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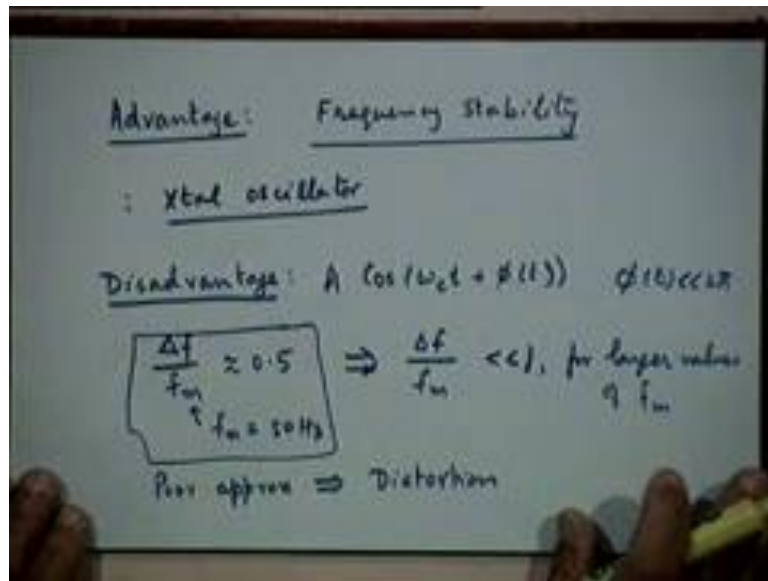
**Lecture - 18**  
**FM Generation and Detection**

Today, we will continue our discussion on FM Generation and Detection that we started yesterday. And to summarize our discussions yesterday we learnt that we could generate FM signals using two different kinds of methods the direct and the indirect, we will look at the indirect method may be the Armstrong method, and how it should be used in practice.

We looked at the typical commercial example that would be a commercial design for an FM transmitter, in which we have to choose and measure a narrowband FM generator. We had to initially make a narrowband FM generator followed by frequency multiplication. The narrow band FM generator generates a single with a small modulation index; frequency multiplication produces an FM signal of a wideband signal of a large modulation index.

In the process the carrier frequency also gets multiplied and to get the requisite carrier frequency we may have to go through one or more stages of frequency multiplication followed by frequency transmission, so these are the things that we discussed easily. The advantage of using this method, you may ask, why should we use this method at all, after all we have to go through so many stages of operation? The main advantage is the frequency stability that is associated with indirect method, now let me explain what that means.

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So, one of the main advantages of the indirect method is what you say frequency stability and this may look little odd, because we are talking about frequency modulation here or phase modulation here. Frequency and phase modulation are naturally associated with variation of instantaneous frequency, so what do you mean by frequency stability. Whenever we have a transmitter or anything that modulates any carrier for that method, for various reasons it is desirable that the carrier frequency remains fixed.

What do you mean by remain fixed, basically what we are trying to say here is that, it does not drift with time; all electronic components are subjected to electronic drifts, thermal drifts. Because of temperature variations, there can be variations in the properties of the circuit and one of the properties that one has to worry about in oscillators is the frequency of oscillation as a result of these drifts.

For example, as I mentioned earlier the frequency of oscillation is typically dependent on the resonant circuit, the LC components, the LC components themselves will keep on varying if there is a temperature variation. And that leads to a drift in the frequency, center frequency of the oscillation, so that is not desirable in any electronic system. Because that will create..., because for example, if there is a significant frequency drift at the transmitter and the receiver does not know it, then the receiver will get affected

And there are so many millions of receivers particularly in a broadcast situation. So it is very much desirable in many broadcast applications or communication applications to

keep your frequency of operation fixed and do not allow to drift too much. Typically fixed frequency of operation requires highly stable sources of resonant frequency and typically this term what are called crystal oscillators. Crystal oscillators which use a quartz crystal in resonant circuit, electrical current can be modeled as a resonant circuit, the parallel LC circuit of very stable sources of oscillation.

If you use the crystal in the tank circuit, in the parallel resonant circuit of an oscillator it is a highly stable source. Now, once you start with the highly stable source like this, there is no way you can change its frequency, whereas in frequency modulation what you require the change in frequency. So, these are contradictory requirements you cannot use crystal oscillators in the oscillators, crystal oscillators at the same time have variation in frequency.

Whereas, the indirect method does not have this problem, because our starting point is not a generation at all, it is generating like an AM signal, the narrowband FM signal generation is based on just the carrier and a DSBSC modulator. The DSBSC modulator has the fixed local oscillator which is used by a multiplier, the other input is the message signal  $m(t)$ , and then  $m(t)$  is being multiplied by  $\sin \omega_c t$ .

And no direct frequency variation is required, the local oscillator which is producing  $\sin \omega_c t$   $\cos \omega_c t$  can come from a highly stable oscillator, the quartz crystal for example, so that is what I mean by this advantage. The center frequency of the carrier is not subject to drift, it is highly stable, it can be made highly stable by making use of a crystal oscillator as the starting point, after that you are just multiplying the frequency.

You are not doing any further generation of the center frequency, so that is the main advantage is there a disadvantage is this point clear, why it will produce a stable frequency, is this point clear. Now, what is the disadvantage there is also disadvantage which I think you should be able to tell me, which we have discussed briefly earlier I am doing, the disadvantage is that the narrowband FM generation is based on an approximation of a FM signal.

Because, we represent  $\cos(\omega_c t + \phi(t))$  by making, we represented in terms of quadrature components assuming that  $\phi(t)$  is much less than  $2\pi$ , there is an approximation involved here. And this approximation is quite good, for example if you

consider the design that we discussed yesterday, we made this condition, we chose this condition this is of the order 0.5.

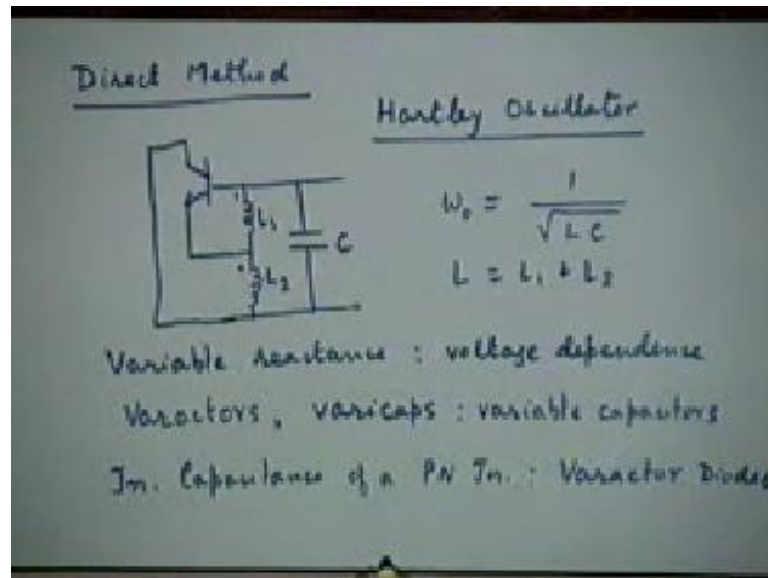
Of course, if this is of the order of 0.5, we chose this for doing this we chose FM to be the lowest frequency component present in a message signal, it was of the order of 50Hertz. I am explaining that now, the reason why we chose to be the lowest this, because if we ensure that this is small for the lowest frequency component, this will automatically imply that this condition at  $\Delta f$  by  $f_m$  or  $\phi$  of  $t$  is much less than 1 for the larger values of  $f_m$ .

So, it is definitely going to assure that this is much less than 1 for larger values of  $f_m$  of  $f_{sub m}$ , so as far the higher frequencies are concerned in the modulating signal for those frequencies the approximation will be quite good, we do not have to worry too much. But, for the low frequency components present in the input signal, this approximation may not turn out to be very good, because it is not much really less than 1. We are still it could be much better, but then if you make it too small the multiplication factors explode and the design becomes impractical.

So, these are the considerations which kind of requires to work with some kind of, may be somewhat poor approximation at lower frequencies and the poor approximation leads to some kind of distortion. That is the signal that you generate is not a pure FM signal, it is slightly distorted version of the FM signal. Because, you are not generating this directly, you are generating something that approximates this and that approximation is poor that there is a distortion.

Typically this distortion will take the form of undesirable amplitude variations in the signal, which original signal does not have, the original signal has constant amplitude, but this approximate signal will have some amplitude variations. So, these are the pros and cons of using the indirect method, so at lower frequencies there is a slight fall, but otherwise it is a very good method and one may be extensively used.

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And against these indirect methods, let us now spend little more time on the direct method or direct generation of FM. As I mentioned the direct method can be based on either oscillators or relaxation oscillators, resonant oscillators or relaxation oscillators. I will discuss here the case of the resonant oscillators and just to set the ball rolling, let me just draw a kind of schematic diagram, schematic diagram of such an oscillator. This is the schematic diagram I have shown a transistor, but I have not shown many of the other components that this circuit will have.

For example, the transistor has to bias the amplifier, so to have the biasing circuits I have not shown any of that here. How will you look at the AC modulator which is relevant for our discussion? So, this is let us say the transistor is somewhere, some amplified and there is a resonant circuit here, this parallel AC circuit and as you can see there is a feedback mechanism in this.

These two inductors that I have shown which are otherwise in series, they have mutual coupling, which is shown by this dots and that essentially means that if you take the output of the circuit as from here. If this is your output, the portion of this output is also being fed again at the input, so the coupling and that gets amplified etcetera and you have this oscillating up. So, basically as I mentioned the basic principle, I am not really going into a very detailed discussion of how an oscillator works here.

So, the important point is that, they has to be a feedback, the feedback is if you consider the feedback path here and that feedback deals up the circuit to produce the oscillation at a certain frequency, because this positive feedback occurs at the resonant frequency of the circuit. So, whatever is the resonant frequency of the feedback circuit is really the frequency of oscillation, so other than that I will not go into a detailed discussion, there are many kinds of oscillators this particular one is called a Hartley oscillator.

I am sure you will have a much more detailed discussion of the Hartley oscillator in our analog electronics course, so the oscillation frequency of this will be  $\frac{1}{\sqrt{LC}}$ , where this L let us say this is  $L_1$  and  $L_2$ ,  $L_1$  plus  $L_2$ . In other oscillators this feedback circuit will be in some other configuration that will be major difference for one oscillator to some other oscillator. The feedback mechanisms are designed to be in different configurations, suppose let us say in the present discussion that such an oscillator is available to us.

The question that we have to address here is can I use the oscillator as a source of FM generation, if so what do I need to do with the circuit, what features do I need to have in this circuit, so that this becomes an FM signal generator. Any suggestions for that, would you like to make any suggestions what do I need to do, I need to modify a make this frequency of oscillation voltage dependant. How can I make it voltage dependant, some obvious answers are there. I am sure you can produce one or two answers, at least one obvious answer

Student: ((Refer Time: 15:45))

You require a variable reactors of some kind, either you have inductor L or you have capacitor C should be made to depend on the once I applied voltage, it is as simple as that. You have to think of the most simple solution that comes to your mind and that is the obvious thing that comes to your mind, that if you want your oscillation frequency to depend on the voltage. Basically, the components will decide the oscillation frequency should be made to depend on the voltage.

So, you require some kind of variable reactors, not only variables I am not saying passively variable, but actively variable that is voltage dependant, it must have a voltage dependence, reactors or reactance's which satisfy this property, which have this property are known by the name of varactors variable reactors. Or mostly the time it is very

difficult to find naturally occurring voltage dependant inductors, the very few devices hardly that are known which have this property.

On the other hand, the voltage dependant capacitance is something that you can easily build, so most of the times these varactors are actually variable capacitors and sometimes simply known as varicaps, so these are variable capacitors or voltage dependant capacitors. So, an example of a variable capacitor or a voltage dependant capacitor does something come to your mind again, would you like to suggest something that you know which can act like a variable capacitor or a voltage dependant capacitor.

Any device that

Student: ((Refer Time: 18:08))

The PN junction, the junction capacitors, the junction capacitors of a PN junction, so that is the most commonly form of the variable capacitance. So junction capacitors particularly the reverse bias junction capacitors, in the reverse bias mode. In the reverse bias mode that the PN junction has a depletion region and across the depletion region you have a capacitance and there is voltage dependence. So junction capacitance of a PN junction is an example.

So, sometimes these PN junction diodes which are specially used to exploit this property, they are design to give you a good variation of the capacitance voltage and they are known as varactor diodes of the varicaps. Now, if the actual dependence of this capacitance on the applied voltage may not be a linear relationship, it is typically non-linear relationship there is a square root kind of relationship available, but let us assume for a moment that we have a linear relationship.

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$$\begin{aligned} C &= C_0 - k m(t) \\ \omega_0 &= \frac{1}{\sqrt{L C_0 \left[ 1 - \frac{k m(t)}{C_0} \right]}} \\ &= \frac{1}{\sqrt{L C_0}} \left[ 1 + \frac{k m(t)}{2 C_0} \right] \quad \frac{k m(t)}{2 C_0} \ll 1 \\ &= \omega_0 \left[ 1 + k' m(t) \right] \\ &= \omega_0 + k_f m(t) \quad k_f = \frac{k'}{2} \omega_0 \end{aligned}$$

So, basically what we are saying is that the capacitance  $C$  of this kind of device varies something like this, it oscillates or it varies around some fixed values of the capacitance. So,  $C$  sub 0,  $C$  sub o is the nominal value of the capacitance without application of outside voltage and as we apply an external voltage it varies in this manner. So, if this is the message signal and you apply this across the reverse bias junction, this will create a junction capacitance which has...

Actually it will not be such a nice linear relation, but let us assume for the time being that it is a such a weak relationship, actual relationship will be a little more complicated we will discuss ((Refer Time: 20:25)). So, what happens to your frequency of oscillation it becomes  $1$  by square root of  $L C$  naught into  $1$  minus  $k$  time's  $m$   $t$  upon  $c$  naught. I will just replace  $c$  with this expression let us take  $c$  naught outside this expression and that is how you do it.

This as you can see can be written as  $1$  by square root of  $L C$  naught into  $1$  plus, now this is I am using a approximation once again assuming that this fraction is small. I can write this as  $1$  plus  $k m t$  upon  $2 c$  naught, because this is raised to the power of minus half and being a binomial series expansion taking only the first term you get this. If we assume  $k m t$  upon  $2 c$  naught it is much less than  $1$ , this is nothing but, actually I should not call this omega naught now I will call omega, this factor is your omega naught as defined earlier.



The nominal carrier frequency, carrier frequency that, you will get in the absence of the modulating signal when  $m(t) = 0$ , when  $m(t) = 0$  you will only have this term. So, this is  $\omega_c + 1$  plus, let me call this factor  $k_f k$  by  $2\pi f_c k$  no. I will call it  $k_f'$  or let me call this as  $\omega_c + k_f$  times to put it in the standard form.

So, your  $k_f$  will be equal to  $k_f'$  into  $\omega_c$  and note that even when this condition is satisfied  $k_f$  that you have here the frequency modulator constant is proportional to constant can be quite large, even though this condition is satisfied. Because, we are multiplying  $k_f'$  with  $\omega_c$  to get here, so you can directly obtain fairly large modulation indices, because you do not have a restriction here, we have a restriction is  $k_f'$ .

Because, your  $\omega_c$  is typically quite large this factor is sufficiently large for us to find wide band FM directly, so this is the principle of the direct FM generation. In this capacitance is the crucial way, variable capacitance or voltage ((Refer Time: 23:46)) that means, we can said in come from a reverse biased PN junction. So, the principle is really very simple, now the detail design will depend on the design of the oscillator. And then, you replace the capacitance in that oscillator with this reverse bias junction capacitors and that is it, that will become the FM generator.

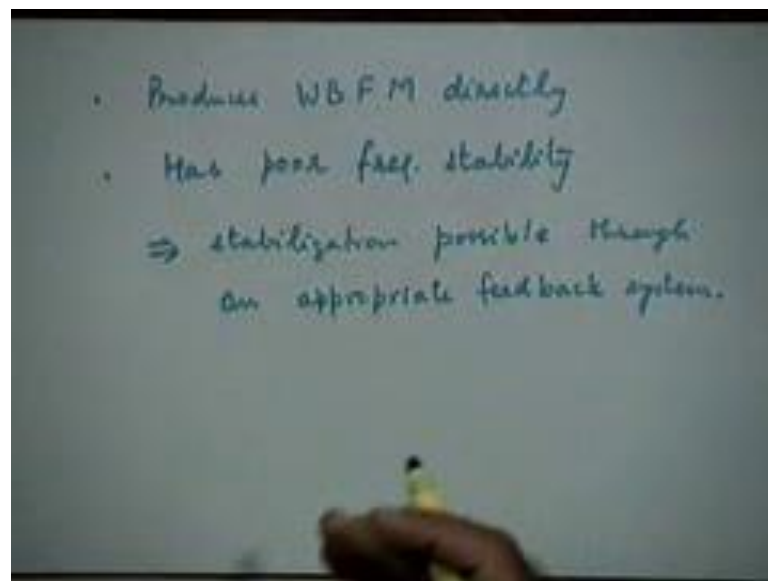
Are you with me on this, now what are the pros and cons of this, vice versa the direct method or indirect method, any comments on that. Let us talk in terms of the factors that we discussed earlier, namely we consider two factors which were important when we discussed the indirect method. One was frequency stability, the other was distortion, and there is no distortion here. If we have perfect capacitance dependence like this, there is no distortion here; we get a perfect FM signal.

Of course, if it is phase modulating signal you need to generate instead of MFT you will work with the derivative of MFT that will be the only difference. Otherwise there is; however as far as frequency stability is concerned what you can say here will it be as good no, because you do not have a crystal oscillator here. The feedback circuit, the LC components are not coming from the quartz crystal, they are have to be separately realized one of the components is the variable capacitance.

In fact, that capacitor by its very name is variable and therefore, it is subject to drift and all the effects that are associated with any passive component or active component in this case. So, all the drift effects will very much be present and you cannot be sure that the  $\omega_0$  that you are going to get will be constant for all time will not drift, so the frequency stability is greatly affected in this case. And one has to use fairly sophisticated feedback systems to ensure that the system produces the required stability that you want from it.

Even then when you do not have frequency stability in the basic oscillator, it is possible to generate a complete system which has other components to make the overall system have frequency stability, but that becomes a complicated system. I will not go into how an FM generator for example, can be made into a stable FM generator, but it can be done but that becomes very expensive. So, these are the pros and cons of the direct and indirect methods, do you have any questions to illustrate.

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So, let me just comment, so it produces advantages, produces wideband FM directly, because it will have very large value of  $k_{\text{sub } f}$ , the big disadvantage is that it has poor frequency stability by itself. However, it can be made to have required frequency can be stabilized, stabilization is possible, so an appropriate feedback system which makes it quite expensive. If we have done some discussion on feedback, we must have also learned that feedback; particularly negative feedback can be used to stabilize systems.

So, one can design a special negative feedback system, control system or some kind which you shows that, even though there is a variation in frequency due to drift if the fact can be minimized. So, with this I think I have given you a reasonably good idea of how FM signals can be generated, there is one more class the methods based on relaxation oscillators, but for lack of time I will not go into discussion of that, I like you to read it by yourself. Now, let me written briefly, before I go to FM detection, demodulation of FM signals I want to written briefly to our discussions on bandwidth of FM signals.

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The image shows a whiteboard with the following handwritten text:

B. W. of Angle Modulated Signals

$$a(t) = \int_{-\infty}^t m(\alpha) d\alpha$$

$$x_{PM}(t) = A \cos [\omega_c t + k_f a(t)]$$

$$\tilde{x}_{PM}(t) = A e^{j k_f a(t)}$$

$$= A \left[ 1 + j k_f a(t) - \frac{k_f^2}{2} a^2(t) + \dots \right]$$

$$x_{FM}(t) = A \left[ \cos \omega_c t - k_f a(t) \sin \omega_c t - \frac{k_f^2}{2} a^2(t) \cos \omega_c t + \dots \right]$$

I want to written to briefly just to emphasize a few points which I have already mentioned, but slightly different way, so let us return to this topic. Basically I want to have a good, if you remember we drew phasor diagram of AM signals, then we discussed detection or demodulation of AM signals using the envelope detector in the presence of interference.

It is important to have a feel for the phasor diagram of all these sectors, so this discussion besides generating some other insights, will also lead us towards this phasor diagram that we are talking about. The first point is I will now do this discussion first for the case of general signals, because so far when we discuss the bandwidth issue, we did the discussion with respect to a very special modulating signal, namely the sinusoidal modulation signal.

Let me take a general signal, so let me define simply start with FM, let me define  $a(t)$  as the integral of  $m(t)$ , so your FM signal is  $A \cos(\omega_c t + a(t))$ , because FM signal is  $k_f$  times integral of  $m(t)$  which I have written it as  $a(t)$ . The complex envelope of the signal, what will be the complex envelope of the signal; it will be simply  $e^{j(\omega_c t + a(t))}$ . This exponential function I can expand to power series,  $e^{j(\omega_c t + a(t))}$  to the power  $x$  I can write the power series for that and write this as  $A \left[ 1 + j k_f a(t) \right]$ .

Let me write one more term minus  $k_f^2 a^2(t)$  and soon, let me go back to the FM signal here, how will I do that multiply by  $e^{j(\omega_c t + a(t))}$  and take the real part of that product. If you have to do that the original signal you can write once again would you like to prompt me what will you get here, this will get multiplied by  $e^{j(\omega_c t + a(t))}$  when you take the real part of this you will get simply  $\cos(\omega_c t + a(t))$  here.

Because,  $1 + j k_f a(t)$  into  $e^{j(\omega_c t + a(t))}$  real part of that, what will you get here  $j k_f a(t)$  times  $e^{j(\omega_c t + a(t))}$  and the real part of that, this will become minus  $k_f a(t)$  into  $\sin(\omega_c t + a(t))$ . Then next term will be  $k_f^2 a^2(t)$  and once again you will get  $\cos(\omega_c t + a(t))$ , because this is getting multiplied once again  $e^{j(\omega_c t + a(t))}$  and so on and so forth.

So, it will give you different way or different look at the same signal, this is also a series of expansion of some kind, earlier also we obtained series expansion, but they are made the differences in these two series expansion, this is a power series expansion. There we use a Fourier series expansion of  $e^{j(\omega_c t + a(t))}$ , because that was a periodic signal, this is not a periodic signal  $a(t)$  here is something arbitrary, that is why we have taken the first power series expansion.

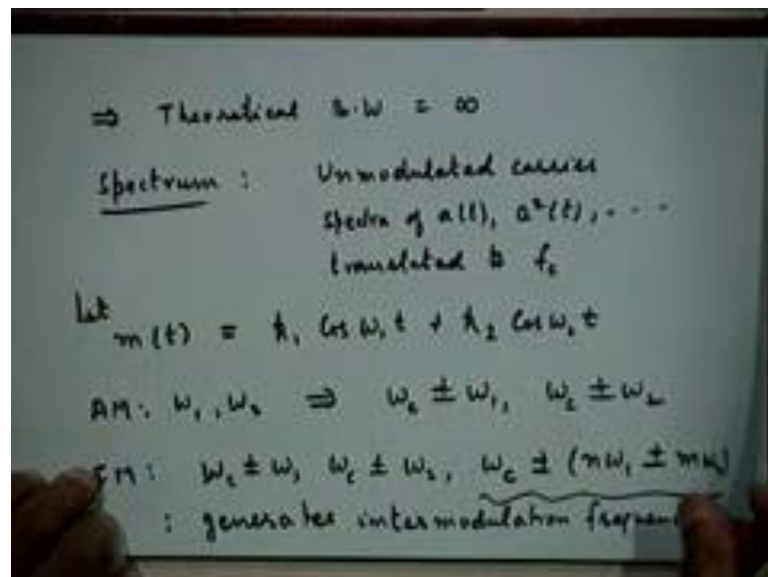
This also gives some very good insight, this is very good understanding ((Refer Time: 33:48)) little bit, yes please, from here to here this is a complex envelope. How do I get a real signal for the complex envelope, multiply the complex envelope by  $e^{j(\omega_c t + a(t))}$  and take the real part of that, that is what I have done, there is going from here to here. Now, you have quite experts in doing that in case we discussed this is what we get, now let us look at this expression in some detail.

We find that FM signal has, in fact it is quite clear that it is not a linear modulation at all, because we have terms like  $a^2(t)$ ,  $a^3(t)$  etcetera, which was otherwise obvious

that it is not a linear model, but this becomes more explicit now. So, what can you say about the bandwidth of the FM signal, such an FM signal it has this pure carrier component, there is a carrier component like we saw earlier.

From various this component due to this modulation, bandwidth of this will be plus minus b around the carrier. The bandwidth of this component will be plus minus 2 b around the carrier, the term corresponding to a to the power m t would have bandwidth of plus minus n b around the carrier. So, as you can see, since this is an infinite series theoretically for bandwidth which is infinite.

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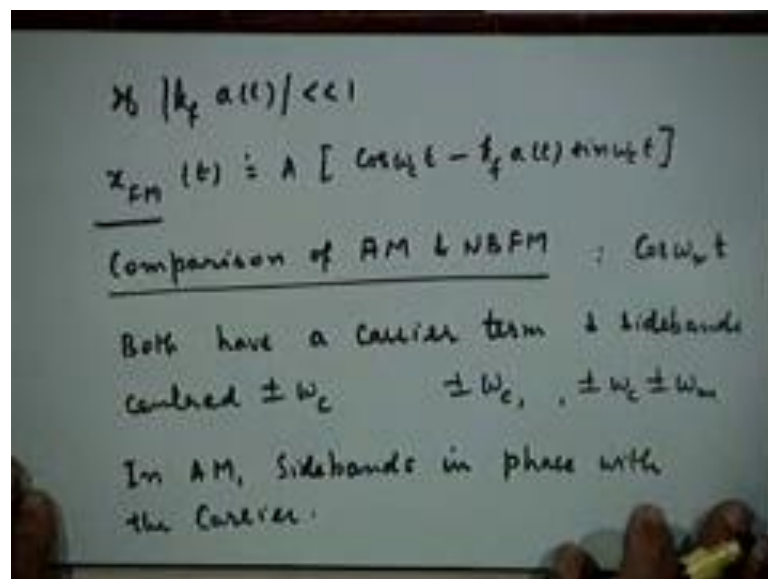
So, this once again shows that theoretical bandwidth is equal to infinity; the spectrum contains various components that are the components due to the unmodulated carrier that is the first term. Then, the spectra a t, a square t etcetera, translating around the carrier or transmitted to f c, these are the various components that you have, so that shows this got infinite bandwidth.

For the narrowband FM we know that we have, we can be approximated by only the first term in that expression then, but let me discuss it again. In particular let us assume that m t your message signal is a linear combination of two sinusoids; it is sum of two sinusoids. So, let m t be cosine sine omega m t plus A 2 cosine omega 2 t, if it was an AM signal or a linear modulation, you will get components after modulation at omega c plus minus omega 1 and omega c plus minus omega 2.

Now, in the case of FM you will get components can you tell me now what components you will get, so for AM the components will be  $\omega_c$ , basically  $\omega_c + \omega_m$  and  $\omega_c - \omega_m$  that transmitted to  $\omega_c + \omega_m$  and  $\omega_c - \omega_m$ . For the case of FM, now not only you will have  $\omega_c + \omega_m$  and  $\omega_c - \omega_m$ , but you will also have  $\omega_c$ , now look at I want the answer to come from the flow.

Look at the  $n$ th term of the expansion, if you remember when we discussed non-linear systems briefly, we discussed that it generates digital frequency components. For example if  $a(t)$  to the power  $m$  will produce that  $n\omega_1$  and  $n\omega_2$  will also produce their sum and differences. So, you will have the most general form of these various components of the  $\omega_c + n\omega_1 + m\omega_2$ , where  $n$  and  $m$  are suitable integers or they can be any set of integers. These are the intermodulation terms that the non-linear components produce, so the FM, therefore generates like any non-linear system intermodulation frequencies.

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However, if you go back to the series expansion and say that the assumption that  $k_f t$  is much less than 1, then once again we can approximate this series expansion by the first two terms. So, if the assumption that we normally make for narrowband FM is valid, so if  $k_f a(t)$  magnitude is less than 1, then  $x_{FM}(t)$  reduces to  $A \cos(\omega_c t - k_f a(t) \sin(\omega_m t))$

$k f a t$  into  $\sin \omega_c t$ . How will this be different for the case of phase modulated signals, simply this  $a t$  will get replaced with  $m t$ .

Let me finally spend some time, so in this I hope you get a different insight of the spectrum of the FM signal, this analysis does not help us to obtain the mathematical expression for the bandwidth, but that was not the objective of this discussion. Objective was simply to appreciate better, what will happen to the spectrum what will be the spectral components we will see, when the input signal contains more than one frequency, when the modulating signal contains more than one frequency.

We will not only get  $\omega_c \pm \omega_1$  and  $\omega_c \pm \omega_2$ . But also the so called inter modulation terms, that was the important point that we really wanted to appreciate from this. The last point in this connection that I want to discuss is the representation of FM signal in terms of the phasor diagram particularly for the narrowband FM. It is difficult to talk about the phasor diagram at this stage, we will come to that also, it is not really that difficult, but we will come to that at an appropriate time.

Right now let us look at, because it is interesting to see, because the FM and AM signal there will be narrowband, FM signal and the AM signal will have similar spectra. How do the phasor diagrams differ from each other and how do they physically generate different kinds of signals. They have similar spectra and yet they physically generate different some kind of signals looking in the time domain, they are looks in the time are different.

One is the constant amplitude signal whose frequency varies; the other is the amplitude varied signal even though the spectra is very similar. So, we will look at the comparison of FM, particularly narrowband FM. So, let me call it as AM and narrowband FM both of them have a carrier term, both have the term  $\cos \omega_c t$  ((Refer Time: 43:07)) and sidebands center at  $\pm \omega_c$  that is the common picture  $\omega_c \pm \omega_1$ .

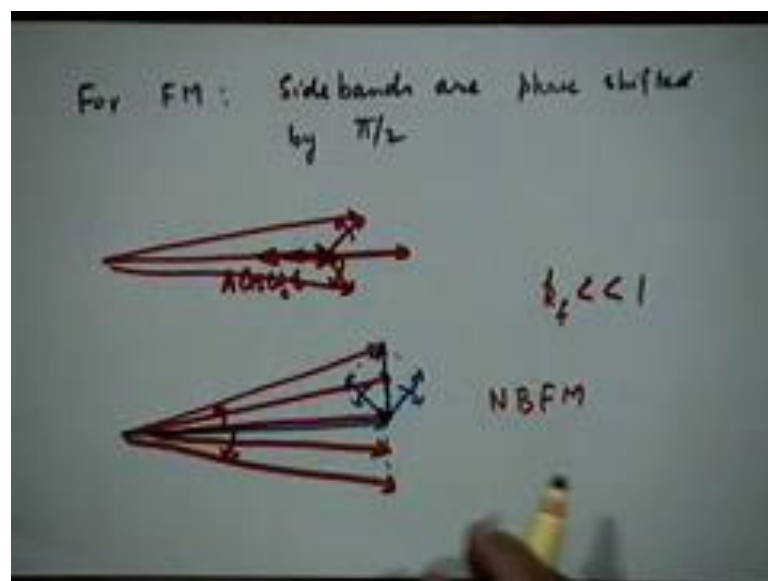
For this discussion when you talk about phasor diagram, typically talk of phasor diagram when the input signal is once again  $\cos \omega_m t$  or  $\sin \omega_m t$ , when you look at that situation. So, that means you have  $\pm \omega_c$  and you have  $\pm \omega_c \pm \omega_m$ , these are the components which are present in the AM

signal as well as the narrowband FM signal, this is the only frequency component which are present.

In what way the spectra are different, we need to appreciate that in the FM signal or let us talk about the AM signal first, in the AM signal the sidebands and the carrier face each other. If you remember the expansion you will have cosine  $\omega_c t + m t$  times cosine  $\omega_c t$  again that this these are not quadrature carriers, they are two carriers involved here and here they are in the same phase, whereas the two carriers involved here are in quadrature phase.

So, in FM spectrum or let us say in AM, sidebands in some sense. In the sense I have just mentioned here, that they are in phase with the carrier, whereas in the FM case they are in quadrature phase, they are in phase with respect to the carrier.

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You can think of this term as the side band producing term, this are the pure carrier term, side band producing term has a carrier which is in quadrature with the pure carrier term, basically that is what we are trying to say. So, sidebands are phase shifted by  $\pi$  by 2, let us see, what is the effect of that in the phasor diagram. If this point is clear it should not be very difficult.

Let us see the phasor diagram that we drew for the AM signal, remember we are talking about a rotating plane, rotating at a angular speed of  $\omega_c$  radianc specific, so that



the carrier phase looks like fix vector of length equal to the amplitude of the carrier. So, this represents your cosine  $\omega_c t$   $A \cos \omega_c t$  and the two sidebands are the components  $\omega_c + \omega_m$  and  $\omega_c - \omega_m$  and with respect to this fixed phasor, there will be moving phases, rotating phases.

One moving let us say anticlockwise, the other moving clockwise and the resultant phasor at any time is the resultant of all these three factors and were will this resultant will always lie, it will lie always along this line. As these phases rotate electrically around this phasor, the resultant of these two will always be on this line and therefore, the overall result will always go around the fixed amplitude of the carrier here.

The resultant phase will move along this line around this nominal amplitude that is the phasor diagram that we discussed earlier for the case of AM signal, how do the things change now? We have the same carrier as before, the only difference is that this rotating phases are now associated with the term sine  $\omega_c t$ . So, at  $p$  equal to 0 they are in quadrature with the carrier component, so they are rotation the clockwise or the anticlockwise rotations can be represented like this.

For these two rotating phases, where will the resultant lie, it will lie perpendicular to the phasor always, as they rotate the resultant of these two phases will be somewhere on this line, at this point it is here. If they here the resultant will be somewhere here, but this resultant will move like this and what will the overall resultant look like, so your overall resultant will have different positions like this, as these two phases rotate and this resultant phase are moves up and down.

The overall resultant moves in this region and what do you see now, if you assume that this amplitude is not very large this variation in amplitude which is visible here would not be very large, because these phases will be small. However, there will be a angular variation and this angular variation is what causes angle modulation and it also makes clear why that condition is required, if you good approximation to  $m \ll 1$ , there are to the FM signal.

The condition is required, because of this amplitude is small, and then the amplitude variation associated with this resultant phasor would be negligible. On the other hand, if this is not small there will be a considerable amplitude variation as well and there will be

distortion signal that will generate, will not be the pure FM signal, but FM signal with some amplitude variations as well.

Where also even this will not be a very good representation of the variation that we actually want to see from the FM signal, proportional to FM signal magnitude, so this is the phasor diagram of the narrowband FM signal. So, unlike the AM signal which the resultant is holding the amplitude, where the resultant is wearing the amplitude as well as phase. There will be only hope that the amplitude variations can be minimized by using the approximation that here  $k_f$  is much smaller than 1.

Not clear yes  $k_f$  is much smaller than 1,  $\beta$  is much smaller than 1, each of this will move in a circle and the resultant will keep on moving like this. So this you please keep in mind when we are discussing FM in some context, where we need to use the phasor diagrams representation. I wanted to start the discussion on the FM demodulation also today, but I think our time is up and we will start the discussion on FM demodulation next lecture.

Thank you very much.