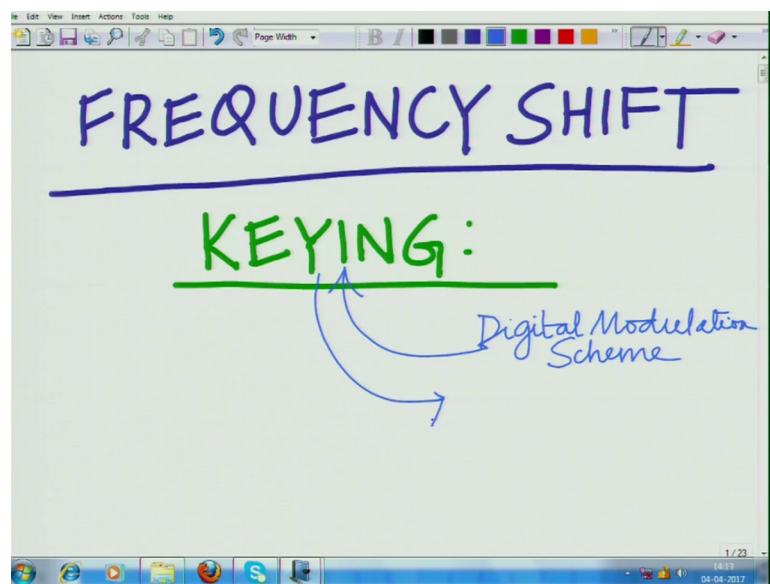


Principles of Communication Systems – Part II
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Department of Electrical Engineering
Indian Institute of Technology, Kanpur

Lecture - 15
Introduction to Frequency Shift Keying (FSK)

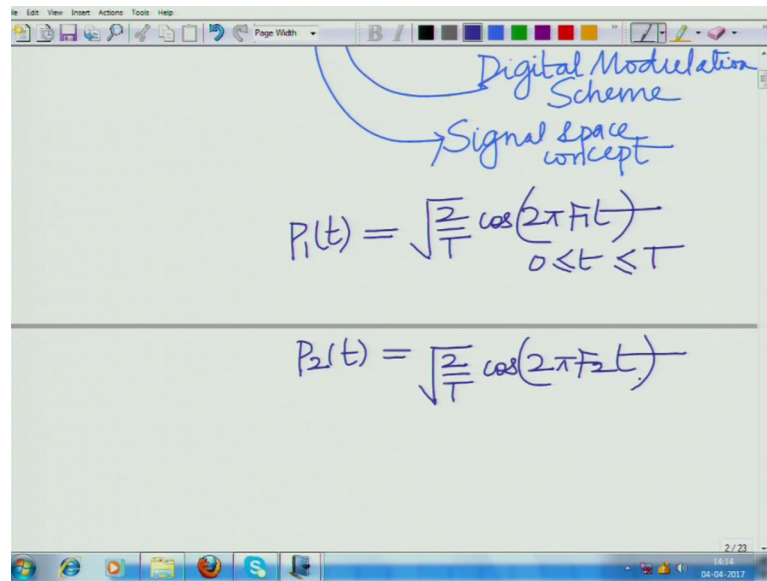
Hello, welcome to another module in this massive open online course. In this module, let us look at another digital modulation technique that is frequency shift key.

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So, we will be looking at a new digital modulation scheme which is termed as frequency shift key. This is a digital modulation scheme as we have already said and it is based on the signal space concept that we have previously described.

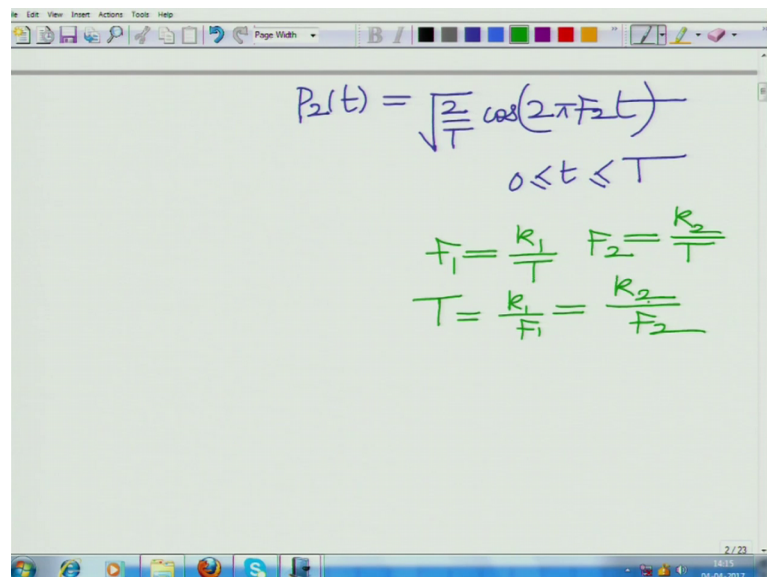
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Handwritten notes on a digital modulation scheme and signal space concept. The notes are written in blue ink on a white background. At the top, there is a title "Digital Modulation Scheme" with a curved arrow pointing to "Signal space concept". Below this, the first pulse is defined as $P_1(t) = \sqrt{\frac{2}{T}} \cos(2\pi F_1 t)$ for $0 \leq t \leq T$. The second pulse is defined as $P_2(t) = \sqrt{\frac{2}{T}} \cos(2\pi F_2 t)$ for $0 \leq t \leq T$.

Where we have 2 pulses correct which form an orthonormal basis for the signal space. So, let us again consider our 2 pulses $P_1(t) = \sqrt{\frac{2}{T}} \cos(2\pi F_1 t)$ for $0 \leq t \leq T$ and we also have $P_2(t) = \sqrt{\frac{2}{T}} \cos(2\pi F_2 t)$ for $0 \leq t \leq T$.

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Handwritten notes on the relationship between frequency, period, and number of cycles. The notes are written in blue and green ink on a white background. The first pulse is defined as $P_2(t) = \sqrt{\frac{2}{T}} \cos(2\pi F_2 t)$ for $0 \leq t \leq T$. Below this, the frequencies are defined as $F_1 = \frac{K_1}{T}$ and $F_2 = \frac{K_2}{T}$. The period T is then defined as $T = \frac{K_1}{F_1} = \frac{K_2}{F_2}$.

$0 \leq t \leq T$ and we have F_1 or basically we have well T equals K_1 contains K_1 cycles or the wave F_1 equals over T and F_2 equals K_2 or T over T is basically K_1 over F_1 and K_2 equal over F_2 equals K_1 over F_1 and this is

also equal to K_2 over F_2 contains K_1 cycles of P_1 that is a sinusoid for sinusoidal frequency F_1 and contains K_2 sine cycles of the other sinusoid at frequency F_2 and we have also seen that these 2 pulses are basically.

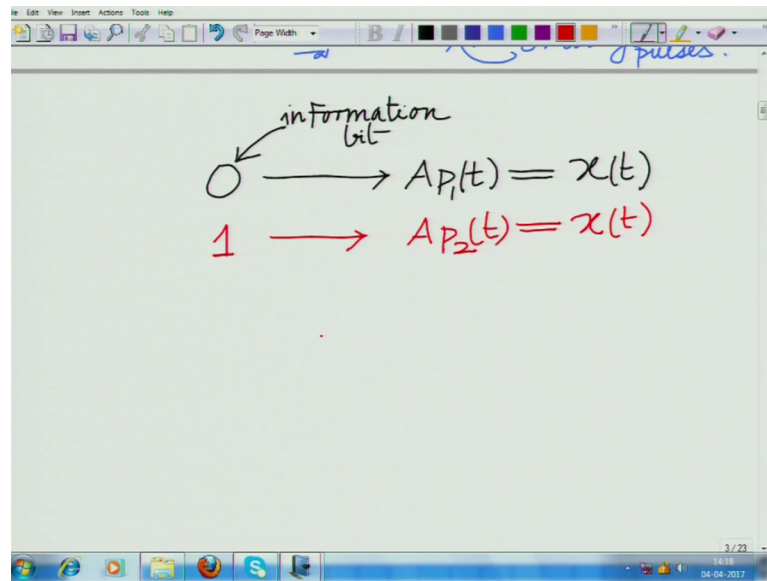
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The image shows a digital whiteboard with handwritten mathematical expressions. At the top, the energy of two pulses is calculated: $E_p = \int_{-\infty}^{\infty} P_1^2(t) dt = \int_{-\infty}^{\infty} P_2^2(t) dt = 1$. An arrow points from the result '1' to the text 'Both have unit Energy'. Below this, the inner product of the two pulses is calculated: $\int_{-\infty}^{\infty} P_1(t) P_2(t) dt = 0$. An arrow points from the result '0' to the text 'orthogonal pulses'.

The first thing is that the energy of the pulses is E_p equals P_1 squared dt equals minus infinity to infinity P_2 square equals one that is both pulses have unit energy and more importantly if you look at the inner product this is equal to 0 which means these are orthogonal pulses these are orthogonal pulses.

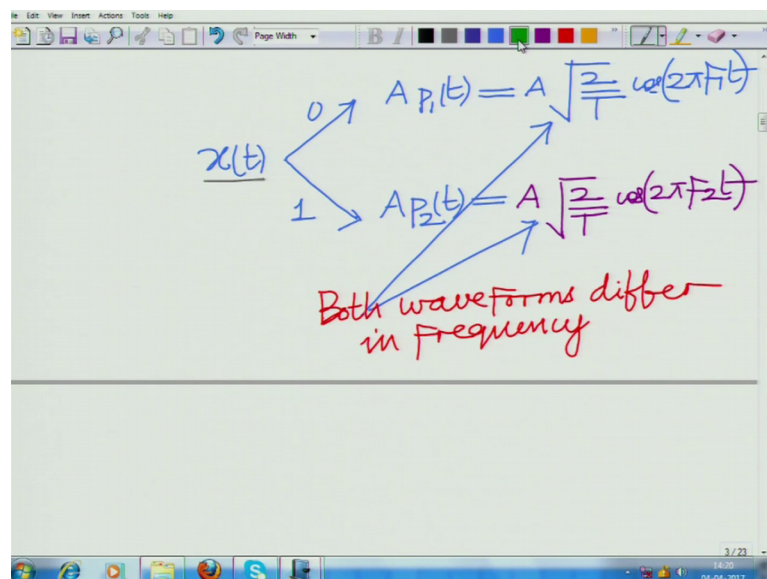
So, we have 2 pulses which are orthogonal to each other and also the pulses are basically both of them have unit energy all right these are orthonormal these form the basis orthonormal basis of the signal space.

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Now, the digital modulation signal is given as $x(t)$ equals well A times $P_1(t)$ that is let us look at it this way that is corresponding to the information symbol 0, this is the information bit, 0 is mapped to A times $P_1(t)$ which is equal to $x(t)$ and one the information symbol, 1 is mapped to A times $P_2(t)$ which is equal to $x(t)$. So, other words you have to write it clearly.

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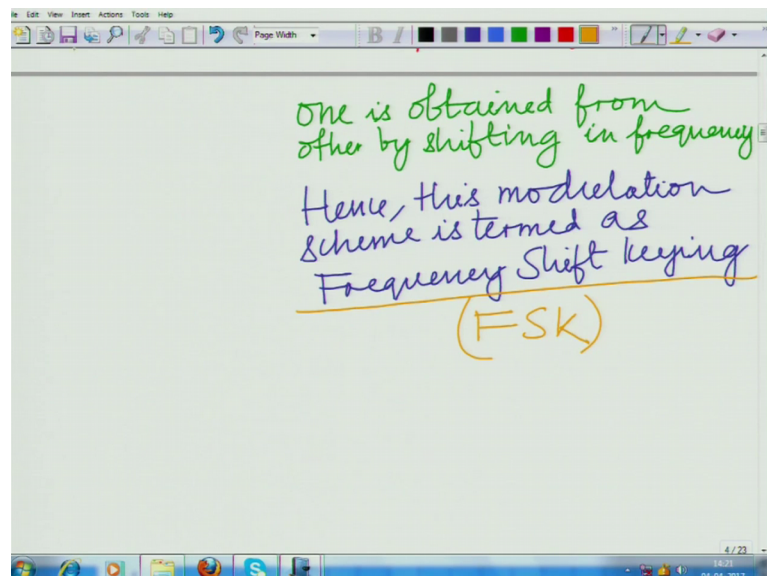
I have x , I have $x(t)$ corresponding to information symbols 0, corresponding to information symbol 1, this is A times $P_1(t)$ which is A times square root of 2 by T cosine

$2\pi F_1 t$ and corresponding to 1. This is $A \cos 2\pi F_1 t$ which is $A \cos \sqrt{2\pi F_1 t}$ and now you can see that if you now look at this $x t$, correct, what you are doing is you are either mapping it to waveform (Refer Time: 06:51) $A \cos \sqrt{2\pi F_1 t}$ or you are mapping it for the information with one to the waveform $A \cos \sqrt{2\pi F_2 t}$.

Now, if you can observe carefully these 2 waveforms $\cos 2\pi F_1 t$ $\cos 2\pi F_2 t$ differ in the frequency that is you are switching the frequency from F_1 to F_2 and F_2 to F_1 , correct. So, to indicate A 1 you are transmitting waveform (Refer Time: 07:17) frequency F_1 to indicate to indicate A 0, you are transmitting the waveform at frequency F_2 to indicate A 1 you are transmitting a waveform with frequency F_1 , correct and therefore, since these 2 waveforms are obtained from each other by shifting their frequencies this freq scheme this digital modulation scheme is termed as frequency shift key.

So, which is an important point? So, you can observe that both waveforms are waveforms differ in frequency that is one is obtained from the other by shifting in frequency.

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Hence this modulation scheme is termed as hence this modulation scheme is this is termed as frequency shift key or F S K, this is the waveforms are basically frequency shifted versions of each other.

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Assuming 0, 1 occur with probability $= \frac{1}{2}$ each,

$$\text{Average energy} = \frac{1}{2} A^2 E_p + \frac{1}{2} A^2 E_p.$$

Further once again assuming that 0 and 1 occur with equal probabilities occur with probability equals half we have average energy equals half A square E P plus half A square E P.

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$$= A^2 E_p.$$

$$A^2 E_p = E_b.$$

constant energy/bit E_b .

$$\Rightarrow A^2 = E_b.$$
$$\Rightarrow A = \sqrt{E_b}.$$

Which is equal to A square E P and similar to previous, if we want to set similar to the previous modulation schemes, if you want to set energy per bit is constant that is you want to set a constant energy per bit E_b implies a square E P has to be E_b and we know

E_p equals 1 which implies a^2 equals E_b which implies A equals square root of E_b .

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Handwritten derivation on a whiteboard:

$$\Rightarrow A^2 = E_b$$

$$\Rightarrow A = \sqrt{E_b}$$

Then, the transmitted waveform $x(t)$ is defined as:

$$x(t) = \begin{cases} 0 & \sqrt{\frac{2E_b}{T}} \cos(2\pi F_1 t) \\ 1 & \sqrt{\frac{2E_b}{T}} \cos(2\pi F_2 t) \end{cases}$$

So, that is what we have and therefore, once again the transmitted waveforms are either A times square root of 2 over T that is remember we will get 2 square root of E_b over T cosine $2\pi F_1 t$ or these this correspond to 0 correspond to one 2 square root of cosine $2\pi F_2 t$ corresponding to will corresponding to 0.

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Handwritten receiver model on a whiteboard:

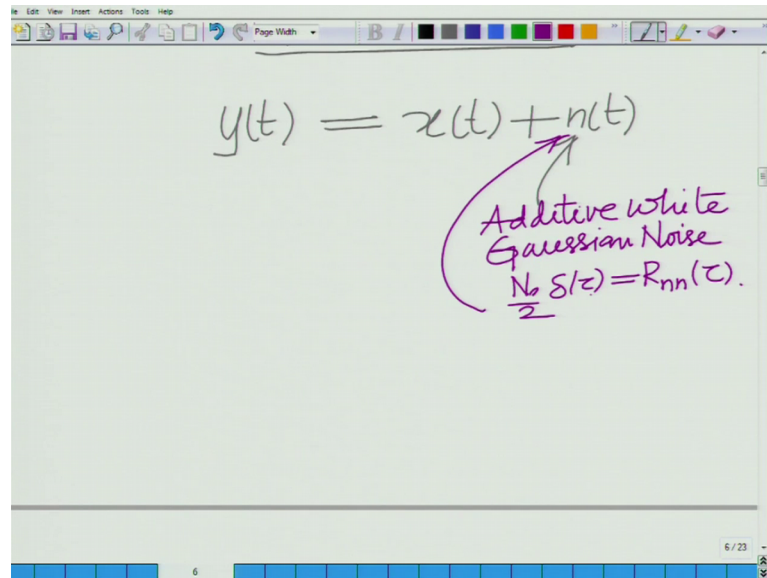
Receiver

$$y(t) = x(t) + n(t)$$

(An arrow points from $n(t)$ to the text "noise" in the original image, but the text is not present in the provided image.)

So, now, therefore, now consider what happens at the receiver at the receiver in the digital communication system once again we have $y(t) = x(t) + n(t)$ where $n(t)$ is additive white Gaussian noise.

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The image shows a presentation slide with a whiteboard background. The equation $y(t) = x(t) + n(t)$ is written in black. A purple arrow points from the text "Additive white Gaussian Noise" to the $n(t)$ term. Below this text, the autocorrelation function is written as $\frac{N_0}{2} \delta(\tau) = R_{nn}(\tau)$.

This is your standard which we have seen so many times. So, far this is your additive white Gaussian noise with P S D or with autocorrelation $\frac{N_0}{2} \delta(\tau)$ this is the autocorrelation.

Now, corresponding to the transmission of 0 you have be remember there are 2 pulses A times P 1 t is corresponding to the transmission of 0 A times P 2 t is corresponding corresponds to the transmission of the information bit one therefore, corresponding to the transmission of 0 you will have the addition of noise A times P 1 t plus $n(t)$ corresponding transmission of one you will have the addition of noise A times P 2 t plus $n(t)$.

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Gaussian Noise
 $N_0 S(t) = R_{nn}(t)$

0 $\rightarrow y(t) = A p_1(t) + n(t)$
1 $\rightarrow y(t) = A p_2(t) + n(t)$

Two pulses $p_1(t), p_2(t)$

So, let us note that. So, corresponding to the transmission of 0 we have $y(t)$ equals A times $p_1(t)$ plus $n(t)$ corresponding to the transmission of y of 1 or bit one we have A times $p_2(t)$ plus $n(t)$.

Now, we have an interesting conundrum or an interesting puzzle over here we have 2 pulses $p_1(t)$ and $p_2(t)$.

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Matched Filter:
 $h(t) = p_1(T-t) ?$
 $= p_2(T-t) ?$

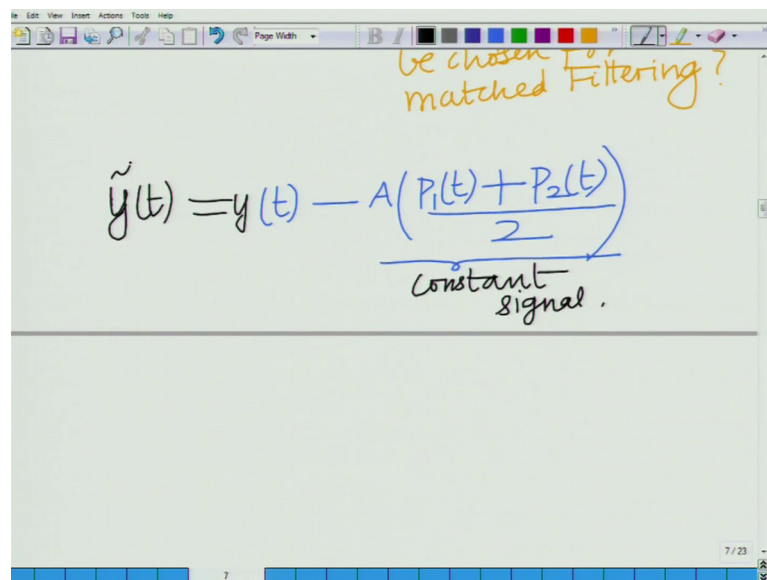
Which pulse should be chosen for matched filtering?

Now, what we need to determine is when we matched filter if you want to use we are not seen a situation like this before if you use a matched filter is $h(t)$ equals p_1

t minus τ or is it equal to surprisingly is it equal to P to t , so which one should we choose? So, which pulse should be chosen for matched filtering that is the basic question that we are trying to address, which pulse which pulse should be chosen which pulse should be chosen for matched filtering and it is not immediately obvious which pulse should be chosen for matched filtering. So, let us try to address in a different way.

So, let us what we will do is we will try to modify this received signal $y(t)$ a little and try to see if we can derive from it what the optimal matched filter should be.

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The image shows a digital whiteboard interface with a toolbar at the top. Handwritten in blue ink is the equation $\tilde{y}(t) = y(t) - A \left(\frac{P_1(t) + P_2(t)}{2} \right)$. Below the fraction in the parentheses, the words "constant signal." are written in blue ink. In the top right corner, the text "be chosen for matched filtering?" is written in orange ink. The bottom right corner of the whiteboard shows the page number "7 / 23".

$$\tilde{y}(t) = y(t) - A \left(\frac{P_1(t) + P_2(t)}{2} \right)$$

constant signal.

be chosen for matched filtering?

So, for that what I am going to do is from $y(t)$ I am going to subtract A times $P_1(t)$ plus $P_2(t)$ divided by 2, now this is a constant signal. So, I can always do it constant signal and let us denote this as let us denote this by $\tilde{y}(t)$.

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For 0, we have,

$$y(t) = A p_1(t) + n(t)$$
$$\tilde{y}(t) = y(t) - A \left(\frac{p_1(t) + p_2(t)}{2} \right)$$

—

Now, therefore, corresponding to the transmission of 0 for 0 we have or for information bit 0 we have $y(t)$ equals A times $p_1(t)$ plus $n(t)$ $\tilde{y}(t)$ equals $y(t)$ minus A times $p_1(t)$ plus $p_2(t)$ divided by 2.

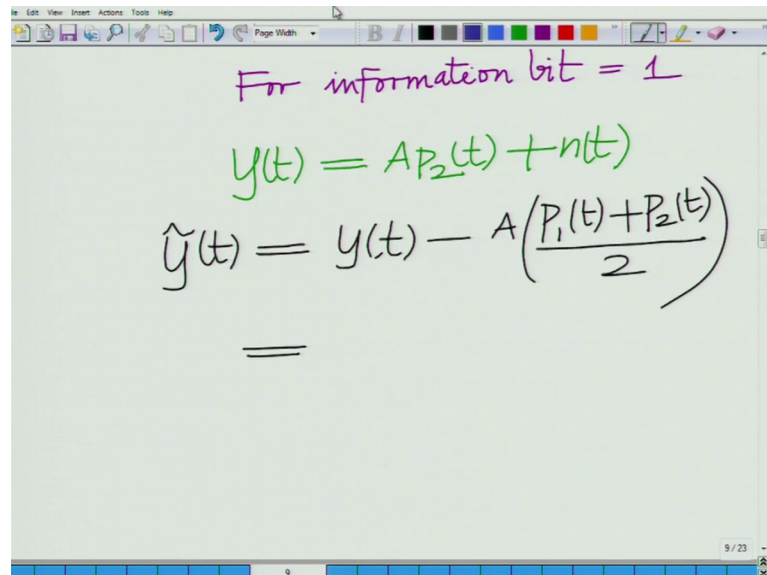
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$$y(t) = A p_1(t) + n(t)$$
$$\tilde{y}(t) = y(t) - A \left(\frac{p_1(t) + p_2(t)}{2} \right)$$
$$= A \left(\frac{p_1(t) - p_2(t)}{2} \right) + n(t)$$
$$\quad \quad \quad \tilde{p}(t)$$
$$= A \tilde{p}(t) + n(t).$$

Which is equal to well A times $p_1(t)$ minus A times $p_1(t)$ it has $p_1(t)$ plus $n(t)$ minus A times $p_1(t)$ plus $p_2(t)$ divided by 2 this is equal to A times $p_1(t)$ minus $p_2(t)$ divided by 2 plus $n(t)$.

Similarly, so let us call this as \tilde{P}_t that is your $P_1(t) - P_2(t)$ divided by 2 is \tilde{P}_t . So, that is $A \tilde{P}_t + n(t)$.

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For information bit = 1

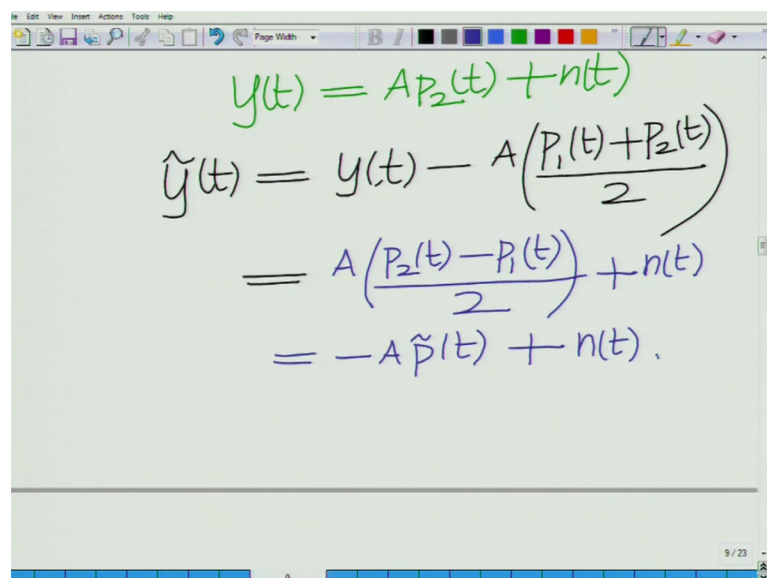
$$y(t) = A P_2(t) + n(t)$$

$$\hat{y}(t) = y(t) - A \left(\frac{P_1(t) + P_2(t)}{2} \right)$$

$$=$$

And further for one or for information bit one we have $y(t)$ equals A times $P_2(t)$ plus $n(t)$ and $\hat{y}(t)$ equals $y(t)$ minus A times $P_1(t) + P_2(t)$ divided by 2 which is equal to well it is $P_2(t)$ plus $n(t)$ minus A times $P_1(t) + P_2(t)$ divided by 2.

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$$y(t) = A P_2(t) + n(t)$$

$$\hat{y}(t) = y(t) - A \left(\frac{P_1(t) + P_2(t)}{2} \right)$$

$$= A \left(\frac{P_2(t) - P_1(t)}{2} \right) + n(t)$$

$$= -A \tilde{P}_t + n(t)$$

This is equal to A times $P_2(t) - P_1(t)$ divided by 2 plus $n(t)$. Now if you see; this is $P_2(t) - P_1(t)$ divided by 2 which is minus \tilde{P}_t . So, this is minus $A \tilde{P}_t + n(t)$.

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$$= -A\tilde{p}(t) + n(t).$$
$$\hat{y}(t) \begin{cases} A\tilde{p}(t) + n(t) \\ -A\tilde{p}(t) + n(t) \end{cases}$$

And now you have something very interesting if you look at $\hat{y}(t)$, but $\hat{y}(t)$, this reduces to $A \tilde{p}(t) + n(t)$ and in the other scenario this is minus $A \tilde{p}(t) + n(t)$.

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$$\hat{y}(t) \begin{cases} A\tilde{p}(t) + n(t) \\ -A\tilde{p}(t) + n(t) \end{cases}$$

This is similar to BPSK
and is obtained by replacing

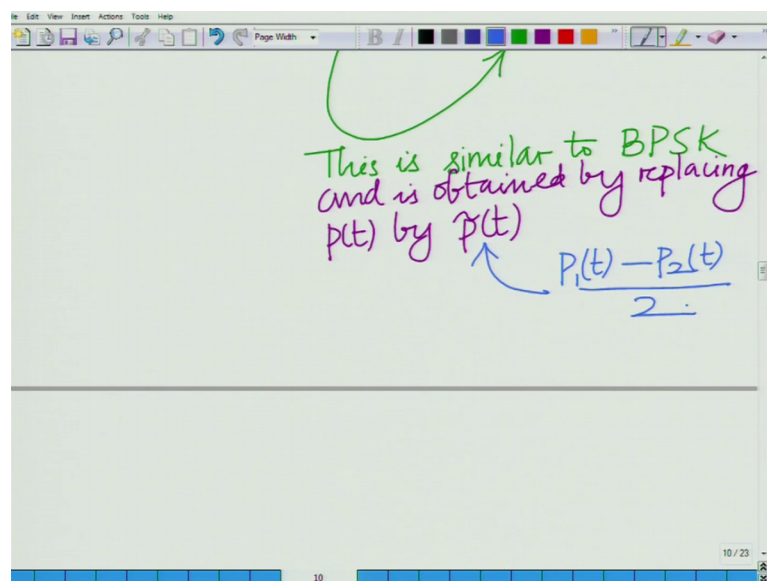
And if you look at this if you look at this; this is similar to binary phase shift key that is a type in binary phase shift keying we have $A \tilde{p}(t)$ and minus $A \tilde{p}(t)$ here we have $A \tilde{p}(t)$ and minus $A \tilde{p}(t)$ only difference is we are replacing P by \tilde{p} in comparison to binary phase shift key.

Therefore you can see that the optimal matched filter will be $\tilde{P}(t)$ not $P_1(t)$ or $P_2(t)$, but $\tilde{P}(t)$ and $\tilde{P}(t)$ is nothing, but $P_1(t) - P_2(t)$ divided by 2 the 2 is simply a scaling factor. So, basically the optimal matched filter will be proportional to $P_1(t) - P_2(t)$ neither $P_1(t)$ nor $P_2(t)$, but basically $P_1(t) - P_2(t)$ that is the important point that one has to realize.

We will justify a in a moment, but this is element that we keep in mind. So, c over I reduction is the goal we said sectorization is a good one for us to hang now what are the methods can we do for reducing the reducing the interference. So, there is another method that is use for reducing interference let me just highlight that for you. So, reducing c over I so here is a cell again we will use the same frequencies $F_1 F_2 F_3$ are available to me I have assigned F_1 to my user and let us assume that there is another co channel cell somewhere in the vicinity which also has a subscriber to whom F_1 has been assigned.

Now; obviously, there is interference between these 2 users. So, this particular user we could change the frequency from $F_1 F_2$ or so, this is similar. So, note that this is similar to binary phase shift key and is obtained by replacing and is obtained by replacing $P(t)$ by $\tilde{P}(t)$, $\tilde{P}(t)$ is basically your $P_1(t) - P_2(t)$ divided by 2 therefore, optimal matched filter.

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Therefore, the optimal matched filter is,

$$h(t) = \tilde{p}(T - \tau)$$

matched to $\tilde{p}(t)$
 $= \frac{p_1(t) - p_2(t)}{2}$

The image shows a digital whiteboard interface with a toolbar at the top. The handwritten text is in black ink. The equation $h(t) = \tilde{p}(T - \tau)$ is enclosed in a blue rectangular box. A red arrow points from the \tilde{p} in the box to the text 'matched to $\tilde{p}(t)$ '. Below this, the expression $= \frac{p_1(t) - p_2(t)}{2}$ is written in red ink. The bottom status bar of the whiteboard shows '11 / 23'.

Therefore the optimal matched filter is $h(t)$ equals $\tilde{p}(t)$ minus τ that is match to \tilde{p} tilde t \tilde{p} tilde t equals $P_1(t)$ minus $P_2(t)$ divided by 2.

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matched to $\tilde{p}(t)$
 $= \frac{p_1(t) - p_2(t)}{2}$

Factor of $\frac{1}{2}$ is simply a scaling factor which does not affect SNR

The image shows the same digital whiteboard interface. The red text from the previous slide is visible. A new note in blue ink is added below it. It says 'Factor of $\frac{1}{2}$ is simply a scaling factor which does not affect SNR'. A blue arrow points from the circled '2' in the denominator of the red equation to the 'Factor of $\frac{1}{2}$ ' text. The bottom status bar shows '11 / 23'.

Finally note that this 2 this factor of half is simply a scaling factor it does not affect the S N R, hence which does not affect the S N R.

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Hence, optimal matched Filter

$$h(t) = P_1(T-t) - P_2(T-t)$$

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Hence, optimal matched Filter

$$h(t) = P_1(T-t) - P_2(T-t)$$

$$= \sqrt{\frac{2}{T}} \cos(2\pi F_1(T-t)) - \sqrt{\frac{2}{T}} \cos(2\pi F_2(T-t))$$

optimal Filter at Receiver which maximizes the SNR.

Hence optimal matched filter is simply you can say $h(t)$ equals well $P_1(t - \tau) - P_2(t - \tau)$ this is your optimal matched filter or basically which is basically square root of well you can square root of $\frac{2}{T} \cos(2\pi F_1(t - \tau)) - \sqrt{\frac{2}{T}} \cos(2\pi F_2(t - \tau))$ this is the optimal this is the optimal or this is the matched filter this is the optimal filter which maximizes the SNR.

Optimal filter at the SNR which maximizes the SNR at the receiver which maximizes the SNR. So, that is basically proportional to $P_1(t - \tau) - P_2(t - \tau)$ one can simply choose $P_1(t - \tau) - P_2(t - \tau)$

P_2 one can choose $P_1(t) - P_2(t)$ that is $P_1(t) - P_2(t)$ divided by 2 or basically simply $P_1(t) - P_2(t)$ therefore, the optimal matched filter will be $P_1(t - \tau) - P_2(t - \tau)$. So, that is basically your matched filter I am sorry I have to simply change this, this is not t , but rather $t - \tau$ similarly over here the optimal matched filter is $F_2(t - \tau)$. So, this is $P_1(t - \tau) - P_2(t - \tau)$. So, it is matched to $P_1(t) - P_2(t)$ which is $P_1(t - \tau) - P_2(t - \tau)$ that is the optimal matched filter.

So, employing the matched filter one can again carry out the analysis the receiver. So, employing this matched filter that we have derived correct employing this matched filter one can again carry out the analysis the rest of the analysis at the receiver derive what is the signal to noise power ratio and also what is the corresponding probability of bit error which we will do sub in the subsequent module.

Thank you very much.