this is the M plus part okay so I can write this M + F- F0 and this is actually M+F+F0 and this I can write as M+ F plus so since on the other side and this is actually M minus s plus s 0 right so these two terms are getting rejected after the finishing so what I will be left with if I represent that as ϕf F that should be half into so two terms which are M -F+F 0 that gets cancelled and this other term which is this, this gets cancelled okay.

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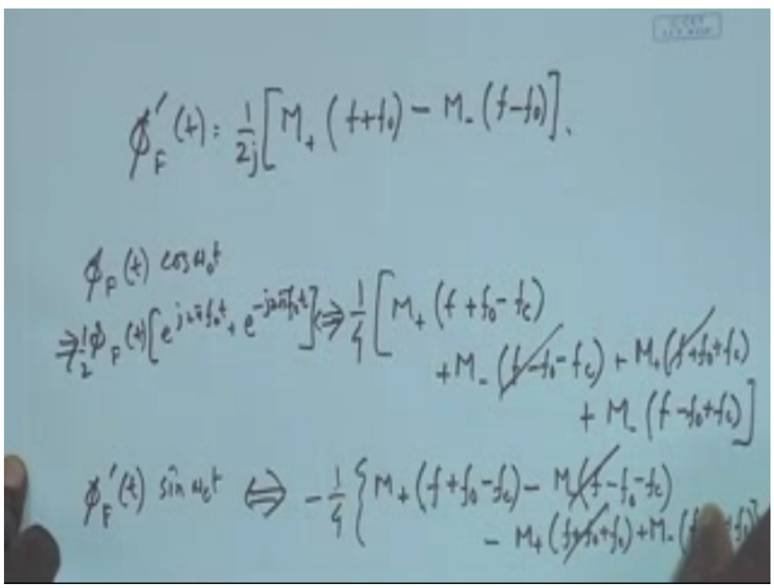

So we are left with M–F-F0 + M + F +F 0 so you have understood this part similarly we will have to do in the lower half also we have to multiply by sin means it should be whenever we do multiplication by sin it will be 1by 2 J and here it should be this plus minus this right and we will have to do the translation so if I do further with the time I will be getting if I just write that as ϕ - after filtering it should be just 1/ 2 J just do it yourself algebraic manipulation F+ M 0- M - f - f zero so we get this okay.

So now these two terms has to be multiplied with cos $\omega_C T$ + sin $\omega_C T$ so if I do cos $\omega_C T$ multiplication so immediately what do we get so suppose the upper arm on trying to do so that is corresponding time domain signals let us say ϕ FT which is having a frequency domain representation we have already done that which is ϕ FF so this into cos $\omega_C T$ right this is what we will have to do which is nothing but or I should say equal to ϕ F P into again.

We can write half it before J 2 ϕ FCT + e^{-J2} ϕ FCT okay so if I just employ same thing same technique of frequency translation will get again this half and there is already one half so that should be 1/ 4 and I will get a term of M +F+ f0-FC this is one term then I will get M- F- f0 -FC just another term so this is multiplication by FC and then -FC will give mean other two terms which is M+F+f0+ FC and then I get M –F- F0 +FC.

So this four terms I get okay after multiplication with cost of course the Fourier transform of that remember it is not equal we are doing the Fourier transform of that in the Fourier transform because it is multiplication bye to the power J 2 ϕ F 0 T it will be just translated at frequency FC and -FC both the terms okay we had two terms M s and a minus and 10 T will be happening when we multiply by minus J 2 ϕ F 0 T so we will translate it to plus F 0 so M plus that plus FC plus, plus X.

So it will have both the terms translated FC this is what we get for the first time in the second down we have that ϕ F^-T already that needs to be multiplied by sin $\omega_C T$ again do the same thing okay sign it can put as 1by 2 J e to the power plus J 2 pi FCT minus e to the power minus J 2 ϕ FCT so if you just try to write that and again do a Fourier transform so what you will get is minus 1 /4 and then the terms will be if you just do that same translation so it should be M plus you already had a component F plus F 0 that will be translated at -FC –M- F -F0 –FC.

And then another minus M plus F plus F 0 plus FC and then you have M minus F minus F 0plus FC right so this is what we get okay now all you have to do is subtraction this to this right because at the end after doing these two multiplication you are just subtracting from this to this now you see some of the terms are getting cancelled because this minus this minus, minus will become plus so you are actually adding these two okay so basically if you see this M minus F minus F 0 minus FC that is plus so this term will be canceled.

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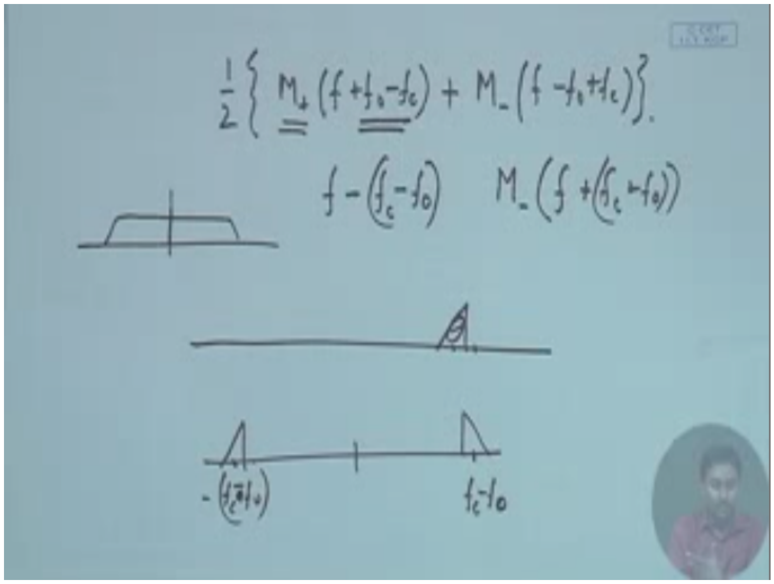

And then this term will be canceled so what you are left with is to this terms and to this term so 2 by 4 it will be which is becoming 1 by 2 and you get M plus F plus s 0 minus FC and you have another term which is M minus F minus F 0 plus FC very nicely this is created by SSB signals if you just see what it is it is the M plus part which is centered at frequency or which is translated at frequency if I just because FC is big soma or FC is the greater one so I can write it FC minusf0 right and this m- has been translated to f plus FC minus f0.

So the M minus part okay which is that lower part has been centered at C so M minus part n minus part is centered at this FC plusf0 right so that should be M minus FC plus f0 okay so this is where it comes this becomes the FC plus f0 the center part okay and this M plus part comes to this again sorry FC minus l0 u minus and this gets centered at the M plus part centered at FC minus f0 okay.

So you get up clear SSB representation but you could see that we have never employed any Hilbert transform over here it is just filtering and that one filter that we have employed where we can actually put roll-off provided there is that 600 hard free zone if there is no free zone I cannot put that filter because I will not be getting even in low pass filter at a very low frequency domain also I cannot get a sharp cut off right that is not possible there is no filter in this world which can have a sharp cutoff so due to that only voice signal.

Where you have that free frequency band at a large or around DC you can actually employ SSB whereas for video which has the DC term and all the frequency bands in the lower frequency zone are occupied you cannot really employ any SSP so SSB modulation is not possible for video signal whereas SSB modulation for voice signals that is possible okay whereas if you see the Y's band is pretty small it is just 300 hard to 3.42 so they are probably we will not be benefitting by reducing this whereas in video band it is almost like four point five megahertz.

If I just modulate it with DSPs C I will be getting almost 9mega but I know that SSB I will not be able to employ it over there because it occupies the voice band sorry video band looks like this so no way I can put up filters like this whatever I do I will never be able to employ this filtering it so there is a big difficulty in this so I will never be able to work out that in video so can we get any other modulation that might not give me as efficient as SSB but some efficiency where I can still have a filter roll-off.

That is where the modulation technique called vestigial sideband gets popular okay and we have just seen the end of probably analog video transmission so, so far it was analog video transmission even in many places I think it is still analog video transmission if it is analog video transmission it is employed with vestigial sideband okay so the vestigial sideband will see that there is a filter roll-off till we can employ and it might not be as efficient.

As the SSB that means it just occupies half the frequency but it will still be good enough okay so what we will do next is we will try to see what is this vestigial sideband modulation okay that we can employ for a video signal well after doing that we'll also try to see how do we do modulate see signal as well as the SSB signal because we still have to know this is all about modulation right we have not talked about demodulation so we will talk about demodulation of these signals later on okay thank you.