VSB Modulation (Contd.)
Superhet Receiver

[1:04] um so we will [noise] um lets recall that last time we were looking Vestigial sideband modulation we will continue with that discussion today for sometime (refer slide at [1:16])

and try to wind it up and in the rest of the time I will introduce the concept of what is called Superhet receiver for amplitude modulation systems which will answer number of questions that have been raised from time to time about tunability etcetera of the receiver right so we will look at these two issues today um to recollect our discussion on Vestigial sideband modulation we said that it is useful to have a sideband filter mechanism in which we don’t try to cut-off one of sidebands completely but permit a portion of the other sideband to be also transmitted
basically it simplifies our filter problem
right we are able um we are able to work with filters which are more practical
which um which have the the requirement of having a gradual roll-off rather than a
certain cut-off
right whether you want a high-pass filter or low-pass filter depending on whether you
want to transmit the upper sideband or lower sideband we need to um do this right
however if you want this feature we need to make sure
that this filter would produce a signal
which would lead to a perfect demodulation of a
transmitted signal
perfect recovery of the message signal right
so when impose this constraint we found that this gradual roll-off filter which leads to
Vestigial sideband modulation must satisfy a certain constraint
right and the constraint derived was

[3:02] this equation
so the
filter H f
which is the sideband filtering filter
Vestigial sideband filtering filter must satisfy the condition
(refer slide at [6:49])

that H f minus f c plus H f plus f c
must be equal to one right
this constant one of course is arbitrary and could be any other constant
main thing is it is a constant value
now to understand what this means
just look at the picture
lets say we talk about high-pass filter
take the upper sideband right
this would have been your ideal um upper sideband filter
the VSB filter on the other hand
so this we are plotting H of f
assuming it to be [4:01] or you consider this to be mode of H of f
[4:04] for the VSB filter we have permitted that this
at the around the cut-off frequency we have a gradual roll-off
right so the transfer function looks something like that
right
and basically what we are saying is this this
transfer function should have a shape that satisfies this condition
when it satisfies this condition
we have a coherent detector
would be able to perfectly recover the message signal m t back right
[noise]
right
this is what we discussed

[5:03] lets see what this means
this constraint means on a filter transfer function can you um
guess what it would it is a kind of symmetry implied here right
what what um for example if you look at this condition carefully this is basically saying
that if i hold this portion of the roll-off
back in to this
region right
then um [5:34] mirror image of this
around this point right
the head of these two curves they turn out to be a flat curve
a constant curve right
so there is a kind of symmetry involved around of f c
so this kind of symmetry that that you want he filter to have is called Vestigial symmetry
so the filter required to have

[6:05] what is called Vestigial symmetry
[noise]
so the filter characteristics must be mus must have this Vestigial symmetry
[noise]
this is the um conclusion of the discussion we had that day ok
so basically what this implies is um when you fold this back in to the region the sum of these two curves the solid curve and the dotted curve turns out to be a constant one ok think about that it is really obvious now

[7:00] if you were to go back to our issue of representation right this um so far we have talked about how to design the Vestigial sideband filtering filter at the rece at the transmitter so that the message is perfectly recovered now let us look at the issue of representation of the VSB signal (refer slide at [14:59])

right yes please

not the advantage it is a requirement it is not that we are seeking any advantage unless we have the Vestigial symmetry we will not be able to recover the message signal back right this a um our our entire discussion was based on how to design the filter H of f so that m t is recovered by the coherent detector we wanted the output v not t of the coherent detector to equal m t k times m t right
so if we want that to happen this condition must be satisfied we can’t choose any arbitrary
filter with gradual roll-off
to do the VSB filtering that’s the implication right you have to have a filter which has the
symmetry around fc the roll-off portion must exhibit the symmetry if the perfect
recovery is to be made possible
so the VS filter VSB filter has to satisfy this constraint
it’s not he advantage we are talking about it is the requirement
ok so please understand that
ok
now
like the other modulated signals that we have discussed so far
the VSB signal is also band-pass signal
and we know that every band-pass signal has quadrature representation
so lets say it has an in phase component s of I t into cosine two pi fc t and a quadrature
phase component s Q t s of Q t sine two pi fc t right this is in phase component and this is
the quadrature phase component which every band-pass signal must have

for the dbsc signal s of I t is m t right
and s of Q t is zero
for the um
SSB signal single sideband signal this is m t and this is Hilbert transformer of m t ok
the question is what are the in phase an quadrature phase components
for the case of
VSB signal
that’s what we need to understand
ok
to proceed further lets look at the frequency domain relationship between s I
f and s f
right
um what is the spect how can we express the spectrum of s I t in terms of spectrum of s t

Ask your self a question how will you recover s I t from s t we already know that remember we discussed quadrature multiplexing and demultiplexing right
if i want the in phase component from this message signal s of t
what should i do
[noise] i have to do coherent detection with respect to cosine two pi fc t
i must have a coherent detector with a local carrier of cosine two pi fc t as th the carrier
signal multiply these two and pass the product to a low-pass filter so if i multiply s t wilth cosine tow pi fc t what is the spectrum of that signal
that will s f minus fc plus s f plus fc
right
and that spectrum wi this spectrum would have [noise] excuse me a component around baseband and a component around
two fc
[11:00] out of that when you do the low-pass filtering what do you do you will take only the component around the baseband
so basically what we are saying is that the signal s I t that you will recover will have spectrum which equals this
for f less than the bandwidth
right
mode of f less than the bandwidth but equals zero for mode of f greater than bandwidth
because you are low-pass filtering it
right
this has two this two [11:33]
non zero in this region and non zero around two f c
the that portion is being cut-off by the low-pass filter and therefore the spectrum of s I t
that you have in this representation must be given by this expression
are all with me
hmm
because you can see that s I t can be obtained from s t by multiplying s t with cosine two pi f c t and passing the result to low-pass filter and that fact leads us to this expression

[12:03] for the spectrum of s I t in terms of the spectrum of s I s t
any questions in that
now
you may recollect
that we had um an expression for s [noise] of f as
half A c times M f
into H f minus f c
sorry
um
it was um
not this S of f was
S of f is your VSB signal how was it how was it being generated
you are generating in by [noise]

um

[13:02] first multiplying m t with
cos omega omega c t and then
passing through the VSB filter right
so that um that spectrum would be M f minus f c plus M f plus f c
right this is our dsbsc signal for the frequency domain
and we are passing it through
H of f
so this was S of f
now use this result here
to get an expression for S I f
and we will do similar exercise later for s Q f
right but lets first complete the exercise for S I f
are you with me
we are saying that this the expression for the spectrum of s t which is a VS VSB signal
[noise] Vestigial sideband signal right
if i use that expression here i get the spectrum of
the in phase component of the VSB signal
right

[14:06] so
this would then become S I f would be able to tell what it will be
you will have
um
if you remember we also discussed what was the nature of S f minus f c plus S f plus f c
right we have
gone through one um further step when you are looking at the demodulation of dsbcs
signal
and that contained four terms obviously because two terms come from here and two
terms will come from each of these two will have two terms and therefore we get four
terms
out which only two were in the baseband which is what you want
the other two terms where around two f c
right so
um if you substitute that here you get four terms
two of which are in the baseband and those two terms we need to look at and those tow
terms if you recollect or if you have your notes in front of you would or in fact obvious
and we just look again that it is obvious
(refer slide at [20:25])
it will be half A c M f into H of f minus f c plus H of f plus f c in fact it is from this point that we derived our requirement so this should be equal to constant
Right so this is what you will get
now for
sorry B
for the spectrum of S I f
we know the fact that this filter is anyway satisfying the constraint the condition is equal to one
this becomes half of A c times M f
right
which itself is of course limited to mode of less than B

because M of f is supposed to have a bandwidth of B so i don’t have to qualify this any further
so what does it mean what is s I t s of I t
what is s of I t half of A c into m t
so once again the in phase component remains the same
where as the dsbsc signal
whether it is SSB signal and now whether it’s a VSB signal
the in phase component remains proportional to message signal m t
and that is the reason we are able to recover it through coherent demodulation
right
and that was of course made possible
because we have satisfied this constraint this plus this must be independent of f must be a constant
right
so i think everything falls in place
only thing left to consider is s Q t

[17:04] so if we go through the same exercise once again can you tell me
what will be the spectrum S Q f in terms of spectrum of s t
so we are going back to the expression of
how will you
ok i mean the answer to this question will be
how do we require s Q t from s t
and that that will give us the um relationship in the frequency domain also how will you
recover it
you multiply this with [noise] sine two pi f c t
and pass the results with low-pass filter
right so what will be the spectrum of s t sine two pi f c t
that's what we will see so it will now instead of
S f plus f c uh S f minus f c plus S f plus f c it will be j times
S of f minus f c minus S
of f plus f c

[18:00] just remember what is sine two pi f c t it is [doubt] pi f c t minus [doubt] f c t
divided by two [noise]
j right
use that result
use the frequency translation [18:14] before you transform and this is what you get right
straight forward
so this is of course this will the result
again as you know when you do this um when you when you pass s t
when you multiply s t with sine two pi f c t you get two te terms one around baseband and
the other around two f c so that you low-pass filter result right
so we are only looking at the portion of the spectrum this spectrum for
mode of f less than B once again because outside this it has to be zero because of low-pass filtering
right s Q two s of Q t will also be a low-pass signal
so

[19:02] this if you follow the [doubt] again substitute for S of f
right and go through that you get j by two A sub c M of f what will the difference
instead of this plus this we will get this minus this
ok everything else will remain the same
[noise]
ok
that’s what we will do
now how do you interpret this result
hmm interpretation of this result is that the low-pass signal \( s \) of \( Q \) which forms the quadrature phase component of the VSB signal right can be obtained from message signal \( m \) by passing the message signal \( m \) to a filter who’s transfer function is this right we are multiplying the spectrum of \( m \) with this transfer function so that means this signal \( s \) of \( Q \) is obtained by pass the message signal \( m \) to a filter with this transfer function right \( s \) of \( Q \) (refer slide at [24:52])

\[
\begin{align*}
\text{\( \xi_{Q}(t) \): obtained by passing } m(t) \text{ through a filter with TF} & \text{ \( H_{Q}(f) \), } \nonumber \\
\text{\( H_{Q}(f) = j \left[ H(f-f_{c}) - H(f+f_{c}) \right] \), } \nonumber \\
\text{\( m(t) \rightarrow m'(t) \), } \nonumber \\
\text{\( \xi_{VSB}(t) = \frac{1}{2} A_{c} m(t) \cos(2\pi f_{c} t - \frac{1}{2} A_{c} \cos(2\pi f_{c} t) \right) \), } \nonumber \\
\end{align*}
\]

can be therefore considered as obtained by passing
original message signal \( m \) through a filter with transfer function
let us say \( H \) of \( f \)

\[
\begin{align*}
\text{\[21:01\]} \text{ \( H \) of } f \text{ is } \\
\text{\( j \text{ times } H \text{ of } f \text{ minus } f \text{ c minus } H \text{ of } f \text{ plus } f \text{ c } \), } \nonumber \\
\text{\( \text{[noise]} \), } \nonumber \\
\text{\( \text{lets call this signal } \), } \nonumber \\
\text{\( m \) t convolved with or passed through} \nonumber \\
\end{align*}
\]
H Q f as m prime t

right
[doubt] result that so far we are s of Q t i am giving a new name to it which is i am calling it m prime t ok
[noise]
then
basically what we are saying is that the VSB signal
i have slightly changed my notation this time we have been using x of t so far
simply i have shifted to s of t i hope you don’t mind that
i didn’t realize while i simply was changing so your VSB signal therefore can be though of as half of A c times m t times cosine two pi f c t minus half of A c m prime t sine tow pi f c t
where
m prime t in obtained like this
ok
so it has some similarities with what is done in the SSB case
what was in the SSB case this was m hat t the Hilbert transformer so you are passing m t to a special filter called the Hilbert transformer
instead of passing through Hilbert transformer we are passing it through something different similar but not exactly Hilbert transformer
right it is something slightly different from that
to appreciate what is the difference lets look at the transfer function H Q f little more carefully but before i discuss that this also gives you an alternative method for generating dsb sc signals VSB signals

[23:07] right
just like the phasing method for SSB we can have a phasing method for the VSB right
what will be the um block diagram
you will have the message signal m t coming in
the in phase part will be obtained by simply multiplying with cosine two pi f c t
and the quadrature phase part before we multiply by sine two pi f c t we will pass it through
this filter H Q f
right and then multiply this with sine two pi f c t
the quadrature carrier and simply add the two are subtract the two
depending on which sideband you want to transmit fully or [doubt] partially
right so that’s your

[24:02] [noise]
and the de detection or the demodulation can be done coherently
ok
lets just spend a few minutes now discussing the nature of H Q f what does it look like
lets first recollect or lets
as we as we notice this discussion that i have had is valid for
SSB as well
right
because SSB does satisfy this requirement the filter SSB
this filter
this condition is satisfied this plus this is equal to one because this portion is zero and this portion is one something like that right and therefore that condition is satisfied this whole thing as to what H f m i um
this whole thing should be applicable also to SSB signals
so therefore lets first look at the nature of H Q f

[25:02] for SSB signals right
for SSB signal what is your transfer function
it is like this
(refer slide at [28:00])

basically what we hope to see
what do we hope to see that if we derive H Q f from H of f from this commission that we have just discussed
we must get the Hilbert transformer because we know that that is how quadrature component of SSB is generated
to the Hilbert transformation of m t
right
so H Q f should turn out to be the Hilbert transformer
lets see whether it does right this is your H of f
centered around um this is the high-pass filter this is the cut-off frequency of f of c
now
[26:00] Let's look at this filter \( j \) times \( \text{le} \) let me plot \( j \) times \( \tilde{H} \) \( Q \) \( f \)
or \( \text{um one by} \) \( j \) times \( \tilde{H} \) \( Q \) \( f \) or \( \tilde{H} \) \( Q \) \( f \) divided by \( j \)
because if you recollect
if we look at this expression \( \tilde{H} \) \( Q \) \( f \) is this
so I am taking \( j \) on the left on side and plotting \( \tilde{H} \) \( Q \) \( f \) upon \( j \) right that is equal to \( \tilde{H} \) \( f \) minus \( f \) \( c \) minus \( \tilde{H} \) \( f \) plus \( f \) \( c \)
ok
so \( \tilde{H} \) \( Q \) \( f \)
\( \tilde{H} \) \( f \) minus \( f \) \( c \)
takes you shifts this to right
to this point
so you get this
right of course this portion will go around two \( f \) \( c \) we are not interested in the that
right we are only interested in the portion in the low-pass region right
\( \tilde{H} \) \( f \) plus \( f \) \( c \)

[27:02] will shift this portion to the left
Right and you are taking the difference of these two
so here you will get plus one and here you will get minus one
right so you get your signal function back
right so \( \tilde{H} \) \( Q \) \( f \) upon \( j \) is equal to signum minus signum um so basically what you get is \( \tilde{H} \) \( Q \) \( f \) upon \( j \) is equal to minus signum of \( f \)
right
and that's precisely what the Hilbert transformer is \( \tilde{H} \) sub \( Q \) \( f \) is equal to minus \( j \) times
signum \( f \) which is a Hilbert transform
[noise] so as expected this theory leads us to a filter \( \tilde{H} \) \( Q \) \( f \) which is nothing but the
Hilbert transformer for the case of
SSB right

[28:00] for the VSB the filter would be slightly different that's about all
let's [doubt] what the what the filter be in this case
(refer slide at [31:14])
this is the VSB signal
f c this goes to left side f c minus beta
this is minus f c and this comes to f minus f plus beta
right
this is a VSB filter
right
now lets apply this result here and plot once again one by j times H Q H sub Q f

[29:05] [noise]
When you shift this to right what we will get is
something like that
right
when you shift this to the left we will get something like that
when you subtract the two
right
this is coming form H f minus f c
this is coming from
H f
plus f c
by looking only the translation towards the baseband of course each of these components
also gets translated to two f c which we are not looking at
so now if we subtract the two
the filter that you will see
is going to have
this kind of

[30:02] characteristic so this is one by j H Q f
right
i have not plotted this very nicely but
it will be symmetrical as it should be
[noise]
right
it will this kind of characteristic
we are subtracting from this so at this point what will you get
half minus half which is zero it is passing through the origin
right and then of course at this point we get
a gradual decrease towards zero and at this point um become more and more negative
right
till it becomes minus one so this is plus one here minus one here and between minus beta
to plus beta there is a gradual roll-off
so H Q f is j times this
right so it’s not exactly signal function anymore right

[31:03] that’s the only difference
so it’s very similar to um sorry the limit as beta tends to zero
this will reduce to the signal function
ok
[noise]
are you with me
so lets try to wind up the discussion on Vestigial sideband modulation
in Vestigial sideband modulation therefore what we have learned is a
we will use slightly more bandwidth than the bandwidth of a SSB signal right
the amount that you will like to use more will depend on the kind of
flexibility you want in your filter designer
right
the value of beta that you use is the design parameter you can choose it to be a small
value you could choose it to be a large value depends on how easy
it is to a realize a filter that you want
right or how much cost you are ready to incur for designing for um making the filter
right because um whether you want less complex filter or more complex filter right if you
want a gradual roll-off you can make it a less complex filter only thing is cost will be in
terms of a larger value of beta that means a poorer

[32:14] bandwidth efficiency right
but if you pay this cost the advantages will be that you can save considerable amount of
bandwidth
still
for large bandwidth signals like picture signals which have
rich low frequency content and VSB as we learned as we discussed last time is
particularly relevant to those signals which have a fairly
large amount of low frequency content and therefore you cannot tolerate any distortion in
the low frequency region but you clearly face distortion which is likely to happen when
you design a sharp cut-off filter and then the phase characteristics at the point of
cut-off will typically be highly known in your
which will be very very undesirable

[33:02] particularly for signals like picture signals and fax signals
ok so in a nutshell that summarizes discussion on VSB so therefore now we can
imagine for tv transmission
hmm
this would be a natural choice when you are transmitting pictures [noise]
right we will discuss tv transmission later
but you can keep this in mind
right
where Vestigial sideband signals will be particularly the right kind of modulation
to use for tv transmission for tv picture transmission in fact that is the case
so um picture transmission in television actually uses Vestigial sideband modulation
right because you have very large bandwidth signal here
if you don’t cut-off one of the sidebands you will [33:50] too much bandwidth
five megahertz will imply ten megahertz bandwidth around the carrier right
which is very very large amount of bandwidth so to save that bandwidth and still not have
these problems that we have discussed you decide to go for

[34:05] VSB because you can’t go for SSB we discussed
ok we will return to this point when we discuss tv transmissions
any questions
any questions
none everything is clear
good lets hope it is really clear
[noise]
no no m prime t is not the Hilbert transformer of m t
[noise]
no neither m prime f
i think you have not understood what i have said then
[noise]
no no only for the case of SSB
i just proved to you that for the case of SSB this theory that i have developed is
consistent with what we have discussed earlier that’s all i said

[35:07] ok whatever theory we have developed for the general sideband filtering
is consistent with what we have learned for the case SSB signal because this filter H Q f
then reduces to
the Hilbert transformer
but in general it is this kind of a filter
right which is not a Hilbert transformer [noise]
[noise]
sorry
[noise]
yes the roll-off that you will have will be dependant on the value of beta
right if the value of beta becomes zero
we get the single sideband phase back and it reduces to the Hilbert transformer
[noise]
yes please
[noise]
neither in SSB nor in VSB
[noise]
yes
[noise]

[36:01] um we will discuss that later it is a good question but the answer is that actually
we don’t we can’t tolerate any
phase in coherence in quadrature multiplexing not even one bit
there will be a cross talk if there is even a small amount of either frequency difference or
phase difference which you don’t which we don’t like to have where as in the case of
SSB and VSB if you remember have a second method of demodulation in which i add a
carrier locally and do then do envelope detection
which is much less sensitive to these problems than the coherent detector is
[noise]
and i will be able to the same thing for VSB
one method of demodulating the VSB signals would be precisely the second method that
we discussed for the SSB signal
that is the Carrier Reinsertion Method
we look at that in fact the theory is exactly the same
there is no difference except that m hat t m hat t gets replaced with m prime t

[37:04] so we have a second method and that is it is it is a existence of the second method
which makes VSB and SSB more acceptable than
[37:15]
i hope i have answered your question
any other questions ok
lets proceed further then [noise]
now i am going to slightly change the topic here it is we are on a subject of broad of
amplitude modulation and its um applications but now we are coming to some technical
aspects and before that um one of the technical aspect we need to discuss is the kind of
receivers that we need to use of course we have discussed the structure of the
demodulator
right for example if you are doing am transmission we know um what we are going to use
we use we are going to use envelope detector

[38:01] and we can use the envelope detector also for the case of SSB and
um VSB with carrier reinsertion but at the moment lets lets concentrate on the amplitude
modulation aspect because that’s the one of the most commonly used analogue
modulations
amplitude modulation with carrier because of the simplicity of detector
but um as many of you asked at the time of this discussion
what will happen if i want to receive different signals
right because when you are having broadcast receiver you
don’t want to listen to only signal at one carrier frequency
there are so many different broadcasting stations each operating at a different frequency
right
and you want to be able to tune in to any one of them right
what are the issues concerned with that kind of a receiver
so we are now looking at rather than a just detector we want to look at the receiver as a whole

[39:01] right as to what all features receivers should have so that it is possible for me um
to have detection of a signal of my choice from amongst multiple signals that may be
available from the broadcasting areas around you right
so to do the discussion i first want to digress in to um slight generalization of our previous discussion on
(refer slide at [44:57])

Frequency Translation and Mixing
these are not new terms we have already discussed these
there is one issue i would like to discuss before i come back to the receiver
[noise]
so the term frequency translation so far we have been using in the context of going from baseband to a band-pass signal or from the band-pass signal to a low-pass signal

[40:02] right but actually for communication application the um term frequency translation and mixing are used in a much wider context typically we sometimes in fact it is quite often that we need to go from one frequency to another frequency right not necessarily from baseband to some fc and back right arbitrarily um we may want to go from band-pass signal at center at one frequency to band-pass signal center at another frequency right so let me start with that [noise] it is desirable in many applications [noise] to translate [noise] and translate i mean frequency translate a band-pass signal to a new center frequency [noise] in this the mechanism is the same the mechanism is

[41:01] multiplying [noise] the band-pass signal that you want to translate by a suitable carrier signal i suitable um carrier signal which could be [41:14] or a corresponding periodic signal and an appropriate signal multiplying an appropriate filter [noise] right [noise] and this process of multiplying appropriate filtering is what we call Mixing [noise] and if you remember um i introduced some other names also for this [noise] Converting [noise] right and Hetrodyning Mixing Converting Hetrodyning these are the names which are interchangeably so what we are saying is you have a band-pass signal lets say m t cosine omega one t for the sake of simplicity i am taking only the in phase component

[42:08] but you could take a general band-pass signal for general representation right with out any um loss of discussion um without any loss of generality so i have taken a band-pass signal which is centered around m t cosine omega one t where is it centered omega one right the center frequency is omega one [noise] lets say i want to translate this to a center frequency of omega two what should i do hmm i should have a local oscillator what should be the frequency of this local oscillator [noise]
hmm
omega one
[noise]
either omega one minus omega two or omega one plus omega two any one of the two
will be fine
right [noise]
right so i take the local oscillator
with a um with a frequency of either omega one plus omega two or omega one minus
omega two

[43:05] and
then when i multiply these two
i will get a sound frequency component and a
[43:13] frequency component right
some frequency component will be at a frequency
two omega one cen will be centered around tow omega one
[noise] plus minus omega two [noise]
right
which lets assume as much
quite large and can be removed by an appropriate filter but the complier that we want is
centered around
omega two so what kind of filter you need here
not l low-pass filter
the band-pass filter with a center frequency of omega two
ok that will lead to your m t cosine omega two t
as an exercise please check up that if instead of m t cosine omega one t i had return the
more general representation of band-pass signal

[44:03] right that is m t cosine omega one t um some in phase component and some
quadrature phase component [noise]
and through this process you still get the
translation in tact right
i will leave that as an exercise
so um [noise] this signal at this point
basically the theory is that if you look at this signal e t this you can right as m t cosine
omega two t
plus m t
cosine
two omega one plus minus omega two t [44:41] the trigonometric identities
right and through the band-pass filtering you are
eliminating
this component
retaining only
this [44:54]
so this process in general is called Mixing
[45:01] now there is a very common problem that happens in this process that one encounters in this process right it is a common problem in all mixing all mixers asso of this kind it is associated with all mixers of this kind suppose you take omega one plus omega two t here right and lets say at the input along with this signal i also have a signal at frequency omega one [noise] plus two omega two t ok so along with this i have another signal which is centered around omega one plus two omega two t now what will be the output of this system have you understood the question the question is if at the input with the same system which is adjusted to signal of frequency omega one that is adjusted to translate the signal of frequency omega one to omega two

[46:04] if at the input of the system i have a signal omega one plus two omega two what will be the output [noise] think carefully see basically um look at the sound frequency argument what will be the frequency components that you will have here omega one plus two omega two minus omega one plus omega two and of course some component the some component we don’t have to worry about and what is this equal to omega two again so this will also be passed by this band-pass filter right so for every frequency that it is supposed to translate there is also what is called an image frequency which it will translate to the same frequency not only it will translate omega one to omega two it will also translate omega one plus two omega two to omega two right

[47:01] so omega one plus two omega two that frequency is said to be the image frequency of omega one ok so basically if um if you have not understood let me repeat it (refer slide at [50:42])
what we are saying is there is a common problem with mixers
so i want to elaborate on this problem
and this problem is as following
the inputs of the form \( k \cdot t \cos(\omega_1 \pm 2\omega_2) \)
he here i had taken the bas um local oscillator will be \( \omega_1 + \omega_2 \)
in fact again \( \omega_1 - \omega_2 \)
then the image frequency would have been \( \omega_1 - 2\omega_2 \)
right basically the sine would be the same

[48:04] so inputs of the form \( k \cdot t \cos(\omega_1 \pm 2\omega_2) \) are also translated to

to a center frequency of \( \omega_2 \)
to see this lets go through the trigonometry
what we are saying is \( \cos(\omega_1 \pm 2\omega_2) \) multiplying with \( \omega_1 \pm \omega_2 \)
t
when you multiply this with
sorry \( k \cdot t \cos(\omega_1 \pm 2\omega_2) \) when you multiply that with two cosine
what are we multiplying with what’s your local oscillator frequency
[noise] either \( \omega_1 + \omega_2 \) or \( \omega_1 - \omega_2 \)
right
this leads to two components [noise]

[49:04] one is the difference frequency component
and the other which is
the sound frequency component [noise]
right
and this will again show up at the output of the same band-pass filter this will get
eliminated
right
this is point i was making
so when you use a local oscillator [49:30] frequency
basically what you are saying is omega one will go to omega two
omega one plus minus omega two will also go to
omega two
right
so this [noise]
sorry this will be two omega two ok omega one plus and of course the sign will be same
more precisely
if L o is omega one plus omega two
the image frequency will be

[50:00] omega one plus two omega two
if your L o is omega one minus omega two
the image frequency would be
ok this misnomer i shouldn’t use IF IF is typically used to denote something else but let
me for the moment use IF
this will be omega one sorry minus two omega two
ok
and therefore if both the signals were simultaneously present
the signal at omega one and the signal at
omega one plus two omega two [50:38] local oscillator of this frequency
the band-pass signal output will contain components
corresponding to both of them right and there will a cross talk
you will get some of two signals
is it clear right
so that’s the problem you need to keep in mind when you are working with mixers there
is concept of an image frequency which has a potential to interfere with this frequency of
interest

[51:06] right
th the image frequency if the signal at that frequency is present will interfere with the
signal of interest which is at frequency omega one
[noise]
is it clear
basically that’s the point that i want to convey
keep that in mind and now we shall discuss um
[noise]
ok so we will discuss next time the concept of a Superhetrodyne thank you very much