Fundamentals of Analog Signal transmission

[01:03] 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 right.

Um 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.

As we know we must have a transmitter and the transmitter has a this block has everything in it that’s required.

Input to this is the message signal $m(t)$.

[02:01] I will denote the message signal most of the time by the notation $m(t)$. Output is output of the transmitter is a waveform or a signal $x(t)$ which typically um in the base band context would be a suitable scaled version of $m(t)$ right.

 Appropriately amplified so as to drive the communication channel appropriately().

Nicely the channel here in general can be modeled by two parts.

A part which embodies the effect of distortion that it might introduce right 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?

You can modify the amplitude of the various frequency components or can modify the phase of the various frequency components right.

[03:04] All these effects can be nicely modeled if you think of the channel as a filter of some kind.

It’s a very simple and convenient model for a channel communication channel and that’s true for every kind of communication channel is to think of this as a filter with some transfer function $h(c, f)$.

So this models the 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 points of the communication system starting from the transmitter going right up to the receiver right.

All these effects again will be clubbed into the channel and therefore the channel will have a second part which is denoted like this right.
so this here this lock 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’s what you receive at the receiver right
so the received signal suppose you were to call it y of t right contains effect of transmission through this channel
this filtering that is affected by the channel of some kind and 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 some residual noise still left which we are 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 right
that’s the job of the receiver
main job of the receiver okay
now this is a model this is a framework in which we will carry out the discussion for today most of them in this 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 right
this adds on to the signal
so right first let’s concentrate in fact in today’s class let’s first discuss only about the distortion part right
what we like to ideally have is what we call distortionless transmission and in this discussion we will ignore the presence the noise right
let’s assume that noise is not there at all
so from we will assume n t is equal to zero 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 right
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 right
now suppose you forget about the channel for a minute
suppose you would like this signal here or the input to the receiver to be a replica of the transmitted signal without distortion right
that’s the ideal thing you would like to have
y of t should be replica of x of t without distortion
so how can we express this relationship in the most general form

[08:00] 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 doesn’t matter
that is not distortion right
if it simply gets attenuated because i can compensate for attenuation very simply by
suitable amplification right
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 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 that’s acceptable to me as a received signal as a replica of
the transmitted signal
you agree with that
so ideally speaking by distortionless 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 zero right

[09:13] so this out condition for distortionless transmission
if this happens if you channel only introduces a scaling factor and a delay it’s an ideal
channel
nothing could be better
when channel is a friendly channel and that’s the kind of channel i am looking for
physically right
now let’s look at this effect of frequency domain
what are we saying therefore
what are the characteristics of the channel for a distortionless transmission
look at the what is the um 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 two pi f t naught into x
of t right

[10:04] so what are we saying about the transfer function x here
it’s equal to k times e to the power minus j two p f t naught
(Refer slide time [10:11])
so the ideal channel $h_{cf}$ has a transfer function given by a constant $k$ into $e$ to the power $-j2\pi ft_0$ right
if you were to characterize this ideal channel in 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 $f$ right and what are we expecting from which in terms of phase characteristics anyone

[11:09] what will be the angle of $h_{cf}$ what kind of characteristics we are expecting as a function of frequency the exponent here which is $-j2\pi ft_0$ as a function of frequency why kind of it’s a straight line with a negative slope right the angle is zero at zero it’s positive for positive frequency, um negative for positive frequencies and positive for negative frequencies right and the slope will be equal to $-2\pi t_0$ where $t_0$ is a delay that that’s a slope slope of the straight line right

[12:10] 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 right 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 phased shift is a linear function of a frequency right required phase shift right so that’s an ideal characteristic of course this kind of an ideal channel is too much to expect to be available in practice right

[13:00] um it’s 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 right so therefore we can make this conditions less stringent by saying that these conditions should hold within the message bandwidth right

[14:00] so let us say if the message bandwidth is b right we would like that the transfer function magnitude is constant equal to k between minus b to plus b whether it’s constant beyond that is not of any interest to us right that may or may not be because the signal does not have any frequency components they are so it won’t bother us similarly these linear characteristics should hold between minus b to plus b after that it becomes slightly non linear or something else happens to it it’s not a straight line after this it doesn’t bother us very much because the signal does not have any components in that so 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 it okay good (Refer slide time [15:03])
[15:04] any questions
so we call here
we could then say that the ideal channel transfer function instead of qualifying
unconditionally this being equal to minus j two pi f t zero will say that this should be so
far mod of f less than b
that’s our condition for distortionless transmission right
a t sub zero denotes the propagation delay and b here is the message bandwidth and k is
the attenuation constant right
now this is what the ideal channel is supposed to do

[16:00] real channels unfortunately do not in practice even meet these less stringent
conditions right
even within the bandwidth of interest the magnitude transfer characteristic() function is
not necessarily a constant value
there are variations and the phase characteristics are not necessarily linear functions of
frequency right
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 right
that’s why whatever signal you transmit it gets in the same form as the receiver right
okay but in this case it will not be so if these conditions are not satisfied

[17:02] so what kind of distortions can be introduced
so let’s discuss types of distortion
incidentally these kinds of distortion that i am discussing here we also refer to them as linear distortion right
um linear because they are arising from the non ideal characteristics of a linear filter right
which is which is being used to model the channel right
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 right
any filter usually will not have that right
so when that is not so then one kind of distortion is introduced

[18:00] if the phase characteristics are not linear with respect to frequency that leads to another kind of distortion but both these kinds of distortions are linear distortions because they are arising from non ideal characteristics of a linear filter which is modeling the channel okay
so there are two kinds of linear distortion
we are talking about 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 okay
if the magnitude characteristic of the channel are 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

[19:06] in the sense that the different frequency components present in the message waveform m t are being amplified or attenuated differently by the channel
(Refer slide time [19:15])
an ideal channel would treat all frequency components in the same way right so that to keep their relative magnitudes the same as the receiver right 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 right so this amplitude distortion does not refer to it does not tell us what kind of effect takes place on the output waveform right 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

[20:02] that’s what it means so don’t think of any other connotation of amplitude distortion other than the one implied in the frequency domain okay it doesn’t tell us any thing specific about what kind of waveform you might see right that should be very clear in your mind for example i will give you typical attenuation characteristics that you might see in the telephone channel for example usually these characteristics magnitude characteristics are plotted on a log scale so i am plotting here minus twenty log of h c f what is it called decibels right so you are plotting the amplitude response in decibels so the units are decibels

[21:00] so a typical characteristic that you might see may look something like this right so you can see that there is a lot of variation let’s call this delta in the amplitude or amount of attenuation difference between frequency components may go through right i am plotting here only for positive frequencies you can replicate for the negative frequency axis okay and for a so we are going to see characteristics like this which are not flat right which is what you ideally like to see as long as this delta is small overall variation is small you can ignore it it doesn’t matter effectively it doesn’t matter very much right what is the extent to which you can tolerate it when this variation is within a d b or so within one decibel or so right

[22:02] that’s a rule of thumb of you to remember right if the variation of the attenuation as a function of frequency let’s call that delta is within a d b within one decibel the amplitude distortion is negligible then we can ignore it but if it’s more than a d b it becomes significant then we have to take that into account okay so that’s 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?
a linear function of frequency passing through the origin right odd function it has to be any real function will have the phase function as an odd function that’s 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 um angle of h f is not equal to minus two pi t naught f actually it won’t matter if it’s this plus minus tell me something

plus minus how much

a multiple of pi

isn’t it

if it’s a multiple if i if you go back to this equation that we were looking at right 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 one or minus one which couldn’t be absorbed in the attenuation factor right

(Refer slide time [24:22])

so it doesn’t really make any difference to the basic phase characteristics

so the required condition therefore really speaking is this plus minus m pi right

[25:03] so if this is so this 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 right if this is so what is the amount of delay introduced at frequency f t naught right and it’s the same for all frequencies right

if this is not so basically what it implies implies is that different frequency components undergo different amounts of delay and that’s why when they combine together all these
frequency components the other end the output signal does not appear to be the same as we started with right
so if this linear characteristic do not hold it implies different delays for different frequency components and that’s why we also call it delay distortion right

[26:15] 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 delays there is a delay distortion
now fortunately in analog signal transmission like speech right particularly speech delay distortion is not of much consequence
reason reason is simple
our ear is insensitive to delay distortion right
this happens to a property of our perception hearing right but for speech this is an exceptional situation
for pictures for example that’s not true right

[27:01] delay distortion matters because the eyes are not insensitive to the way i see a picture is not insensitivity to phase information or delay information and similarly for data delay distortion has can cause havoc
i will repeat that
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 theory distortion we are not too much worried about it and the reason is um because ears are insensitive to this kind of distortion

[28:12] so to understand why that is so of course you have to go into how the ear perceives the signal right
we don’t have we are going to that into that right now
that’s just a matter of information for you
on the other hand delay distortion becomes um fatal if it’s present when you are doing a particular data transmission right
so data transmission is highly sensitive to delay distortion
similarly for video
similarly for pictures
it’s highly sensitive to delay distortion all right

[29:00] so i will repeat “ “ also
in any kind of pulse transmission right which data transmission is a special case of that delay distortion is going to be fatal right
it causes a very severe kind of distortion and if you don’t compensate or if you don’t take that thing into account in designing your receiver you will not be able to do a good digital communication right
so these are two kinds of distortion which fall within the domain of what I have mentioned as linear distortion
whether we have amplitude distortion or phase distortion or both we are talking about linear distortion
one important characteristic of linear distortion is that basically the existing frequency
components the message signal will get treated differently by the channel
(Refer slide time [29:45])

![Image](image.png)

either in terms of their attenuation characteristics are different frequencies or in terms of
delay characteristics have different frequencies right
so that’s linear distortion

[30:12] one nice thing about linear distortion is that at least in theory in concept in
concept it’s very easy to compensate for it
isn’t it
it’s obvious
what should be done at the receiver
put a filter which is the reciprocal of the channel transfer function
so in theory it’s possible to remove any distortion through a process called equalization
right and that of course will have to be done at the receiver right
so what we are saying is the message signal goes through a channel that the transfer
function h c f
so what we must do at the receiver
have a filter with transfer function let’s say h sub e q f where the product of these two
together appear as a distortionless channel right

[31:15] so if this is an actual channel and this is you equalizer what we like to have is that
the equalizer should have this kind of transfer characteristic right
so that the product of these two appears like an ideal channel right
the product of these two is the 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 two pi f t naught prime let’s say some other value of propagation
of course this is required only for mod of f less than the bandwidth so that the product you can think of the product as some kind of an equivalent channel right satisfies the ideal characteristics
so in theory it’s 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 not of difficulties in actually implementing such an equalizer

[32:41] can you think of some difficulties um
there are two major difficulties that will come up
one is we are assuming that we know what channel transfer function is high
in reality you will very rarely know anything about what the channel is doing right

[33:02] you are transmitting a 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 right there is no question of trying to equalize further
so that of course means more complication
to learn the channel characteristics so that 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 um is due to the fact that we have ignored another very important component which is present in the channel noise right
suppose the channel h c f has a null or in the range of some frequency components within the passband within the message bandwidth of interest

[34:03] 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 right
then what will be the required um 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 right
so you may undo the effect of distortion but you may be introducing or enhancing the effect of noise right which may not be desirable
so in theory this is all fine but in practice one has to work on a solution which takes care of these concerns okay
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
this (student speaking)
we are assuming in this-, that’s 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 right because i talked about linearity but i didn’t talk about time invariance
it’s a good question
the answer to this question is yes in some channels no in some other channels
the discussion that we are right now is for those channels where it can be modeled as a l t i filter right
there can be channels in which it cannot be modeled as it can be modeled as a linear filter but not necessarily time invariant right
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 right
obviously it’s a linear characteristic but it’s not time invariant

[36:03] it’s linear time varying characteristics
then of course our discussion has to this equalization business becomes even more difficult because not only you don’t know it keeps varying with time right
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
(Refer slide time [36:20])

but at least you know that it can be done and what the concerns are okay
any other questions
so that is as far as out discussion on linear-, linear distortion is concerned
now i have been emphasizing this concept i mean this adjective linear right and that’s because we can also have a kind of non linear distortion and as the same obviously implies this kind of distortion would arise in the in a communication system in which in some part of the communication system or the other there is some kind of linearity okay

[37:28] can you think of any kind of non linearities which can crop up in a communication system which are the components which can give rise to some non linearity amplifiers very correctly said particularly power amplifiers right not the voltage current amplifiers but power amplifiers you have done some course on power amplification some discussion on power amplification class b class c amplifiers right and these are dependent on let’s say let’s take one of the these power amplifiers whatever knowledge you have is sufficient for discussion

[38:04] under what conditions do they work most efficiently or very well um they can they are supposed to work very well they work very well if your amplitude in the input signal remains constant right if the amplitude fluctuates a lot um first of all they don’t work very efficiently but that’s not our concern here efficiency is not 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 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

[39:19] 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 in amplifier um so do we 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’s 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 um

[40:06] 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 right
so in fact ignore the time variable here
that’s that’s of no consequence right
what will be the output input characteristics or transfer characteristics of an ideal amplifier um
straight line isn’t it
input is small output should be small
correspondingly if the input is large the corresponding output should be large simple
noting beyond that
so it’s 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 right
that’s for characteristics

[41:00] now practical power amplifiers will exhibit a kind of non linearity which is very commonly encountered called the saturation non linearity right
you know the effect
it’s basically arises because you have some finite value
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 non linear or situation more like this
it doesn’t remain linear for all values of the input amplitude 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 right then the output is no longer proportional to the input right
y of t is no longer a times x t right 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

[42:05] you have to introduce can you suggest some model for this curve that you have here
some suitable parametric model
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 this characteristics and that’s the usual thing that is done to model non linear components non linear devices right
(Refer slide time [42:29])
use polynomial models
so a typical model for example or such non linear things would be that \( y(t) = a_1 x(t) \)
which would have been the ideal thing right but you also have additional terms like \( a_2 x(t)^2 \) plus a three times \( x(t)^3 \) and so on and so forth and theoretically it’s possible to find a set of coefficients \( a_1, a_2, \ldots \) with the polynomial of some degree suitable degree such that this resulting characteristics resemble the actual curve that we have plotted right.

\( y \) versus \( x \) curve that we have plotted

[43:27] now suppose that is the case
suppose let’s take a very simple non linearity of the kind a one \( x(t) \) plus a two \( x(t)^2 \)
one thing is clear that the output is not a replica of \( x(t) \) right and therefore there is a distortion right because we would like the output to be simply proportional to \( x(t) \) but we have additional components like \( x(t)^2 \) and so on and so forth right 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 very important and very significant way

[44:12] if you remember i had made a comment when we talked about linear distortion that linear distortion which arises due to non ideal characteristics of this channel in terms of its transfer function can only do something good or something bad relatively to different frequency components that exist in the message signal right so but it cannot create additional frequency components isn’t it
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’s all it can do
that’s the linear distortion
non linear distortion on the other hand has a potential to generate frequency components
in the output message signal which were not even present in the input signal right and
that’s the very important way in which non linear distortion differs from the linear
distortion

[45:09] let me give an example
suppose the input signal is a pure sine wave just for the sake of appreciating this point
right
pure sine wave of frequency let’s say f one right
now what are the components present here
this will be a one cosine two pi f one t plus a two cosine two pi f two t whole square
cosine square right
so cosine square you can write as one plus cosine four pi f one t right
so now what are the components present in the y of t
they are f one and two f one
two f one was not present
input was containing only f one right
so the input signal contains let’s say a frequency component f one
output contains f one as well as two f one

[46:09] this is a very simple situation
suppose now the input signal contains two components f one and f two
what will you have now
we will obviously have f two um sorry f one two f one
you will also have f two two f two
in addition you will have f one plus minus f two right
you can see that
it’s a matter of just writing down the trigonometry and seeing that you will get terms
which is cosine of two pi into f one plus f two times t and two pi f one minus f two times
t
so this additional frequency components that are getting generated is the real problem in
non linear distortion right and that we call these components we call um intermodulation
components and this non linear distortion is for this reason also called harmonic
distortion or intermodulation distortion because different frequency components
modulate with each other modulate each other to produce additional frequency
components right
(Refer slide time [47:22])
so um therefore another name for non linear distortion is rather set of names in fact harmonic distortion because additional frequency components are being present or inter modulation distortion and that’s is that good or is that bad

that’s very bad and why is it bad

um power is an issue but it’s 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 right and obviously you are going to speak into somebody else’s band

suppose you are allotted a certain frequency band to work with you are transmitting outside that band

you 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 right

so this will cause what is called radio um it’s also called co channel interference or sometimes also simply called cross talk right

(Refer slide time [49:15])
yes please
(student speaking)
yes because you can see that these characteristics that will not depend on the value of these coefficients right
so how the power gets divided into frequency components will depend on the values of these coefficients in the model of non linearity
any other questions
(student speaking)
what you are saying is if f one and f two f one plus f two (student speaking)
[50:08] the point is you are not only going to see you are definitely going to create a distortion for yourself right
that's 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 right
f one minus f two will typically lie within and f one plus f two will lie outside right so both kinds of things are bad
f one minus f two creates distortion for us
f one plus f two creates distortion for somebody else
cross talk for somebody else and so on and so forth
okay we will stop here “ “
thank you very much