Communication Engineering
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Lecture – 12
VSB Modulation (Contd.)

We will let us recall that last time; we were looking Vestigial Sideband Modulation.

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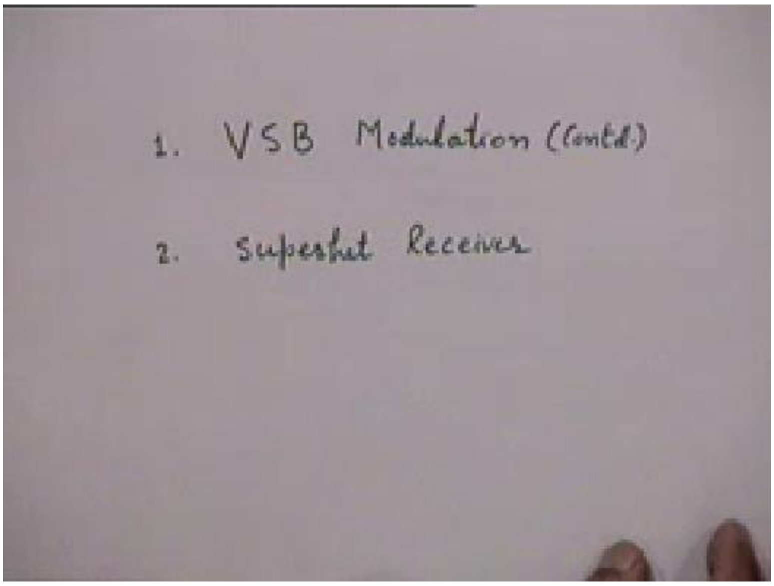

We will continue with that discussion, today for some time and try to wind it up. And in the rest of the time I will introduce the concept of what is called super heterodyne receiver for amplitude modulation systems which will answer number of questions that have been raised from time to time abouttunability, etcetera of the receiver. So we will look at these two issues today.

To recollect our discussion on vestigial sideband modulation, we said that it is useful to have a sideband filter mechanism in which we do not try to cut off one of sidebands completely, but permit a portion of the other sideband to be also transmitted. Basically, it simplifies our filter problem we are able to work with filters which are more practical which have the requirement of having a gradual roll off rather than a certain cut off. Whether you want a high pass filter or low pass filter depending on whether you want to transmit the upper sideband or lower sideband we need to do this.

However, if you want this feature we need to make sure, that this filter would produce a signal which would lead to a perfect demodulation of a transmitted signal, perfect recovery of the message signal. So, when impose this constraint we found that this gradual roll-off filter which leads to vestigial sideband modulation must satisfy a certain constraint.

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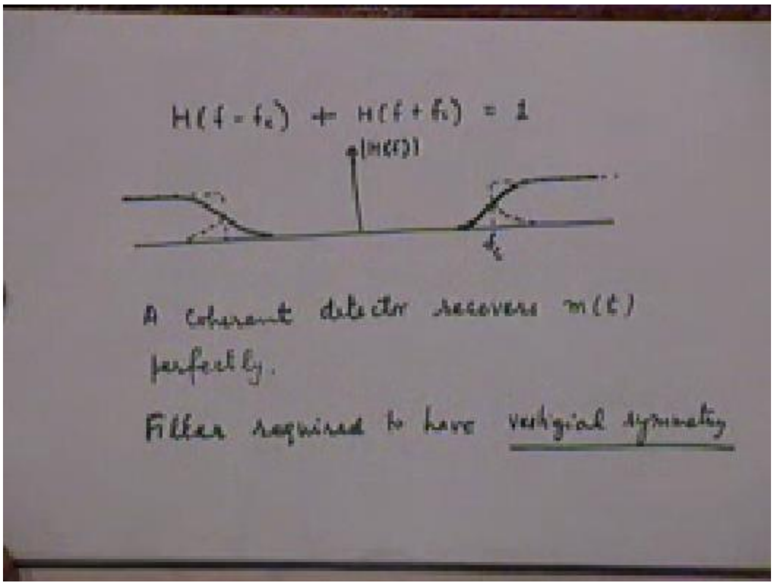

And the constraint derived was this equation, so the filter H f which is the sideband filtering filter vestigial sideband filtering filter must satisfy the condition that H f minus f c plus H f plus f c must be equal to 1this constant one of course, is arbitrary and could be any other constant main thing is it is a constant value. Now, to understand what this means just look at the picture, let us say we talk about high-pass filter take the upper sideband right, this would have been your ideal upper sideband filter, the VSB filter on the other hand, so this we are plotting H of f.

Assuming it to be or you consider this to be mode of H of f, for the VSB filter we have permitted that this at the around the cut off frequency we have a gradual roll-off. So the transfer function looks something like that and basically what we are saying is this, transfer function should have a shape that satisfies this condition. When it satisfies this condition, we have a coherent detector would be able to perfectly recover the message signal m t back, this is what we discussed.

Let us see what this means, this constraint means on a filter transfer function can you guess, what it would it is a kind of symmetry implied here. For example, if you look at this condition carefully this is basically saying that if I hold this portion of the roll-off back in to this region. Then mirror image of this around this point, the head of these two curves they turn out to be a flat curve, a constant curve, so there is a kind of symmetry involved around of f c.

So, this kind of symmetry that you want the filter to have is called vestigial symmetry, so the filter required to have what is called vestigial symmetry, so the filter characteristics must be must have this vestigial symmetry. This is the conclusion of the discussion we had that day, so basically what this implies is that when you fold this back in to the region the sum of these two curves the solid curve and the dotted curve turns out to be a constant one.

Think about that it is really obvious, now if you were to go back to our issue of representation this. So, far we have talked about how to design the vestigial sideband filtering filter at the transmitter, so that the message is perfectly recovered.

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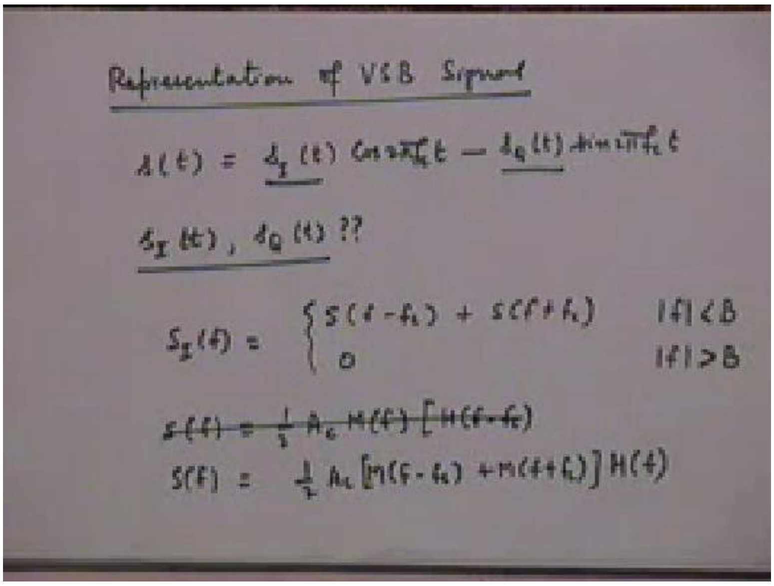

Now, let us look at the issue of representation of the VSB signal yes please, not the advantage it is a requirement, it is not that we are seeking any advantage. Unless we have the vestigial symmetry we will not be able to recover the message signal back, this our entire discussion was based on how to design the filter H of f. So, that m t is recovered by the coherent detector we wanted the output v naught t of the coherent detector to equal M t, k times m t

So, if we want that to happen this condition must be satisfied, we cannot choose any arbitrary filter with gradual roll off, to do the VSB filtering that is the implication, you have to have a filter which has the symmetry around f c the roll off portion must exhibit the symmetry if the perfect recovery is to be made possible. So, the VSB filter has to satisfy this constraint, it is not the advantage we are talking about it is the requirement, so please understand that.

Now, like the other modulated signals that we have discussed, so far the VSB signal is also band pass signal and we know that every band pass signal has quadrature representation. So, let us say, it has an in phase component s of I t into cosine 2 pi f c t and a quadrature phase component s Q t s of Q t sin 2 pi f c t, this is in phase component and this is the quadrature phase component which every band pass signal must have for the DSBSC signal s of I t is m t and s of Q t is 0.

For the SSB signal single sideband signal this is m t and this is Hilbert transformer of m t, the question is what are the in phase a quadrature phase components, for the case of VSB signal that is what we need to understand. To proceed further, let us look at the frequency domain relationship between s If and s f, how can we express the spectrum of s I t in terms of spectrum of s t, to do that. Ask yourself a question, how will you recover s I t from s t we already know that remember we discussed quadrature multiplexing and demultiplexing. If I want the in phase component from this message signal s of t, what should I do,

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I have to do coherent detection with respect to cosine 2 pi f c t; I must have a coherent detector with a local carrier of

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Cosine 2 pi f c t as the carrier signal multiplies these two and passes the product to a low pass filter. So, if I multiply s t with cosine tow pi f c t what is the spectrum of that signal that will s f minus f c plus s f plus f c and that spectrum with this spectrum would have a component around baseband and a component around 2 f c.

Out of that, when you do the low pass filtering what do you do, you will take only the component around the baseband, so basically what we are saying is that the signal s I t that you will recover will have spectrum which equals this. For f less than the bandwidth mode of f less than the bandwidth, but equals 0 for mode of f greater than bandwidth, because you are low pass filtering it, this has two ((Refer Time: 11:33)) non 0 in this region and non 0 around 2 f c.

That portion is being cut off by the low pass filter and therefore, the spectrum of s I t that you have in this representation must be given by this expression, are all with me. Because you can see that s I t can be obtained from s t by multiplying s t with cosine 2 pi f c t and passing the result to low-pass filter and that fact leads us to this expression, for the spectrum of s I t in terms of the spectrum of s t, any questions in that.

Now, you may recollect, that we had an expression for s of f as half A c times M f into H f minus f c, not this of f is your VSB signal how was it being generated. You are generating in by first multiplying m t with cos omega c t and then passing through the VSB filter. So, that that spectrum would be M f minus f c plus M f plus f c this is our DSBSC signal for the frequency domain and we are passing it through H of f, so this was s of f. Now, use this result hereto get an expression for s I f.

And we will do similar exercise later for s Q f, but let us first complete the exercise for s, if are you with me, we are saying that this the expression for the spectrum of s t which is a VSB signal vestigial sideband signal. If I use that expression here, I get the spectrum of the in phase component of the VSB signal.

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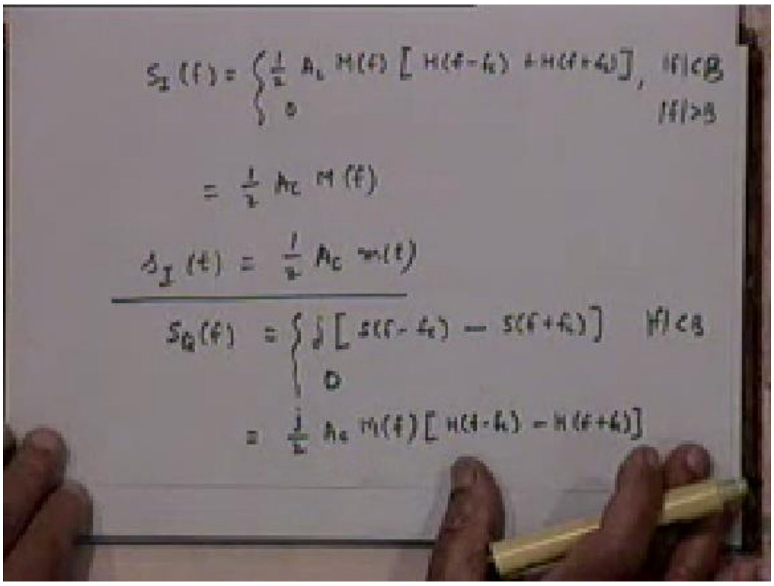

So, this would then become s I f would be able to tell what it will be, you will have, if you remember we also discussed what was the nature of s f minus f c plus s f plus f c. We have gone through one further step, when you are looking at the demodulation of DSBSC signal and that contained four terms. Obviously, because two terms come from here and two terms will come from, each of these two will have two terms and therefore, we get four terms.

Out which only two were in the baseband, which is what you want, the other two terms where around 2 f c. So, if you substitute that here you get four terms, two of which are in the baseband and those two terms we need to look at and those two terms if you recollect or if you have your notes in front of you would or. In fact, obvious and we just look again that it is obvious it will be half A c M f into H of f minus f c plus H of f plus f c. In fact, it is from this point that we derived our requirement, so this should be equal to constant, so this is what you will get.

Now, for the spectrum of s I f, we know the fact that this filter is anyway satisfying the constraint the condition is equal to 1, this becomes half of A c times M f which itself is of course, limited to mode of less than B because M of f is supposed to have a bandwidth of B. So, I do not have to qualify this any further, so what does it mean, what is s of I t half of A c into m t. So once again the in phase component remains the same, whereas the DSBSC signal, whether it is SSB signal and now whether it is a VSB signal.

The in phase component remains proportional to message signal m t and that is the reason we are able to recover it through coherent demodulation and that was of course, made possible. Because we have satisfied this constraint this plus this must be independent of f, must be a constant. So, I think everything falls in place only thing left to consider is s Q t, so if we go through the same exercise once again can you tell me, what will be the spectrum S Q f in terms of spectrum of s t.

So, we are going back to the expression of how will you, I mean the answer to this question will be how do we require s Q t from s t and that will give us the relationship in the frequency domain also how will you recover it, you multiply this with sin 2 pi f c t and pass the results with low-pass filter. So, what will be the spectrum of s t sin 2 pi f c t that is what we will see, so it will now instead of, s f minus f c plus sf plus f c it will be j times, so f f minus f c minus s of f plus f c.

Just, remember what is sine 2? pi f c t it is pi f c t minus f c t divided by 2juse that result, use the frequency translation before you transform and this is what you get, straight forward. So, this is of course, this will be the result again as you know when you do this, when you when you pass s t. When you multiply s t with sin 2 pi f ct you get two times one around baseband and the other around 2 f c. So that you low-pass filter result, so we are only looking at the portion of the spectrum this spectrum for.

Mode of f less than B once again because outside this it has to be 0 because of low-pass filtering, s of Q t will also be a low-pass signal. So this if you follow the again substitute for s of f and go through that you get j by 2 A sub c M of f what will the difference. Instead of this plus this we will get this minus this, everything else will remain the same. That is what we will do.

Now, how do you interpret this result, interpretation of this result is that the low-pass signal s of Q t which forms the quadrature phase component of the VSB signal can be obtained from message signal m t by passing the message signal m t to a filter whose transfer function is this. We are multiplying the spectrum of m t with this transfer function, so; that means this signal s of Q t is obtained by pass the message signal m t to a filter with this transfer function.

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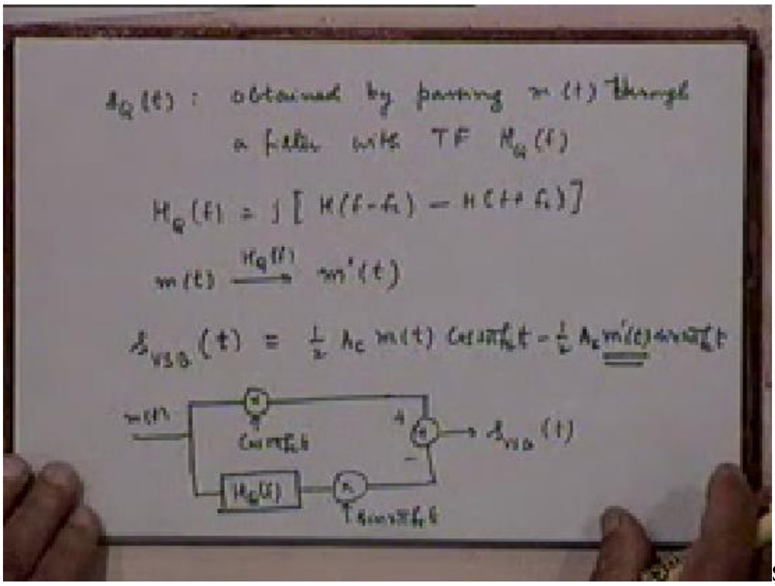

s o s of Q t

So, s Q t can be therefore, considered as obtained by passing original message signal mt through a filter with transfer function. Let us say H of Q f. H of Q f is j times H of f minus f c minus H of f plus f c, let us call this signal m t ((Refer Time: 21:21)) with or passed through. H Q f as m prime t, result that, so far we are s of Q t, I am giving a new name to it, which is I am calling it m prime t. Then, basically what we are saying is that the VSB signal I think I have slightly changed my notation this time we have been using x of t, so far.

Simply, I have shifted to s of t I hope, you do not mind that I did not realize while I simply was changing, so your VSB signal therefore can be thought of as half of A c times m t times cosine 2 pi f c t minus half of A c m prime t sin 2 pi f c t, where m prime t in obtained like this. So, it has some similarities with what is done in the SSB case, what was in the SSB case, this was m hat t the Hilbert transformer, so you are passing m t to a special filter called the Hilbert transformer.

Instead of passing through Hilbert transformer, we are passing it through something different similar, but not exactly Hilbert transformer, it is something slightly different from that. To appreciate what is the difference, let us look at the transfer function H Q f little more carefully, but before I discuss that, this also gives you an alternative method for generating VSB signals, just like the phasing method for SSB, we can have a phasing method for the VSB right what will be the block diagram.

You will have the message signal m t coming in, the in phase part will be obtained by simply multiplying with cosine 2 pi f c t and the quadrature phase part before we multiply by sin 2 pi f c t. We will pass it through, this filter H Q f and then multiply this with sin 2 pi f c t the quadrature carrier and simply add the two are subtract the two, depending on which sideband you want to transmit fully or partially. So that is your and the detection or the demodulation can be done coherently.

Let us just spend a few minutes, now discussing the nature of H Q f what does it look like, let us first recollect or let us as we notice this discussion that I have had is valid for SSB as well. Because SSB does satisfy this requirement the SSB filter, this filter, this condition is satisfied this plus this is equal to 1 because this portion is 0 and this portion is 1 something like that. And therefore, that condition is satisfied, this whole thing should be applicable also to SSB signals.

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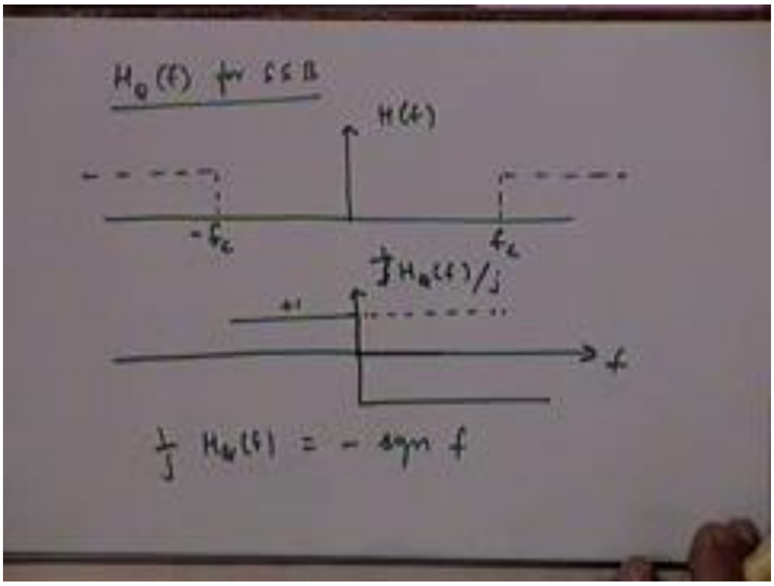

So, therefore, let us first look at the nature of H Q f for SSB signals, for SSB signal what is your transfer function. It is like this basically what we hope to see, what we hope to see that if we derive H Q f from H of f from this condition that we have just discussed. We must get the Hilbert transformer because we know that, that is how quadrature component of SSB is generated to the Hilbert transformation of m t. So, H Q f should turn out to be the Hilbert transformer.

Let us see whether it does right, this is your H of f centered around, this is the high-pass filter this is the cut off frequency of f of c. Now, let us look at this filter j times, let me plot j times H Q for1 by j times H Q f or H Q f divided by j because if you recollect, If we look at this expression H Q f is this. So, I am taking j on the left on side and plotting H Q f upon j that is equal to H f minus f c minus H f plus f c. So, H Q f, H f minus f c takes you shifts this to right to this point so you get this.

Of course, this portion will go around 2 f c. We are not interested in that, we are only interested in the portion in the low-pass region. H f plus f c will shift this portion to the left and you are taking the difference of these two. So, here you will get plus 1 and here you will get minus 1, so you get your signal function back, so H Q f upon j is equal to minus signum. So, basically what you get is H Q f upon j is equal to minus signum of f and that is precisely what the Hilbert transformer is H sub Q f is equal to minus j time's signum f which is a Hilbert transform.

So, as expected this theory leads us to a filter H Q f which is nothing, but the Hilbert transformer for the case of SSB, for the VSB the filter would be slightly different, that is about all, let us what the filter be in this case.

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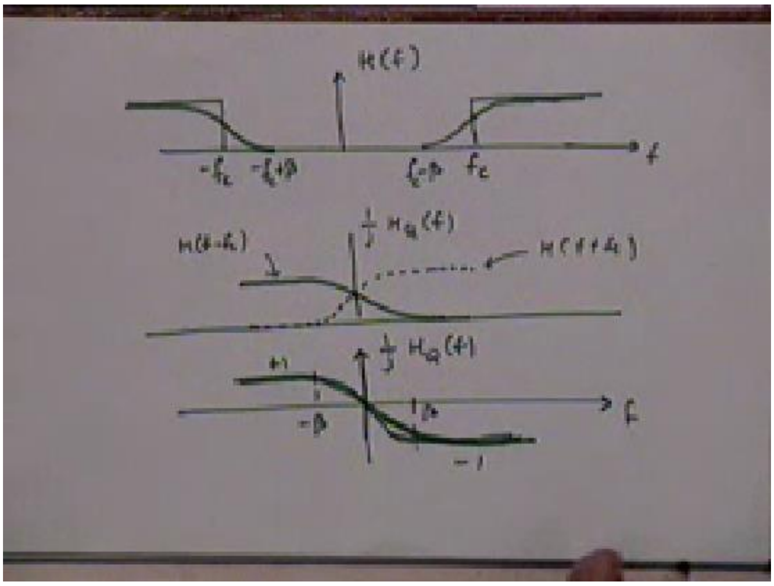

This is the VSB filter f c this goes to left side f c minus beta, this is minus f c and this comes to f minus f plus beta, this is a VSB filter. Now, let us apply this result here and plot once again 1 by j times H sub Q f, when you shift this to right what we will get is something like that, when you shift this to the left we will get something like that. When you subtract the two this is coming from H f minus f c, this is coming from H f plus f c, by looking only the translation towards the baseband. Of course, each of these components also gets translated to 2 f c which we are not looking at.

So, now if we subtract the two, the filter that you will see is going to have this kind of characteristic, so this is 1 by j H Q f, I have not plotted this very nicely but it will be symmetrical as it should be. It will, this kind of characteristic we are subtracting from this. So, at this point what will you get half minus half which is 0 it is passing through the origin and then of course, at this point we get a gradual decrease towards 0 and at this point become more and more negative till it becomes minus 1. So, this is plus 1 here minus 1 here and between minus beta to plus beta there is a gradual roll-off, so H Q f is j times this.

So, it is not exactly signal function anymore. That is the only difference. So, it is very similar to the limit as beta tends to 0, this will reduce to the signal function, are you with me. So, let us try to wind up the discussion on vestigial sideband modulation, in vestigial sideband modulation therefore, what we have learnt is we will use slightly more bandwidth than the bandwidth of a SSB signal, the amount that you will like to use more will depend on the kind of flexibility you want in your filter designer.

The value of beta that you use is the design parameter. You can choose it to be a small value, you could choose it to be a large value depends on how easy it is to a realize a filter that you want or how much cost you are ready to incur for designing for making the filter because whether you want less complex filter or more complex filter. If you want a gradual roll off you can make it a less complex filter only thing is cost will be in terms of a larger value of beta; that means, slightly poorer bandwidth efficiency.

But, if you pay this cost the advantages will be that you can save considerable amount of bandwidth, still for large bandwidth signals like picture signals, which have rich low frequency content and VSB. As we learned as we discussed, last time is particularly relevant to those signals which have a fairly large amount of low frequency content. And therefore, you cannot tolerate any distortion in the low frequency region. But you clearly face distortion, which is likely to happen. When you design a sharp cut off filter and then the phase characteristics at the point of cut off will typically be highly known which will be very, very undesirable. Particularly for signals like picture signals and fax signals, so in a nutshell that summarizes discussion on VSB. So, therefore, now we can imagine for TV transmission, this would be a natural choice when you are transmitting pictures. We will discuss TV transmission later, but you can keep this in mind, where vestigial sideband signals will be particularly the right kind of modulation to use for TV picture transmission. In fact, that is the case, so picture transmission in television actually uses vestigial sideband modulation.

Because you have very large bandwidth signal here, if you do not cut off one of the sidebands you will too much bandwidth 5 megahertz will imply 10 megahertz bandwidth around the carrier, which is very, very large amount of bandwidth. So, to save that bandwidth and still not have these problems that we have discussed you decide to go for VSB because you cannot go for SSB we discussed we will return to this point when we discuss TV transmissions. Any questions, none everything is clear, good let us hope it is really clear.

Student: ((Refer Time: 34:29))

No, no m prime t is not the Hilbert transformer of m t, no neither m prime f, I think you have not understood what I have said then not only for the case of SSB.I just proved to you that for the case of SSB this theory that I have developed is consistent with what we have discussed earlier that is all I said. Whatever theory we have developed for the general sideband filtering is consistent with what we have learned for the case SSB signal because this filter H Q f then reduces to the Hilbert transformer.

But, in general it is this kind of a filter which is not a Hilbert transformer, yes the roll-off that you will have will be dependent on the value of beta, if the value of beta becomes 0. We get the single sideband phase back and it reduces to the Hilbert transformer. Yes please.

Student: ((Refer Time: 35:46))

Neither in SSB nor in VSB, yes we will discuss that later it is a good question, but the answer is that actually we do not, we cannot tolerate any phase in coherence in quadrature multiplexing not even one bit. There will be a cross talk, if there is even a small amount of either frequency difference or phase difference, which you do not which we do not like to have. Where as in the case of SSB and VSB if you remember have a second method of demodulation in which I add a carrier locally and then do envelope detection which is much less sensitive to these problems than the coherent detector is. And I will be able to the same thing for VSB, one method of demodulating the VSB signals would be precisely the second method that we discussed for the SSB signal that is the Carrier Reinsertion Method, we look at that. In fact, the theory is exactly the same; there is no difference except that m hat t gets replaced with m prime t. So, we have a second method and that it is an existence of the second method which makes VSB and SSB more acceptable than ((Refer Time: 37:15)) I hope I have answered your question.

Any other questions, let us proceed further then, now I am going to slightly change the topic here we are on a subject of broad of amplitude modulation and its applications, but now we are coming to some technical aspects and before that one of the technical aspect. We need to discuss is the kind of receivers that we need to use of course, we have discussed the structure of the demodulator for example, if you are doing am transmission we know what we are going to use, we use we are going to use envelope detector.

And we can use the envelope detector also for the case of SSB and VSB with carrier reinsertion, but at the moment let us concentrate on the amplitude modulation aspect because that is the one of the most commonly used analogue modulations. Amplitude modulation with carrier, because of the simplicity of detector, but as many of you asked at the time of this discussion, what will happen if I want to want to receive different signals.

Because, when you when you looking, when you are having broadcast receiver you do not want to listen to only signal at one carrier frequency. There are so many different broadcasting stations each operating at a different frequency and you want to be able to tune in to any one of them. What are the issues concerned with that kind of a receiver, so we are now looking at rather than a just detector, we want to look at the receiver as a whole as to what all features receivers should have. So that it is possible for me to have detection of a signal of my choice from amongst multiple signals that may be available from the broadcasting areas around you.

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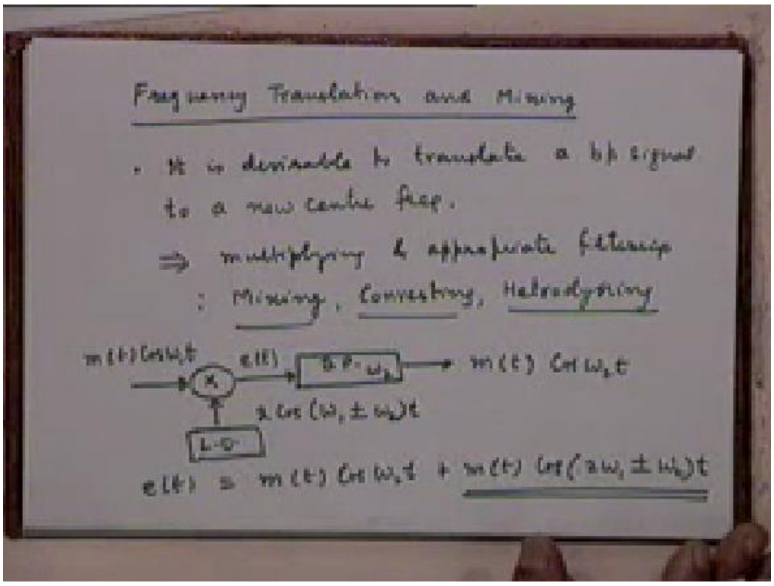

So, to do the discussion I first want to diagnose in to slight generalization of our previous discussion on frequency translation and mixing, these are not new terms we have already discussed these, there is one issue I would like to discuss before I come back to the receiver. So, the term frequency translation, so far we have been using in the context of going from baseband to a band-pass signal or from the band pass signal to a low-pass signal. But actually in communication application the term frequency translation and mixing are used in a much wider context typically.

We sometimes in fact, it is quite often that we need to go from one frequency to another frequency not necessarily from baseband to some f c and back. Arbitrarily we may want to go from band pass signal at center at one frequency to band pass signal center at another frequency. So, let me start with that, It is desirable in many applications, to translate, I mean frequency translate, Ab and pass signal to a new center frequency.

In this the mechanism is the same; the mechanism is multiplying the band-pass signal that you want to translate by a suitable carrier signal. I suitable carrier signal which could be or a corresponding periodic signal and an appropriate signal multiplying an appropriate filter and this process of multiplying appropriate filtering is what we call mixing. And if you remember, I introduced some other names also for this converting and heterodyning mixing, converting, heterodyning, these are the names which are interchangeably.

So, what we are saying is, you have a band pass signal let us say m t cosine omega 1 t for the sake of simplicity I am taking only the in phase component. But you could take a general band-pass signal for general representation, without any loss of discussion without any loss of generality. So, I have taken a band-pass signal which is centered around m t cosine omega 1 t where is it centered omega 1, the center frequency is omega 1.Let us say I want to translate this to a center frequency of omega 2, what should I do.

I should have a local oscillator, hat should be the frequency of this local oscillator, omega 1 either omega 1 minus omega 2 or omega 1 plus omega 2 any one of the two will be fine. So, I take the local oscillator, with a frequency of either omega 1 plus omega 2 or omega 1 minus omega 2 and then when I multiply these two, I will get a sound frequency component and a frequency component. Some frequency component will be at a frequency2 omega 1 will be centered around 2 omega 1plus minus omega 2 which, let us assume as much quite large and can be removed by an appropriate filter, but the component that we want is centered around omega 2

So what kind of filter you need here, not a low pass filter. The band pass filter with a center frequency of omega 2 that will lead to your m t cosine omega 2 t as an exercise. Please check up that if instead of m t cosine omega 1 t, I had return the more general representation of band pass signal that is m t cosine omega 1 t some in phase component and some quadrature phase component.

And through this process you still get the translation intact, I will leave that as an exercise, so this signal at this point. Basically, the theory is that if you look at this signal e t this you can right as m t cosine omega 2 t plus m t cosine 2 omega 1 plus minus omega 2 t the trigonometric identities and through the band pass filtering you are eliminating this component, retaining only this so, this process in general is called mixing. Now, there is a very common problem that happens in this process that one encounters in this process.

It is a common problem in all mixing, all mixers also of this kind it is associated with all mixers of this kind, suppose you take omega 1 plus omega 2 t here and let us say at the input along with this signal, I also have a signal at frequency omega 1plus 2 omega 2 t. So, along with this I have another signal which is centered around omega 1 plus 2 omega 2 t, now what will be the output of this system, have you understood the question the question is if at the input.

With the same system, which is adjusted to signal of frequency omega 1 that is adjusted to translate the signal of frequency omega 1 to omega 2.If at the input of the system I have a signal omega 1 plus 2 omega 2, what will be the output, think carefully. See basically look at the sound frequency argument, what will be the frequency components that you will have here. Omega 1 plus 2 omega 2 minus omega 1 plus omega 2 and of course, some component, the some component we do not have to worry about and what is this equal to omega 2again.

So, this will also be passed by this band-pass filter, so for every frequency that it is supposed to translate there is also what is called an image frequency which it will translate to the same frequency. Not only it will translate omega 1 to omega 2, it will also translate omega 1 plus 2 omega 2 to omega 2, so omega 1 plus 2 omega 2 that frequency is said to be the image frequency of omega 1.

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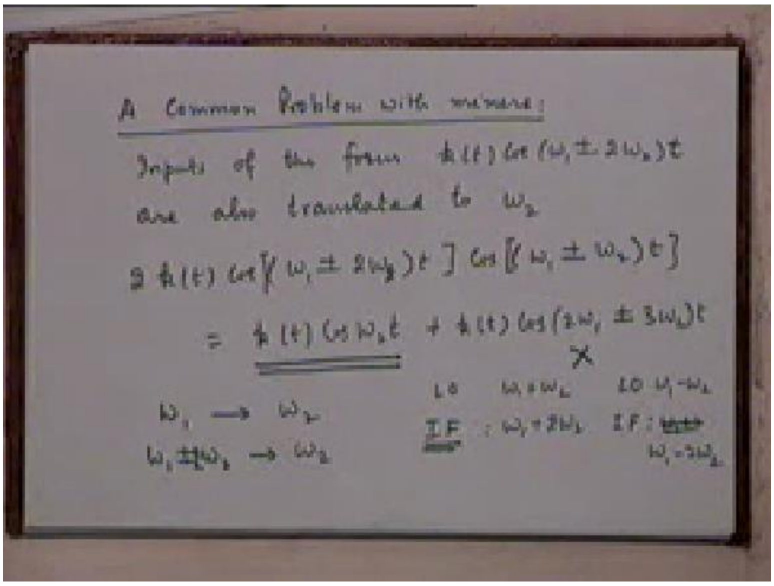

So, basically if you have not understood, let me repeat it, what we are saying is there is a common problem with mixers. So, I want to elaborate on this problem and this problem is as following, the inputs of the form k t cosine omega 1 plus minus 2 omega 2. Here I had taken the local oscillator will be omega 1 plus omega 2.In fact, again omega 1 minus omega 2 then the image frequency would have been omega 1 minus 2 omega 2 basically the sin would be the same.

So, inputs of the form k t cos omega 1 plus minus 2 omega 2 t are also translated to, can you complete the sentence to a center frequency of omega 2 to see this lets us go through the trigonometry what we are saying is2 k t cosine omega 1 plus minus 2 omega2t.When you multiply this with, k t cosine omega 1 plus minus 2 omega 2 t when you multiply that with 2 cosine, what are we multiplying with, what is your local oscillator frequency either omega 1 plus omega 2 or omega 1 minus omega 2.

This leads to two components, one is the difference frequency component and the other which I the sound frequency component and this will again show up at the output of the same band-pass filter, this will get eliminated this is point I was making. So, when you use a local oscillator of this frequency, basically what you are saying is omega 1 will go to omega 2, omega 1 plus minus omega 2 will also go to omega 2.So, this will be 2 omega 2, omega 1and of course, the sign will be same more precisely.

If L o is omega 1 plus omega 2, the image frequency will be omega 1 plus 2 omega 2, if your L o is omega 1 minus omega 2, the image frequency would be this misnomer I should not use if is typically used to denote something else, but let me for the moment use, if this will be minus 2 omega 2.And therefore, if both the signals were simultaneously present the signal at omega 1 and the signal at omega 1 plus 2 omega 2 and you have used local oscillator of this frequency.

The band pass signal output will contain components corresponding to both of them and there will a cross talk you will get some of two signals, is it clear, so that is the problem you need to keep in mind when you are working with mixers. There is concept of an image frequency which has a potential to interfere with this frequency of interest the image frequency. If the signal at that frequency is present, will interfere with the signal of interest which is at frequency omega 1. Is it clear? Basically that is the point that I want to convey, keep that in mind and now we shall discuss. So, we will discuss next time the concept of super heterodyne.

Thank you very much.