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Course
On
Analog Communications

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Lecture 46: Noise Analysis – SSB- SC

So we have started doing the analysis of this SSB-SC right the first task is as we are seeing through the last two lectures that the first task is evaluate the power of the modulated signal that is the most important part. In SSB we have already seen if I just put the previous light. So what we have seen that modulated signal is empty cos ɷct up to that it was all fine but it also has either + or - mht sine ɷc t or 2 π fct so evaluation of power becomes more critical over here because for mt we have a definition the power I know but $m_h(t)$ we do not know.

So that is why we did some derivation step to actually realize that the power of that or the power spectral density of $m_h(t)$ is same as the power spectral density of m(t). Therefore the overall power because they are also similarly band-limited, so the overall power will remain P. So now for the modulated signal if this is P the corresponding multiplication with cos that must be P / 2 and if this is also P the corresponding multiplication with cause or sign whatever you multiply the overall power will be again 1/2 so it should be P / 2 so P / 2 + P / 2 it is the overall P power that is being transmitted. So this is something we could know after doing all these things.

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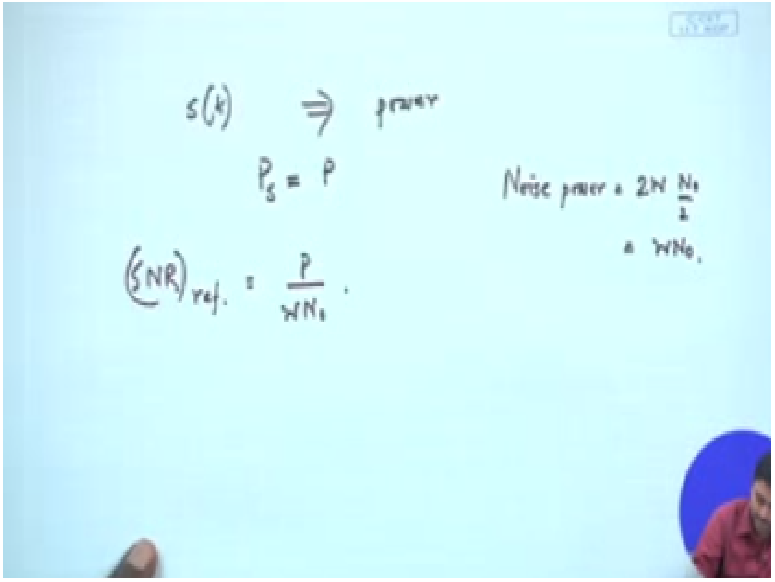

So this s(t) corresponding power or Ps that is actually P / P is the way it is defined P corresponding power spectral density if you integrate from- W to + W that is something you could okay. So now we know the power of it at the modulated, so therefore the baseband also must transmit that same power P fortunately here we are getting exactly and power okay. So if I just transmit empty whatever power it will be that is the same power we are getting okay.

Now this is the power will be transmitting what is the noise again same noise so the noise power that must be it is a low-pass filter from - W to + W with strength n_0 / 2, so 2 W into n_0 / 2 which is nothing but W and 0 right so therefore SNR reference must be P / WN_0 so far it is quite easy. Now let us try to draw the receiver chain because now we will have to go through the receiver and then try to see at the output what happens what the corresponding noise and what is the corresponding signal power okay. So receiver chain if I just draw it let us do it in a new page.

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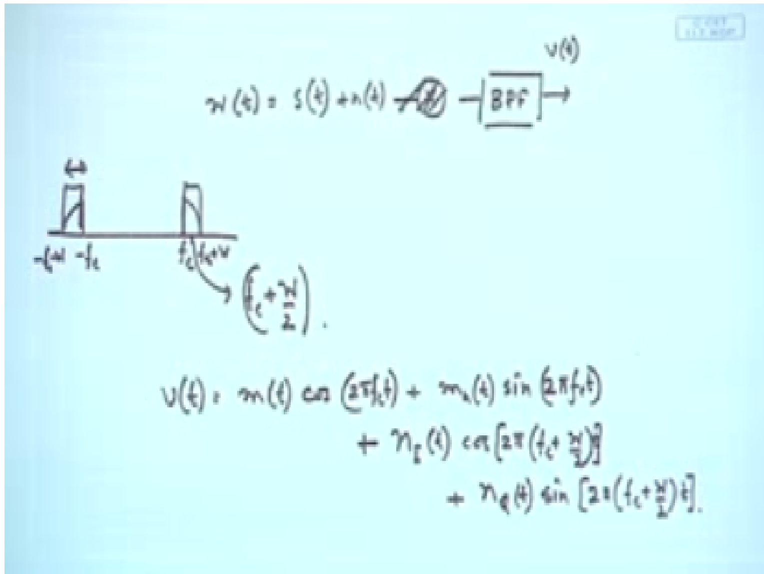

So that is let us say w(t) which is that signal s(t) that we have defined which is a modulated signal + noise this will be incident on the receiver, so that goes to the receiver okay. So in the very beginning of the receiver you will have a multiplier SSB also more demodulated in a similar way as DSP is demodulated so you will be putting a multiplier but before that we have to employ a band pass filter, so this should not be the first part there should be a band pass filter. Now the important thing this particular band pass filter the band is no longer 2w because in SSB we have already restricted it 2 W.

Because we have taken either the upper band or the lower band so the corresponding modulation whichever way it looks it will be either from fc 2 fc + W and- fc 2 - fc - W or the other way okay one of the band will be technique, so therefore if my interest region is only this I will be also employing a band pass filter which is only of this width. So instead of putting a band pass filter of width 2w I will be now putting a band pass filter of width W because I want to cancel as much noise I can.

If I can do that why should I waste my this thing and why should I take some more noise so because it is a more spectral efficient modulation scheme, so I know that I can employ a smaller bandwidth filter okay and there is also another thing what is the center frequency of this particular filter that is no longer fc that is actually fc + W / 2, so this is a big change which happens whenever you employ SSB so the field that filters center frequency is now changed, what will be the consequence of this?

The noise will be characterized in a different way because that is the frequency now of the band pass that is the central frequency of the band pass noise, so therefore if I put it as in-phase and component the central frequency will not be ω c it will be fc + W / 2 or 2 π into fc + W / 2 okay so that is the change which will be happening in SSB okay. So all right I pass through bandpass filter so what do I get I will be getting s(t) will remain as it is because s(t) is exactly passing through this filter which is designed for him only.

So therefore if suppose I call this to be V t so my V t should be that should look like mt so st will remain the same cos 2 π fct + m_Ht sine 2π fct, so these two things are there okay. Now the noise part okay now is a band Limited noise so therefore it must have a in-phase component but the corresponding cause should be at this frequency so 2 πfc + W / 2 + N_Qt sine 2 π sorry the t was missing right.

So this must be my VT slightly modified because of the filter characteristics so means whenever you are doing my analysis it is very important that you understand this entire process okay and you characterize the noise properly it that is the most important part because otherwise the calculation will be wrong. So after that what do we do after passing through the band pass filter will be multiplying /cos 2 π fct.

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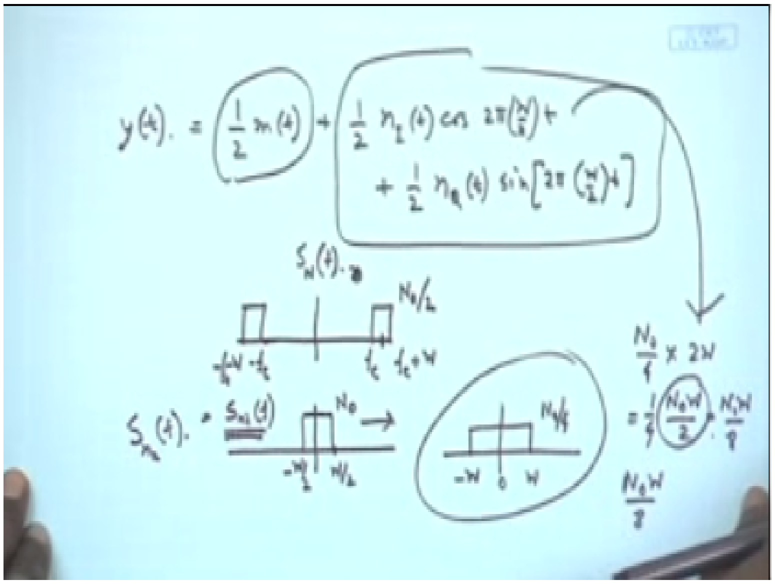

So this Vt will be now multiplied / cos 2 π fct which is nothing but this mt will be multiplied/ this cos square 2 π fct + m_h t sine 2 π fct x cos 2 π fct + n i t cos2π FC + w / 2 into t into cos 2 π fct + n cube t that is sine 2 π fc + W / 2t into cos 2 π right this is what will be happening immediately I can simplify this because this is cos square I can get half and then I will get two terms one is mt and the other one will be mt cos 4 π fct. Similarly I can also take ½ this will be m_ht sign for π fct.

So these are all higher frequency term which will be cancelled / the next low-pass filter now these two it cos a cos P if I just take half a cos a + b+ cos a - bso I can write ½ n_it cos of a + b so that must be 2π into 2fc fc + W / 2 t + 1/2 n II will have cos a - b so that must be2 π W / 2 t right I will have this is again a higher frequency term this will be canceled. Similarly for sine also I will have because it is sign a cos b so it should be sine a + b + sine a - B so it should be ½ and cutie sign a +B means same thing so it should be 2 π2 fc + W / 2 2 t + 1/2 and cut this will be sine a – B.

So that should be signed to Π W / 2 T this will again get cancelled / the low-pass filter so after the low-pass filter will be left with half mt + I will have this half n_it cos 2 π W / 2 x T + 1/2 n_Q t sine 2 π w / 250 right this is what we get at the output, so that is my yt now clearly you can see that has a message term and noise down. So this is the first time because of this filter central frequency shifted we get both the in-phase and quadrate term in my output noise okay and now we will have to probably get the overall perspective of this noise.

Now though all those things will be useful that they have no cross correlation because if they are there are cross collision there will be cross term which will be generated but that thing will be we already know that not probably there it is all the events symmetric spectrum and they do not have any cross correlation term okay and they are orthogonal to each other. So basically the overall power spectral density will be just power spectral density of this + power spectral density of this one.

Okay so we will have to first evaluate the power spectral density of this one, so let us try to see what is the power spectral density first of all what is the ni, so this was actually let us try to evaluate this so this was at fc this is my n or s and if I am trying to draw okay so this was as fc or - fc and this is at - fc - W this is at + fc this is at + fc + W right. So if I now try to evaluate the ni that should be my central frequency this will be shifted over here and this will be shifted over here the strength was n 0/ 2 right so in I or n cube will be going from - W / 2 because it is the central frequency is basically fc + W/ 2.

And that much shift we give so therefore it will be going from - W/ 2 to + W / 2, so - W / 2 to+ W / 2 unlike the previous one and it goes up to n_0 right there gets added. So this is my S_{NI}F or S and Q F both of them okay. Now let us just consider this one now n_It is getting a modulated town so I will have to first get a modulation term on this how much modulation W / 2. So there will be a shift of W / 2 in this side as well as this side and because it is a multiplication with cos, so it must be 1 / 4, so what do I get?

So this n_0 will become n_0 / 4 and will be shifted to W / 2 so this will be shifted to W / 2 so from then it will occupy 0 to W with a strength of n_0 / 4 and shifted on the left so that will occupy this part it will come up to W and the strength will be again n 0 /4, so that should be the spectrum this is fine and i cos 2 π w / 2t should have at his spectrum and why I should consider this whole spectrum because my filter that low-pass filter is having a filter cut off at W. So therefore this entire spectrum as noise will actually enter through the low-pass filter.

So that should be the case what is the corresponding power? That is actually n_0/ 4 x 2 w so n_0 W / 2 fine you have a half term over here, so I will get 1 / 4 so noise due to this noise power due to this will be this much right same thing will happen over here either you multiply / cos or sin in power spectral you will have no effect similar thing will be produced over here because the frequencies is also that shifting frequency or translation frequency is same. So you will get again this is actually n $_0$ W / 8 right, so the other one also will be n $_0$ W / 8overall it should be if I just add these 2 and 0 W / 4.

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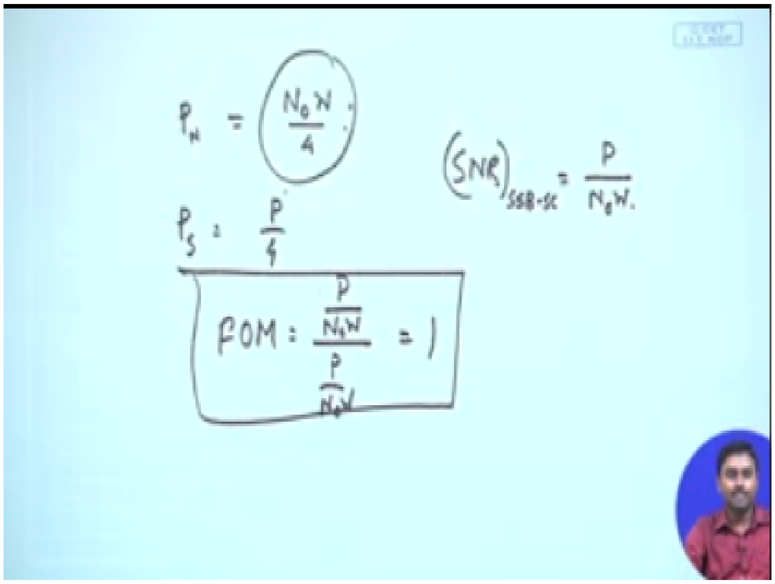

So I do get overall noise power which is n_0 W / 4 this is fine, so this is my noise power what is the signal power empty therefore it should be P and 1/2 factor is there in power spectral density it will be 1/2 ^{2}. So it should be 1 / 4 P so the power is P / 4 so the signal-to-noise ratio for SSB must be P right now the good parts what was the reference signal to noise power exactly the same. So therefore my figure of Merit must be this P / n_0 w divided / p + 0 W again we get a figure of merit of 1.

So even though SSB is a band width efficient modulation scheme we will be expecting that probably it will be taking less amount of noise, so it must be more efficient than DSP SC but after the calculation we could see that is not the case it is exactly equivalent efficient as the baseband and as the DSP SC there is no difference in these two and both of them are equally efficient compared to our amplitude modulation in terms of noise analysis we are just talking about noise analysis, but of course SSB has other advantage because it uses the frequency in a better way.

Because it just with the same frequency it can transmit two signals potentially two similar kind of signal because it just takes the half frequency of DSP SC right. But we could see noise analysis wise there is no difference this is a very fundamental result that comes generally it is a counterintuitive result because we expect that because it is more spectrums efficient. So probably it will if I put a employ a proper filtering probably it will take less amount of noise so it must be more efficient people often also do another mistake while calculating the power of SSB signal.

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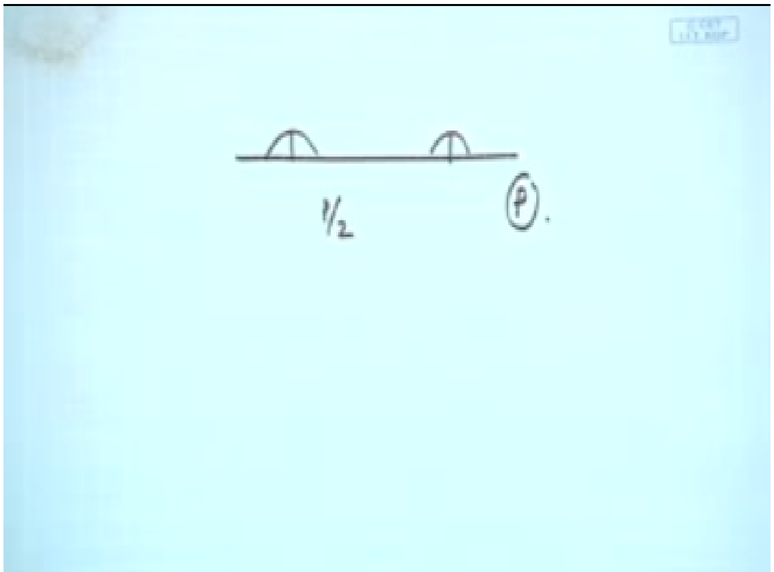

So what people do you think about a filtering method okay so this is a mistake people often do of course it is it should not be done. So if you just think about filtering method what happens your DSB will look like this and then as if half the power will be transmitting okay, what we have seen that SSB exactly transmit P power whereas if you just calculate it this way you will think that this is already DSB so it must be P / 2 and you are transmitting half of that power so it should be P /4 that is not the case.

Because in a way you have to think that whenever you employ a filtering you are actually rejecting a band that means you are wasting the power, so this P / 2basically getting counteracted / the filtering filter is rejecting that that means you have to actually generate more power to produce a certain power in the modulated signal okay suppose you are targeting P out to be transmitted through the modulation process. Then you have to actually generate 2 P power over there then only after filtering you will get P power so that's a common mistake I have seen over the years people.

Just to while calculating the power of DSB SC sorry SSP they just keep doing this twice huffing of power that they say okay it is DSB SC so what all power gets 1/2because I am multiplying /

cos I again employ a filter which takes out half of the power so it becomes P / 4 that is a very common mistake which people often do so we should not do that. And if you do it from the other side which we have done you will see that actually a P power which is to be launched so this probably ends our discussion of noise analysis.

Because the other thing that can be done we have also discussed about two other modulation scheme one is quadrate amplitude modulation QAM and the other one is VSP VSB it is a little bit more trickier because DSB will not have ideal filtering and therefore the noise calculation will not be like that very simple integration okay. So it will have a particular role of filtering and that filtering has to be considered whenever you are doing that and accordingly the overall calculation has to be done because the input filter and output filter for DSB will be and that filtering has to be taken into account to actually evaluate.

The overall noise power that will be linked to your modulation, so that is something you can take as homework so analysis quadrate amplitude is just almost equivalent to DSB because it is just two simultaneously you are putting then they will have both the noise and then you just multiply / cos and sine you have to just see that. So it will be almost similar to DA SB SC analysis there will be no difference in to that okay so it will it will almost be same. So after doing this probably we have we told that we will be comparing all these modulation schemes in terms of various aspect.

So now just the one part which was left when we did that comparison chart that was the noise analysis, now we could see the noise in terms of noise how they perform so probably DSB C and SSP are the best in terms of noise whereas is not that good noise wise and VSP you will see that will be little bit in efficient because we will have a roll off filter. So that will take additional noise so it will not be that as good as that quadrate amplitude will be similar to DSP as we have told already, so it takes means the modulation and demodulation process is almost similar.

So after doing all this one thing you have to keep in mind that probably we have said over here that for SSB analysis we have already told that noise analysis wise it is as good as DSP there is a gross assumption over here which probably we have not stated implicitly we are told that this SSB can be demodulated with a fresh carrier and we have already seen in our earlier classes that SSB carrier recovery is not that easy. So therefore generally in SSB either people add huge amount of carrier like amplitude modulation because the carrier recovery is not as easy as DSP SC.

So huge amount of carrier once you add that SSB noise analysis also will be as poor or even poorer than your amplitude modulation because you will have to add a huge amount of carrier to really and then employ envelope detection, so it will be similar to that particular process if fortunately you have carrier/ some method either / transmitting π lot carrier or some other method because from the signal itself you cannot extract the carrier because that Hilbert transform and the signal will complicate the carrier recovery process.

So carrier recovery cannot be done as it is like putting a Cos as loop or something like that so there you will have to probably if you wish to do a coherent demodulation you will have to either put a π lot carrier or your carrier somehow in a miraculous way should be synchronized, so that is something which has to be done okay. So we have for our analysis assume that probably that π lot carrier is available and that is why SSB is probably looking to be as good as DSP but that's not the case we should be always keeping that carrier recovery is a very important circuit which we have just taken that it is recoverable problem.

Which is not true mathematically we have already proven that and this completes our amplitude modulation schemes and their analysis complete analysis okay, now what we will do from the next class onwards we will start looking at the another very important modulation which is called angle modulation okay where the amplitude will remain fixed whereas over here you might have seen that we are just putting all the things all the message signal in the amplitude of a particular sinusoidal.

In the next class onwards we will try to see that will keep the amplitude fixed and we will put the message signal in the frequency part or phase part either will vary the frequency of the sinusoidal with respect to our message signal so the modulation will be done at the frequency level or will vary the phase part of it and try to modulate it okay. So that is overall these two things are almost similar you will also means we will demonstrate that so you will see that corresponding modulation is called angle modulation.

And especially the frequency modulation part is famous as FM or frequency modulation and phase modulation is called p.m. but p.m. FM effectively they are almost similar things the FM will be also able to prove that it has a much better noise immunity okay, so that is something we will try to prove with our noise analysis again we will try to give the full-blown analysis of FM means how it performs with respect to noise or in presence of noise and how it is much better than the a.m. noise performance.

So that is something we will discuss and historically probably that is the reason why FM become more important because it has a better noise cancellation technique and it survives means it keeps the quality of the signal much better than a so initially people started with him historically but then people could see that FM is much better modulation technique, so people went to FM but FM initially was not being accepted in a glad manner / the community that is because there is a there was initially means a particular doubt in people is mind that FM might have a very huge bandwidth.

So we will also try to do that particular part of analysis that is really what people have thought that FM bandwidth is probably very huge whenever we do FM modulation because the frequency start varying and then effectively people initially said that FM probably has infinite bandwidth we will try to prove that and we will try to see that probably effective bandwidth is still not infinite it is band limited still because if you can only transmit one FM through the entire channel.

Then it is not good because all others transmission will be canceled out but fortunately that has not happened and this was mostly proven / a person whose name is almost synonymous with him that is Armstrong, so we will try to see what he could prove and make FM fundamentally very important modulation scheme in the next class thank you.