Communication Engineering
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Lecture - 6
Fundamentals of Analog Signal Transmission

Our topic of discussion today now is, we are now getting into the mainstream of communication theory and the first thing that we like to understand is some basics about analog signal transmission. Then, to keep this discussion simple, we will use a base band model, we look at the base band signal propagation or transmission through a communication channel. Now, suppose we look at the block diagram of a base band communication signal, transmission system.

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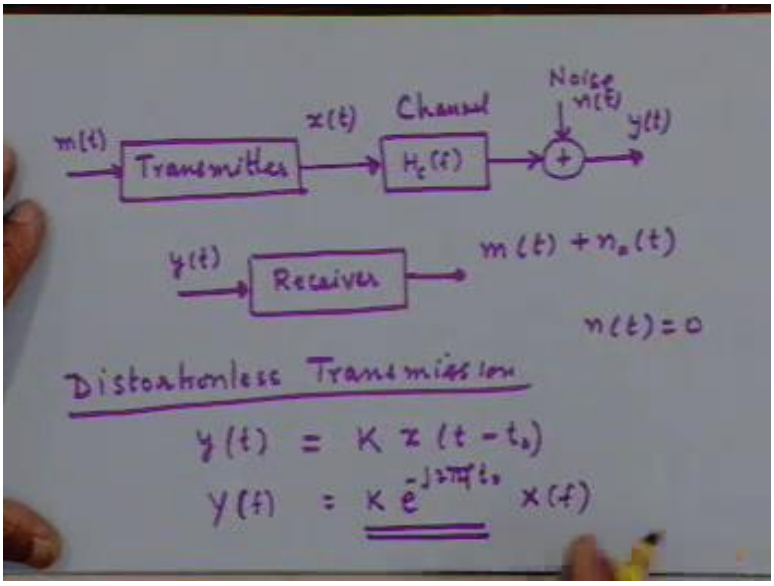

As we know, we must have a transmitter and the transmitter has a, this block has everything it that is required, input to this is the message signal m of t, I will denote the message signal, most of the time by the notation m sub t m of t. Output is output of the transmission transmitter is, a waveform or a signal x of t which typically in the base band context would be a suitable scaled version of m of t, appropriately amplified so as to drive the communication channel appropriate, nicely.

The channel here, in general can be modeled by two parts, a part which embodies the effect of distortion that it might introduce and that part can be modeled conveniently as some kind of a filter, some kind of a linear filter. What do you mean by that, because the channel can modify the amplitude of the various frequency components or it can modify the phase of the various frequency components. All these effects can be nicely modeled, if you think of the channel as a, filter of some kind.

So, very simple and convenient model for a channel, communication channel and that is true for every kind of communication channel, is to think of this as a filter, with some transfer function H sub c f, so this models if channel. The second part of the channel model, will consider the effects like noise, you know that, noise gets transmitted, noise gets added to the signal at various part, various points of the communication system; starting from the transmitter, going up to the receiver.

All these effects again will be clubbed into the channel and therefore, the channel will have a second part, which is denoted like this. So, this here this block here, is essentially an adder which adds to the output of the channel, some noise of an appropriate kind, which is suitable for use and that is what you receive at the receiver. So, the received signal, suppose you were to call it y of t, contains effect of transmission this through this channel.

This filtering that is affected by the channel, of some kind and some additional some noise and the receiver is supposed to work on this input y of t to produce, so this is your receiver. To produce replica of your transmitted message waveform m sub t, m of t together with, a small amount of noise which it cannot eliminate. So there is an input noise n of t and the receiver will try to eliminate as much of this noise as possible and also, but there may be some residual noise still left, which were denoting by n sub o t.

The receiver has to process the incoming waveform y of t, so as to remove the distortion that might have been affected by the channel, might have been introduced by the channel and also to eliminate as much noise as possible that is the job of the receiver, main job of the receiver. Now, so this is the model, this is a framework in which we have to, we will carry out the discussion for today, most of them today’s class. Now, our first you see, we have been talking about channel introducing two kinds of things.

One is this effect, the other is this effect, this essentially carries out what you say distortion, distortion of the signal. This is additive noise; this adds on to the signal, so first let us concentrate, in fact in today’s class. Let us first discuss only about the distortion part what we like to ideally have is, what we call distortion less transmission. And in this discussion we will ignore the presence of noise, let us assume that noise is not there at all.

So, from we will assume n t is equal to 0, because we want to concentrate on what kinds of distortion the channel can introduce and understand their nature and see whether we can do something about it, at the receiver, just to simplify the discussion. Now, in this case, your received signal y of t, which is basically the transmitted signal x t, convolve with the filter impulse response x c t, the filter which models the channel, is what you will get as y t.

Now, suppose you forget about the channel for a minute, suppose you would like, this signal here or that the input to the receiver, to be a replica of the transmitted signal without distortion. That is the ideal thing you would like to have, y of t should be replica of x of t without distortion. So what, how can we express this relationship in the most general form, what kind of effects we can tolerate and what kind of effects we cannot tolerate.

You can tolerate a scaling effect, if the signal gets attenuated, it does not matter, and that is not distortion, if it simply gets attenuated because I can compensate for attenuation very simple by, suitable amplification. The other thing that I can tolerate is, some delay that propagation delay. After all, the signal has to physically transmit physically propagate from one point to another point and no matter what is the form in which it propagates. So it will take a finite amount of time to do so and therefore, there will be some delay.

So, as long as the received signal, differs from the transmitted signal, only in terms of a scaling factor and a constant delay, that is acceptable to me, as a received signal, as a replica of the transmitted signal, you agree with that. So, ideally speaking by distortion less transmission, we mean that the received signal y of t, is some constant k times x t, which is delayed by some amount t sub 0. So, this is the condition for distortion less transmission.

If this happens, if your channel only introduces a scaling factor and a delay, it is an ideal channel, nothing could be better, when channel is a friendly channel and that is the kind of channel I am looking for, physically. Now, let us look at this effect of frequency domain, what are we saying therefore, what are the characteristics of the channel for a distortion less transmission. Look at the, what is the if you have to express the same relationship in frequency domain, this will be y of f equal to, k times e to the power minus j 2 pi f t naught into x of f. So, what are we saying about the transfer function x here, it is equal to k times e to the power minus j 2 p f t naught.

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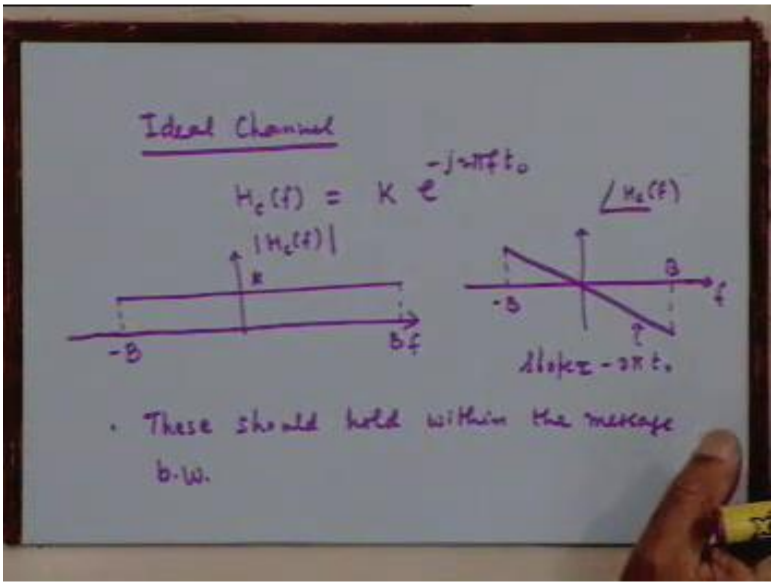

So, the ideal channel H c H sub c f, has a transfer function given by constant k into e to the power minus j 2 pi f t naught. If you were to characterize this ideal channel, in the terms of frequency domain plots, then what is the magnitude characteristic we are expecting from this. A constant k for all values of the frequency f, for all frequencies f and what are we expecting, from which in terms of case characteristics. Anyone, what will be the angle of H c f?

Student: ((Refer Time: 11:12))

What kind of characteristics we are expecting as a function of frequency?

Student: ((Refer Time: 11:18))

The exponent here.

Student: ((Refer Time: 11:21))

Whereas minus j f j to pi f t naught, as a function of frequency what kind of, it is a straight line, with a negative slope. The angle is0at 0, it is positive for positive frequency negative for positive frequencies and positive for negative frequencies and the slope will be equal to minus 2 pi t naught, where t naught is a delay that; that is a slope, slope of the straight line.

So, basically what are we saying, that for a signal to be transmitted without distortion, the ideal channel would have a flat magnitude transfer characteristics, magnitude transfer function and a phase function, which is linearly dependent on the frequency. So, that every frequency component present in the signal, undergoes the same amount of delay. Basically what we are saying is, every frequency component goes through the same amount of delay and for same amount of delay the phase shift, is a linear function of the frequency, required phase shift.

So, that is the ideal characteristic, of course, this kind of an ideal channel is too much to expect, to be available in practice. It is not even required, if you really look upon it because, in real terms you will be transmitting the signal of finite bandwidth, not of infinite bandwidth. So, in that case, our concept of an ideal channel can be made less stringent, in as much as these characteristics hold, within the bandwidth of the signal, we should be quite happy.

Because what happens outside is of no interest, is of academic interest, because the signal does not have any frequency components, beyond those values beyond some values. So, therefore we can make these conditions less stringent, by saying that these conditions should hold, within the message bandwidth. So let us say, if the message bandwidth is B, we would like that the transfer function magnitude is constant, equal to k between minus B to plus B.

Whether it is constant beyond that, is not of any interest to us. That may or may not be, because the signal does not have any frequency components there, so it would not bother us. Similarly, these linear characteristics should hold between minus B to plus B, after that it should become slightly non-linear or something else happens to it. It is not a straight line after this, it does not bother us very much, because the signal does not have any components in that frequency, is that clear? So, an ideal channel, in a practical situation would be one which has these kinds of characteristics within the message bandwidth at least, is that good, any questions?

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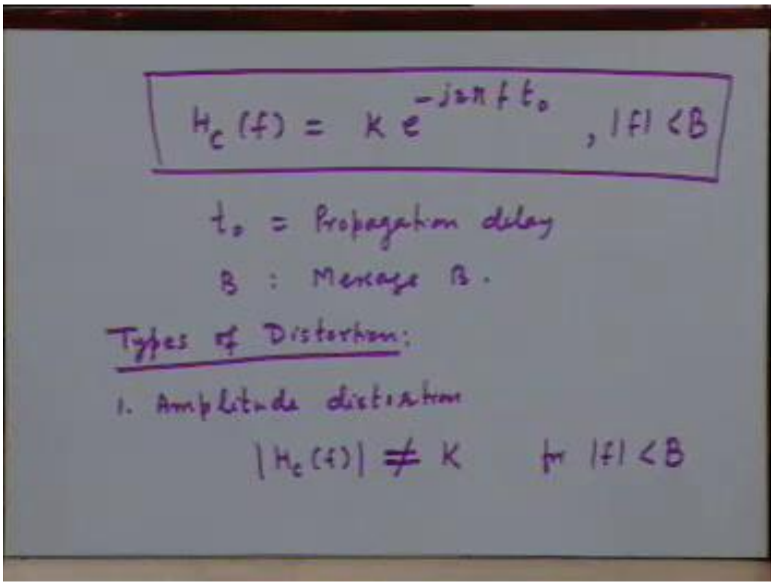

So, we call here, we could say then say that the ideal channel transfer function, instead of say qualifying unconditionally this being equal to minus j 2 pi f t 0, will say that this should be so far, mod of f less than B, that is our condition for distortion less transmission. A t sub 0 denotes the propagation delay and B here is the message bandwidth and k is the attenuation constant. Now this is what the ideal channel is supposed to do.

Real channels unfortunately do not in practice even meet these less stringent conditions, even within the bandwidth of interest, the magnitude transfer characteristic function is not necessarily a constant value, there variations and the phase characteristics are not necessarily, linear functions of frequency. So, when that happens; the received signal would or would not be a replica of x of t, it would not be a replica of x of t, because it will now go through a convolution of x of t with the impulse response h of t.

For the ideal case, what is the impulse response like; the impulse response is an impulse function. That is why whatever signal you transmit, it gets in the same form to at the receiver. But in this case it will not be so, if these conditions are not satisfied, so what kind of distortions can be introduce, so let us discuss types of distortion. Incidentally, these kinds of distortion that I am discussing here, we also refer to them as linear distortion.

Linear because they are arising from the non ideal characteristics of a linear filter, which is being used to model the channel. The ideal characteristics are, it should be a linear filter but besides linearity you want, the magnitude transfer function should be constant, the linear filter may also have magnitude transfer function which is not constant. Any filter usually will not have that, so when that is not so, then one kind of distortion is introduced.

If the phase characteristics are not linear with respect to frequency that leads to another kind of distortion, but both these kinds of distortion are linear distortions, because they are arising from non ideal characteristics of a linear filter, which is modeling the channel. So, there are two kinds of linear distortion, which are though linear distortion here, one is called the amplitude distortion and this arises, when this condition that the magnitude characteristic should be constant is not satisfied, over the bandwidth of interest.

If the magnitude characteristic of the channel is not constant, equal to some value k, which is which denotes the attenuation. Then we say that, over the bandwidth of interest we say that, the channel is introducing amplitude distortion. In the sense, that the different frequency components present in the message waveform m t, are being amplified or attenuated differently, by the channel and the ideal channel would treat all frequency components in the same way.

So, that to keep their relative magnitudes, the same as the receiver, but in this case certain frequency components may be attenuated less, certain other frequency components may be attenuated more and this relative difference in treatment, in terms of attenuation causes amplitude distortion. So, this amplitude distortion does not refer to, does not tell us what kind of effect takes place on the output waveform. This is a characterization of the frequency domain.

But in the frequency domain, the channel does not treat all frequency components, with the same attenuation characteristics, that what it means. So, do not think of any other connotation of amplitude distortion other than the one implied in the frequency domain. It does not tell us anything specific about what kind of waveform you might see, that should be very clear in your mind. For example, I will give you typical attenuation characteristics that you might see.

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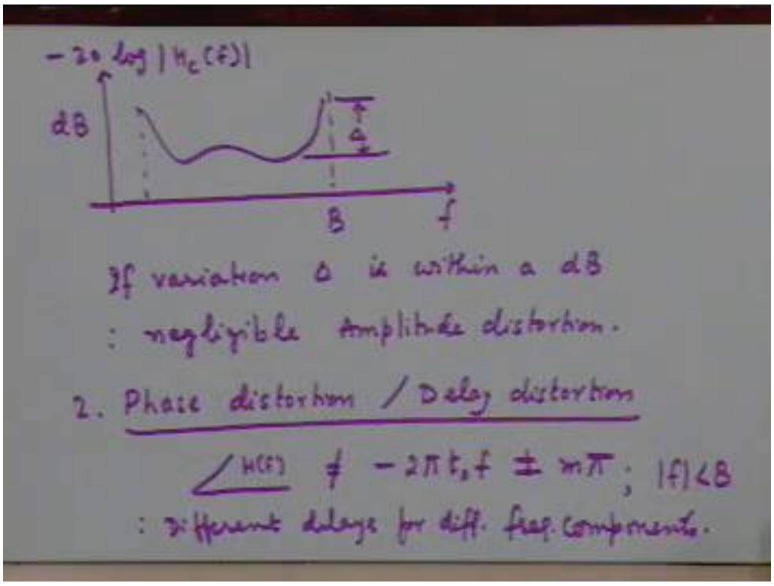

In the telephone channel for example, usually these characteristics, magnitude characteristics are plotted on a log scale, so I am plotting here minus 20 log of H c f, what is it call?

Student: Decibels.

Decibels, so, you are plotting the amplitude response in decibels, so the units are decibels, so a typical characteristic that you might see, may look something like this. So, you can see that there is a lot of variation, let us call this delta, in the amplitude or the at amount of attenuation difference between frequency components, may go through. I am plotting here only for positive frequencies, you can replicate for the negative frequency axis and for a, so we are going to see characteristics like this, which are not flat, which is ((Refer Time: 21:42)) what you ideally like to see.

As long as this delta is small overall variation is small, you can ignore it, it does not matter, and effectively it does not matter very much, what is the extent to which you can tolerate it. When this variation is within a dB or so, within one decibel or so, that is a rule of thumb for you to remember, so if the variation of the attenuation as a function of frequency. Let us call that delta is within a dB, within one decibel, the amplitude distortion is negligible, and then we can ignore it.

But if it is more than a dB, it becomes significant and we have to take that into account, so that is, what amplitude distortion is all about. As against this, we have the second kind of linear distortion which we call by the name of phase distortion or sometimes also called delay distortion. These two things typically mean the same thing and as the name implies, this kind of distortion arises when your channel has phase transfer function which differs from the required ideal transfer function.

What is the required ideal phase transfer function?

Student: ((Refer Time: 23:34))

A linear function of frequency passing through the origin, odd function it has to be of any real feature will have the phase function as an odd function, that is not an issue. So, if it differs from that linear characteristics within the bandwidth of interest, we get the phase distortion, so if angle of H f is not equal to, minus 2 pi t naught f. Actually, it would not matter, if it is this plus minus tell me something, plus minus how much?

Student: multiple of pi.

A multiple of pi, is it not, if it is a multiple if I, if you go back to this equation that we were looking at.

Student: ((Refer Time: 24:27))

If I put an e power minus j m pi or e power plus j m pi where m is an integer, this will be a constant value equal to either plus 1 or minus 1, which could be absorb in the attenuation factor. So, it does not really, make any difference to the basic phase characteristics, so the required condition therefore, really speaking is, this plus minus m pi.

So, if this is so, ((Refer Time: 25:05)) then what is, over the bandwidth of interest, for f mod of f less than b, if this is not so, that is that causes phase distortion or delay distortion, if this is so, what is the amount of delay introduced at frequency f, t naught and it is the same for all frequencies. If this is not so, basically what it implies it implies is, that different frequency components undergo, different amounts of delay and that is why when they combine together all these frequency components the other end, the output, the signal does not appear to be the same as we started with.

So, if this linear characteristic does not hold, it implies different delays for different frequency components and that is why, we also call it delay distortion. If every frequency component present in the message signal undergoes a same amount of delay, there is no delay distortion, no phase distortion. If different frequency components undergo different amounts of delay, there is a delay distortion. Now fortunately, in analog signal transmission like speech particularly speech, delay distortion is not of much consequence.

Reason is very simple, our ear is insensitive to delay distortion, this happens to a property of our perception, hearing, but for speech, this is the exceptional situation. For pictures for example, that is not true, delay distortion matters, because the eyes are not insensitive to, the way I see a picture is not insensitive to phase information or delay information and similarly for data, delay distortion has can cause havoc.

Student: ((Refer Time: 27:19))

I will repeat that.

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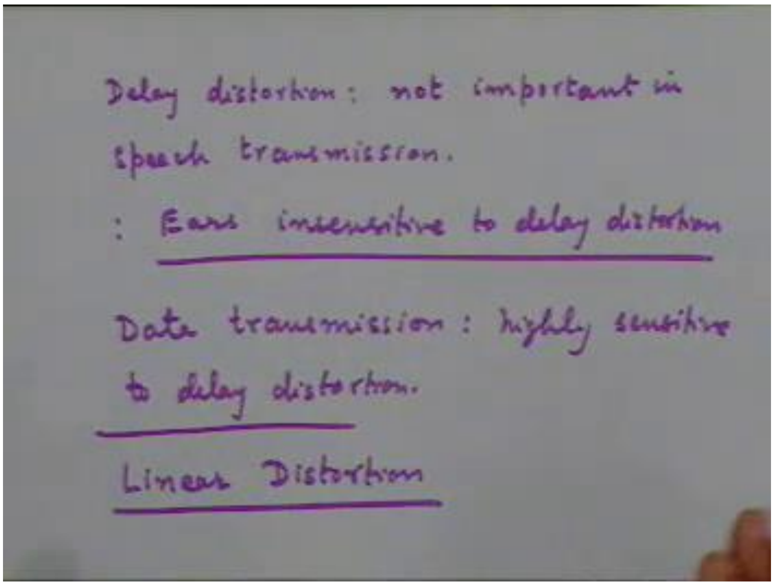

So, this is just a practical observation that I am giving here, that delay distortion, not very important in speech transmission, when your message signal is basically your speech signal. What we are saying is, if the message signal happens to be speech signal and if the channel introduces delay distortion, we are not too much worried about it. And the reason is, because ears are insensitive to this kind of distortion, so to understand, why that is so of course you have to go into, how the ear perceives the signal.

We do not have, we are not going to go that, into that now, and that is just a matter of information for you. On the other hand, delay distortion becomes fatal if it is present, when you are doing a particular data transmission. So data transmission is highly sensitive to delay distortion, similarly for video, similarly for pictures, it is highly sensitive to the delay distortion. So, I will repeat ((Refer Time: 29:00)) also, in any kind of pulse transmission, which data transmission is a special case of that, delay distortion is going to be fatal.

It causes a very severe kind of distortion and if you do not compensate for if you, if you do not take that into account, in designing your receiver, you will not be able to do a good digital communication. So, this these are two kinds of distortion, which fall within the domain of what I have mention as linear distortion, which whether we have amplitude distortion or phase distortion or both, we are talking about linear distortion

One important characteristics of linear distortion is, that basically the existing frequency components the message signal, will get treated differently by the channel, either in terms of their attenuation characteristics are different frequencies or in terms of, delay characteristics have different frequencies, so that is linear distortion. One nice thing about the linear distortion is, that at least in theory, in concept it is very easy to compensate for it, is it not, it is obvious.

What should we do at the receiver?

Student: put a filter ((Refer Time: 30:28))

Put a filter, which is the reciprocal of the channel transfer function.

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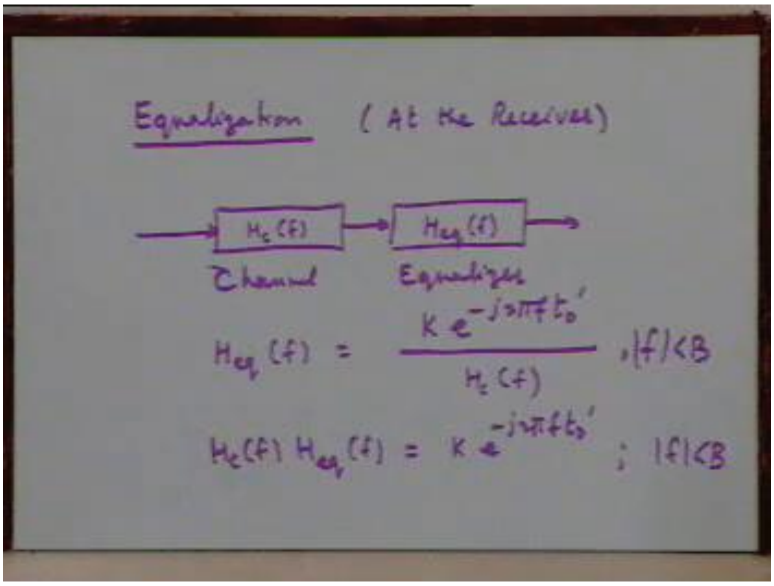

So, in theory it is possible to remove the distortion through a process called equalization and that of course, will have to be done at the receiver. So, what we are saying is, the message signal goes through a channel, that the transfer function H c f, sowhat we must do at the receiver, have a filter with transfer function let us say H, sub e q f, where the product of these two, together appear as a distortion less channel.

So, if this is an actual channel and this is your equalizer what we like to have is, that the equalizer should have this kind of transfer characteristic, so that the product of these two, appears like an ideal channel. The product of these two are net transfer function that the signal will see, the message will see, that should be equal to k times e to the power minus j 2 pi f t naught, prime let us say, some other value of propagation delay. Of course, this is required only for mod of f less than the bandwidth.

So, that the product, the you can think of the product as some kind of an equivalent channel, satisfies the ideal characteristics, so in theory it is possible to take care of linear distortion perfectly. Of course, things are not as simple as they appear here, theory is perfectly fine here, but in practice you can imagine, there will be lot of difficulties in actually implementing such an equalizer. Can you think of some difficulties, there are two major difficulties that will come up?

One is, we assuming, that we know what channel transfer function is high, in reality you will very rarely know anything about what the channel is doing, you are transmitting the message. What you see at the output of the channel, is the received or the distorted message, you know nothing about the channel. So, unless you do something special, unless you make a special effort to learn about the channel characteristics, there is no question of trying to equalize for it.

So, that of course means, more complication, to learn the channel characteristics, sothat you can implement an equalizer filter, which is of this kind is a non trivial effort, non trivial job. The second problem, that may arise, even if you knew the channel transfer is, suppose you knew the channel transfer function, is due to the fact that we have ignored another very important component which is present in the channel, noise. Suppose the channel H c f has a null or in the range of some frequency components, within the pass band, within the message bandwidth of interest.

For a few frequencies or in the neighborhood of some frequencies, the channel exhibits very deep attenuation, very large attenuation, suppose it happens it can happen. Then, what will be the required equalizer characteristics of those frequencies, very large amplification of those frequencies and if you implement that, you can do that. What you are also going to do is, amplify the corresponding noise part, noise components there. So, you may undo the effect of distortion, but you may be introducing or enhancing the effect of noise, which may not be desirable.

So, in theory this is all fine, but in practice one has to work out a solution, which takes care of these concerns. We will not go into further details of this moment; this is a subject which is properly dealt with in a course in digital communication.

Student: Transfer function which are always constant and ((Refer Time: 25:02)) will they vary with time.

We are assuming in this, that is a good question, the question is, the channel can we assume the channel to be having a constant transfer function for all time or in other words, is it proper to think of the channel as a linear time invariant system. The question can be rephrased like this, because I talk about linearity, but I did not talk about time invariance. It is a good question; the answer to this question is yes, in some channels, no in some other channels.

The discussion that we are doing now is for those channels where it can be modeled as a LTI filter. There can be channels, in which it cannot be modeled as it can be modeled as a linear filter, but not necessarily time invariant. For example, a wireless channel in which there are lots of reflections on various objects in the propagation path and these object keep moving or you keep moving. Obviously, it is a linear characteristic, but it is not time invariant, so linear time variant characteristics.

Then of course, our discussion has to, this equalization business becomes even more difficult, because not only you do not know, it keeps varying with time. So, how to handle it both theoretically as well as practically becomes a very major issue, but these are issues which are beyond the scope of this course. At the moment we will not discuss this, but at least you know, that it can be done and what the concerns are. Any other questions?

Student: ((Refer Time: 36:34))

So, that is as far as our discussion on linear transform linear distortion is concerned, now I am I have been emphasizing this concept, I mean this adjective linear.

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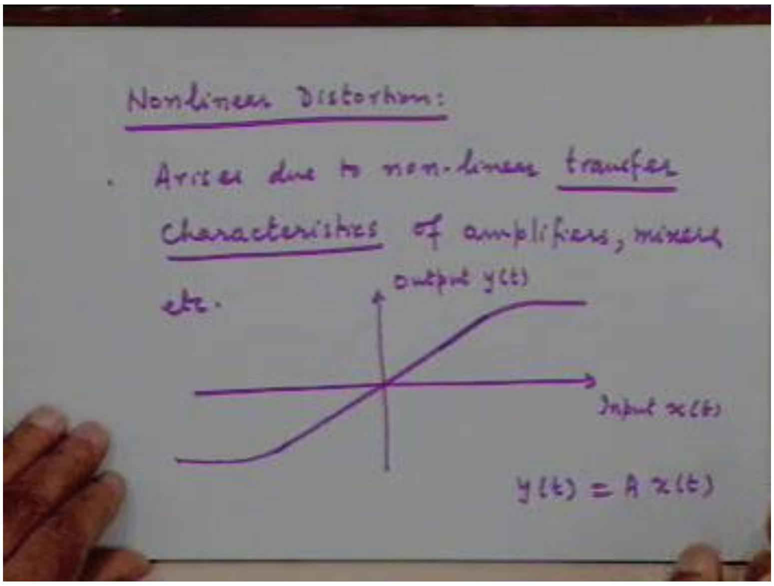

And that is because, we can also have a kind of non-linear distortion and as the name obviously implies, this kind of distortion would arise in a communication system, in which, in some part of the communication system or the other, there is some kind of linearity. Can you think of any kind of non-linearities which can cope up in a communication system, which are the components which can give rise to some non-linearity?

Student: amplifiers.

Amplifiers, very correctly said, particularly power amplifiers, not the voltage or a current amplifiers, but power amplifiers. You have done some course on power amplification, some discussion on power amplification class B, class C amplifiers. And these are dependent on, let us say let us take one of the these power amplifiers, whatever knowledge you have is sufficient for discussion, under what conditions do they work most efficiently or very well?

Student: ((Refer Time: 38:10))

They can they are supposed to work very well; they work very well, if your amplitude in the input signal remains constant. If the amplitude fluctuates a lot, first of all they do not work very efficiently, but that is not our concern here, efficiency is not what is what we are considering here. We are more concerned about the fact, that if the amplifier gives different amounts of amplification, depending on the amplitude of the input signal. It ((Refer Time: 38:44)) reaches the low amplitude signals in some way and the high amplitude signals in some other way, then we have a kind of non-linearity.

So, if you were to typically, so let me first mention what non-linear distortion is, so this arises due to non-linear. Now, this is, please note what I am writing, non-linear transfer not transfer function, because non-linear systems, I cannot study in terms of transfer functions, I cannot study the sense I mean the, we are not talking about frequency domain here. I am saying due to non-linear transfer characteristics, what are transfer characteristics of a device, like an amplifier.

How do you show the transfer characteristics, these are the devices which can exhibit non-linearity. Amplifiers, mixers etcetera, we still have to talk about mixers, we will talk about them later, let us talk about amplifiers, so when I say transfer characteristics of an amplifier if we were to plot it, what is the plot that I make?

Student: ((Refer Time: 40:03))

I simply plot output versus input, output y against input x and we are here referring to the amplitude of the output versus amplitude of the input. So, in fact ignore the time variable here that is of no consequence, what will be the output input characteristics or transfer characteristics of an ideal amplifier?

Student: ((Refer Time: 40:31))

Straight line is it not, input is small output should be small; correspondingly, if the input is large, the corresponding output should be large simple, nothing beyond that. So, it is a constant times the input, y of t should be equal to, ideally y of t should be equal to, some A times x t, where A is the amplification factor, that is for characteristics. Now, practical power amplifiers will exhibit a kind of non-linearity, which is very commonly encountered called the saturation non-linearity.

You know the effect, it is basically arises because you have some finite value of the power supplies and because of this, as your input signal becomes larger and larger in amplitude, instead of being linear like this, you get into this non-linear or situation mode like this. It does not remain linear for all values of the input amplitudes. So as long as your input signal is within the linear range, you are fine, but if your input signal goes beyond the linear range, then the output is no longer proportional to the input, y of t is no longer A times x t and that is a non-linearity.

Now, basically if you were to model this non-linearity, you can see that you cannot model it as a straight line, a straight line kind of a relationship. A linear kind of relationship of this kind, you have to introduce, can you suggest some model for this curve that you have here, some suitable parametric model?

Student: ((Refer Time: 42:14))

For example, we can possibly model it as a polynomial of some kind, we can find out suitable coefficients of a polynomial, such that the resulting curve would look like these characteristics. And that is the usual thing that is done to model non-linear components, non-linear devices, and use polynomial models.

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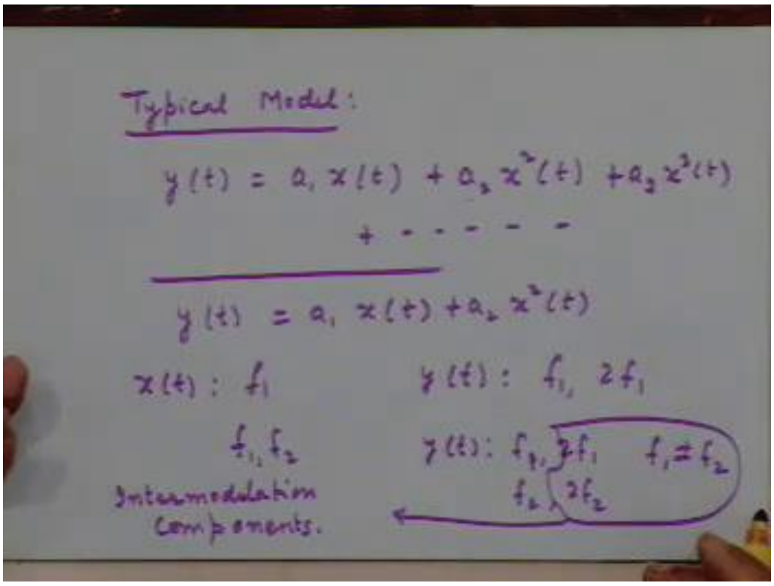

So, a typical model for example, or such non-linear things, would be that y of t is a 1 times x of t, which would have been the ideal thing. But you also have additional terms like a 2 times x square t, plus a 3 times x cube t and so and so further and theoretically it is possible to find a set of coefficients a 1, a 2, a 3 etcetera. With the polynomial of some degree suitable degree, such that this resulting characteristics, resemble the actual curve that we have plotted, y versus x curve that we have plotted.

Suppose, that is the case, suppose let us take very simple suppose we take a very simple non-linearity of the kind a 1 x t plus a 2 x square t. One thing is clear, that the output is not a replica of x t and therefore, there is a distortion, because we would like the output to be simply proportional to x t, but we have additional components like x square x cube and so and so further. So, therefore there is a kind of distortion here, but how does this distortion differ from the linear distortion, that we have discussed.

In one very important and very significant way, if you remember, I had made a comment more when we talked about linear distortion. That linear distortion, which arises due to non-ideal characteristics of this channel, in terms of it is transfer function, can only do something good or something bad relatively to different frequency components that exist in the message signal. So, but it cannot create additional frequency components, is it not?

A linear filter essentially affects whatever frequency components are present in the message signal. It will amplify them differently, attenuate them differently or delay them differently, that is all it can do, that is the linear distortion. Non-linear distortion on the other hand, has a potential to generate a frequency components in the output message signal, which were not even present in the input signal and that is the very important way in which non-linear distortion differs from the linear distortion.

Let me give an example, suppose the input signal can is a pure sine wave, just for the sake of appreciating this point, pure sine wave of frequency, let us say f 1. Now, what are the components present here, this will be a 1 cosine 2 pi f 1 t plus a 2 cosine 2 pi f 2 t whole square cosine square. So, cosine square you can write as, 1 plus cosine 4 pi f 1 t, sonow what are the components present in the input in the y of t, they are f 1 and 2 f 1, 2f 1 was not present.

Input was containing only f 1, so the input signal contains, let us say a frequency component f 1, output contains f 1 as well as 2 f 1, this is a very simple situation. Suppose, now the input signal contains two components f 1 and f 2, what will you have now, we will obviously have f 2, sorry f 1, 2 f 1 you will also have f 2, 2 f 2. In addition you will have f 1 plus minus f 2, you can see that. It is a matter of just, writing down the trigonometry and seeing that you will get terms, which is cosine of 2 pi into f 1 plus f 2 times t and 2 pi f 1 minus f 2 times t.

So, these additional frequency components that are getting generated is the real problem in non-linear distortion and that, we call these components we call, inter modulation components and this non-linear distortion, is for this reason also called harmonic distortion or inter modulation distortion. Because different frequency components modulate with each other, modulate each other to produce additional frequency components. So, therefore another name for non-linear distortion is, rather set of names in fact.

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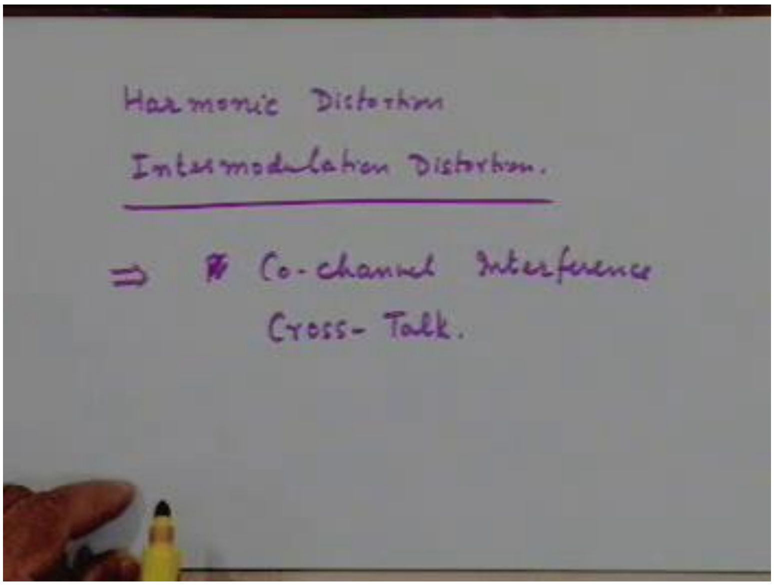

Harmonic distortion, because additional frequency components are being present or inter modulation distortion and that is, is that good or is that bad, that is very bad and why is it bad, can you appreciate?

Student: ((Refer Time: 48:10))

Can you suggest?

Student: ((Refer Time: 48:12))

Power is an issue, but it is less of an issue, it is creating components outside, suppose you have a signal of bandwidth b, your distorted signal will have components much beyond b and obviously, you are going to speak into somebody else is band. Suppose, you are allotted a certain frequency band to work with, you are transmitting outside that band; your received signal contains components outside the band. If your receiver was now trying to look for a signal in that frequency band, they will not only see that, but will also see, some components from this and this will cause a kind of cross talk. So, this will call what is called radio, it is also called co-channel interference or sometimes also simply called cross talk.

Student: ((Refer Time: 49:13))

What?

Student: ((Refer Time: 49:14))

Yes please.

Student: So, if it is highly dependent, what will be the fundamental frequency at that time ((Refer Time: 49:21))

Yes, because you can see that these characteristics, ((Refer Time: 49:29)) that will not depend on the value of these coefficients. So, how the power gets divided into different frequency components, will depend on the values of these coefficients in the model of non-linearity.

Any other questions?

Student: If no frequency which we got, so that will create problem only if they are in the order of the original frequency which we got, suppose ((Refer Time: 49:53)) actually I am taking left out second order, when the frequencies are too much high compared to this one, whether it ((Refer Time: 49:58))

What you are saying is, if f 1 and f 2 f 1 plus f 2.

Student: Comparing to f 1 and f 2, only there are ((Refer Time: 50:05))

The point is, you are not only going to, see you are definitely going to create a distortion for yourself that is obviously happening. So, within the frequency band, you have some frequency components and you are having, you are generating additional frequency components. These additional frequency components may lie within the bandwidth of interest or they may lie, outside the bandwidth of interest, they will typically have both kind of situations, f 1 minus f 2 will typically lie, within and f 1plus f 2 will lie outside.

So, both kinds of things are bad, f 1 minus f 2 creates distortion for us, f1 plus f 2 creates distortion for somebody else, cross talk for somebody else and soon and so further, we will stop here and let us ((Refer Time: 50:54)).

Thank you very much.