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NPTEL ONLINE CERTIFICATON 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 sideband modulation we have already seen the frequency domain representation and through Hilbert transform us could finally get that the SSB signal.

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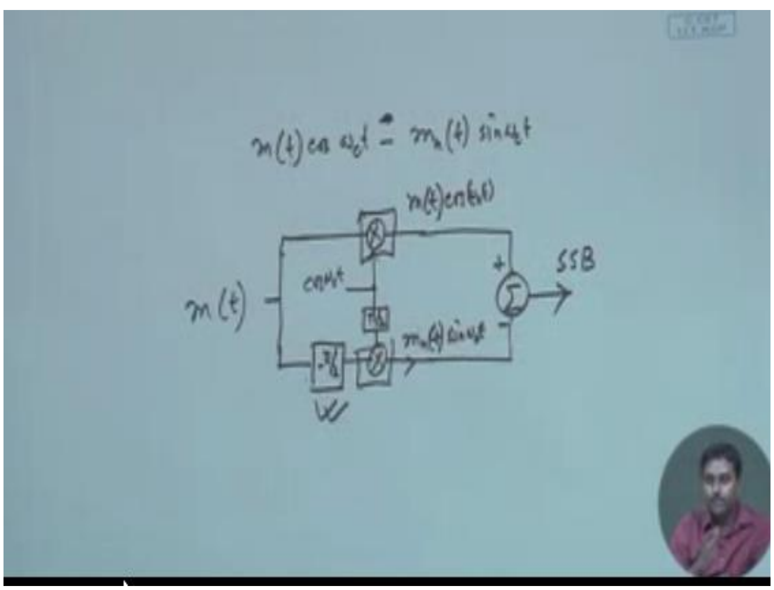

If it is upper sideband it looks like this M (T) cos ω_CT - m_HT sin ω_CT take that as a homework if we just do the lower sideband this will just become + nothing else okay so this can be proven so either upper sideband or lower sideband it is just this separation of this + sign or - sign okay so let 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- $\pi/2$ phase shift okay remember this - π by two phase 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 m(t) x 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 x sin $\omega_C T$ so here you will get m $_H$T sin $\omega_C T$ and here you will get m(T)cos $\omega_C T$.

So if you wish to do upper sideband you just add with this + this - that will be your DSPs sorry SSB signal 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 2 modulators 1 Hilbert transform right.

You can even tell that this is just the DSP - SC modulator multiplication is nothing but the DSB - SC modulated so basically this two RDS BSC modulator and you only have 1π/ 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 π / 2 phase sheet for each frequency that is a very hertz circuit to realize can we get some other circuitry which can do this.

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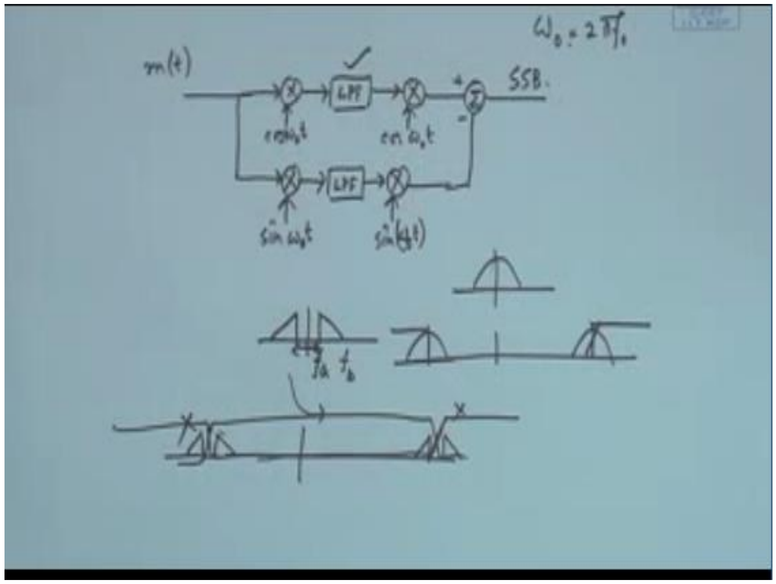

So I will just give the practical circuitry that was proposed by waiver that particular circuitry so the practical circuitry is looks like this so you have empty so this is one arm where you have a multiplier you multiply this by cos $t\omega_C T_0$ T or 2 π F_0 T we will talk about this $\omega_C T_0$ that is a or 2π F_0 what is that we will talk about that and this one in the other arm you multiply by sin $\omega_C T_0$ okay after that you put a low pass filter I will also talk about the bandwidth of the loop means cutoff 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 sin 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 ω_0T 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 hertz 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 3.4 kilo Hertz right so 300 to 3.4 so FM is 300 FB is 3.14 hertz 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 cutoff you will never find that always any practical filter which are stabilize able 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 off it 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 role 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 ω_C whichever frequency I need let us say900 mega hertz 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 heart within 600 hertz 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 higherfrequency I can do that this is what has been done over here.

So if you see now the circuit it is pretty 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.

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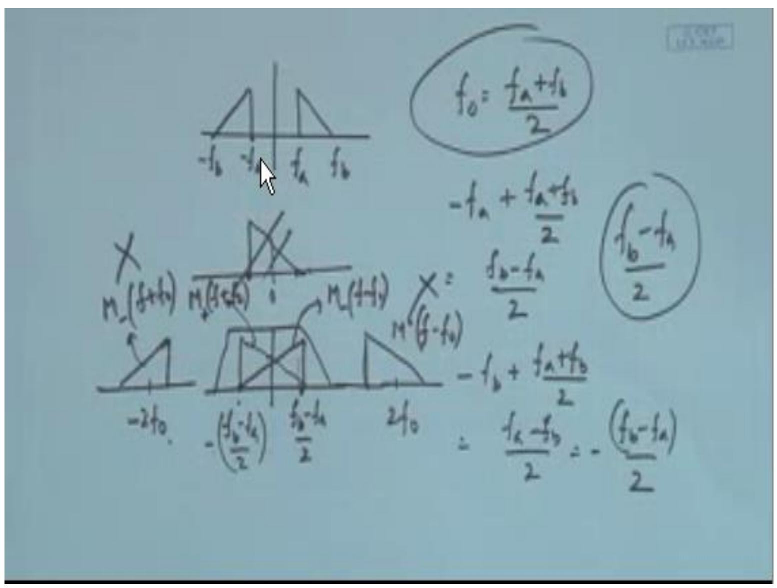

So what he has done is something like this yes so this is the signal representation this is my FA this is my FB this is -FA and this is -FB so the modulation he has employed is at a frequency this F0 is nothing but FA + FB/2 it will be very clear why he has chosen this frequency okay so immediately if you do that what will happen if I do modulate with this frequency so it will be translated by f0 on 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 palsy wave 0 translation let us say it shifts at the positive frequency side then what will happen this F - a + F0 will be the frequency where it gets translated so -FA + FB FA + FB / 2which is nothing but FB - FA/2right so this frequency goes over here this FA and -FB goes to -FB + FA+ FB /2 so that is actually Fa- FB / 2 or nothing but- (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 F b - FA / 2 right and the separation immediately you can see if Fb - Fa 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 2 F0right 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 --2 f0 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 ok 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 cutoff 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 is 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 low 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 hertz per voice right you want to keep it as low 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 hertz 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 ω_CT 0 T or 2 π F0 T right as zero value we already know now fa + FB pi/2 .

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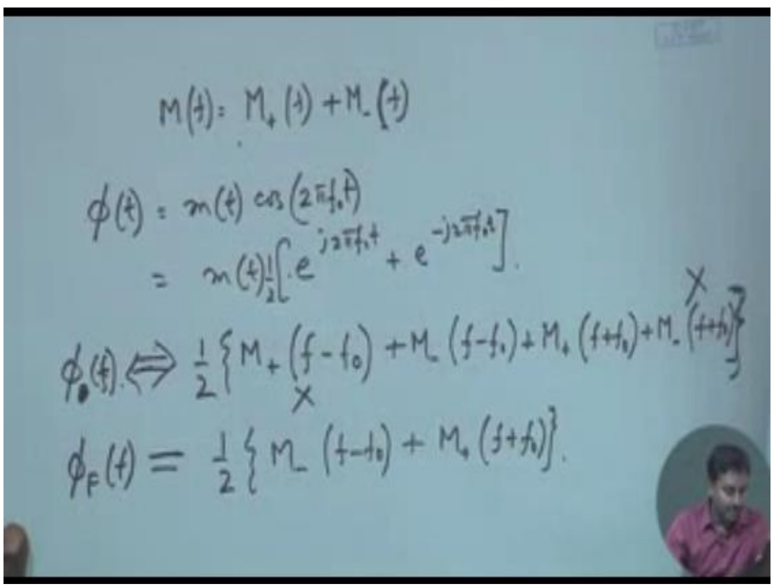

So MT is or equivalent MF can be written as M + (f) + M- (f) that is the upper sideband and lower side better right so what is happening I am if I represent this Φ(t) as my M(t) cos 2 π f0t just after the first multiplication in the upper right so I can write this as M(t)1/2e $^{j2 \pi F0T}$ + e$^{-J2\pi F0T}$ so I can write it this way now if I just do a Fourier transform which is Φ FT ΦF right if I just take a Fourier transform of this.

So what that should be half it is a Fourier transform of M(t) multiplied by this M(t) is already this one multiplied by this gives the frequency translation right in divorce class j will give a frequency translation of F my it will take it 2- F - F0 right and -J2 π F0 will give produce F + F0 right so this will be if I just write m + F-F0 + M-F - F0 because that is the whole M F right inverse multiplied by this that will produce these two terms and MF multiplied by this.

That will produce another two terms which will be at +F 0 so M + s + s 0 M - s +h 0 right this is fine so we get this Φ F alright now what we have employed after Φ F we have put a filter okay so in the filters which are the terms which will be vanished so this is the M + 1 if you just go back to our filter representation so this is so this gets translated to this side that is actually multiplication by $e^{J2\pi F0T}$ 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 + part okay.

So I can write this M + F - F0and this is actually M-F - F0and this I can write as M + F + s0 since on the other side and this is actually M - F + F0 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 gets cancelled okay 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 sin means it should be whenever we do multiplication by sin it will be 1/ 2 J and here it should be this +/- this right and we will have to do the translation so if I do further with the sign.

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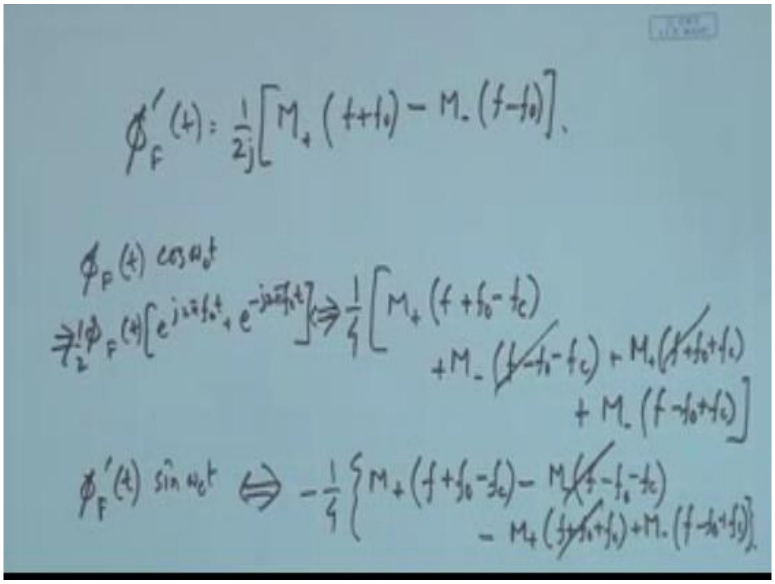

I will be getting if I just write that as Φ‘ after filtering it should be just 1 / 2 J just do it yourself algebraic manipulation F + M 0- M - f - f0 so we get this okay so now these two terms has to be multiplied with cos ω_CT + sin ω_CT CT so if I do cos ω_CT multiplication so immediately what do we get so suppose the upper arm on trying to do so that is corresponding time domain signals lets say Φ F T which is having a frequency domain representation we have already done that which is Φ FF.

So this into cos ω_CT CT right this is what we will have to do which is nothing but or I should say equal to ΦF P into again we can write half it before J2πFCT + e $^{-J2\pi\ FCT}$ 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- F 0 + 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 J2πF0T it will be just translated at frequency – FC and - FC both the terms okay.

We had two terms M + and A- and F0Twill be happening when we multiply by- J 2 π F0T so we will translate it to + F 0 so M + that + FC+ + 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 ω_CT again do the same thing okay sign it can put as 1/ 2 J e $^{+ J 2 \pi FCT}$- e$^{- J 2 \pi FCT}$ so if you just try to write that and again do a Fourier transform.

So what you will get is - 1 / 4 and then the terms will be if you just do that same translation so it should be M + you already had a component F + F 0 that will be translated at - FC - M- F - F 0 - FC and then another - M + F + F 0 + FC and then you have M - F - F 0+ 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 will become plus so you are actually adding these two okay so basically if you see this M- F - F 0 -FC that is plus so this term will be canceled and then this term will be canceled so what you are left with is to this terms and to this term so 2 / 4 it will be.

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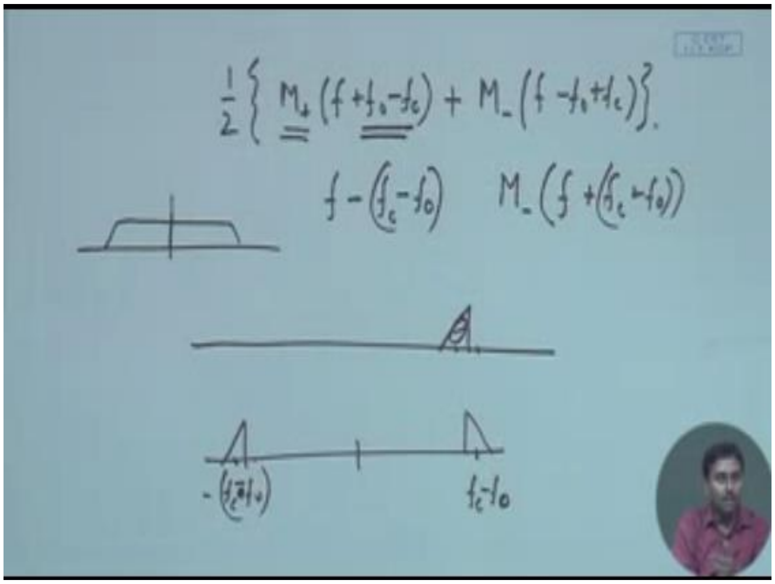

Which is becoming 1 / 2 and you get M + F+ s 0 minus FC and you have another term which is M minus F minus F 0 + FC very nicely this is created by D s be sorry SSB signals if you just see what it is it is the M + part which is centered at frequency or which is translated at frequency if I just because FC is big soma or FC is the greater one so I can write it FC-f0 right and this m minus has been translated to f + FC- f0.

So the M minus part okay which is that lower part has been centered at C so M minus part sorry n minus part is centered at this FC +f0 right so that should be M minus FC+ f0 okay so this is where it comes this becomes the FC + f0 the center part okay and this M + part comes to this again sorry FC –F0- Italy and this gets centered at the M+ part centered at FC - 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 minus off provided there is that 600hertz 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 cutoff 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 SSB band is pretty small it is just 300 hearts to 3.42 the hertz so they are probably we would not be benefitting by reducing this whereas in video band it is almost like 4.5 megahertz.

If I just modulate it with DSB SC will be getting almost 9 mega hertz in but I know that SSB I would 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 minus 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 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 minus 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 will also try to see how do we do modulate or DSB SC signal as well as the SSB SC 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.

Acknowledgement

Ministry of Human Resources & Development

Prof. Satyaki Roy

Co – ordinator, NPTEL IIT Kanpur

NPTEL Team

Sanjay Pal

Ashish Singh

Badal Pradhan

Tapobrata Das

Ram Chandra

Dilip Tripathi

Manoj Shrivastava

Padam Shukla

Sanjay Mishra

Shubham Rawat

Shikha Gupta

K.K Mishra

Aradhana Singh

Sweta

Ashutosh Gairola

Dilip Katiyar

Sharwan

Hari Ram

Bhadra Rao

Puneet Kumar Bajpai

Lalty Dutta

Ajay Kanaujia

Shivendra Kumar Tiwari

an IIT Kanpur Production