

Spread Spectrum Communications and Jamming
Prof. Debarati Sen
G S Sanyal School of Telecommunications
Indian Institute of Technology, Kharagpur

Lecture - 29
Rejection of Narrowband Interference

Hello students. Today we will discuss about the Rejection Process and Rejection algorithms for Narrowband Interference cancellation in a direct sequence spectrum communication system. And today our topic of discussion will be the rejection of this narrowband interference by adaptive filtering. We will learn what is, how narrowband interference is so critical in the spread spectrum communication system, and then we will learn the structure of adaptive filtering, and try to understand how this adaptive filter can help us to reject the narrowband interference.

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Rejection of Narrowband Interference

- Narrowband interference presents a crucial problem for *spread-spectrum overlay systems*, which are systems that have been assigned a spectral band already occupied by narrowband communication systems.
- *Jamming against tactical spread-spectrum communications* is another instance of narrowband interference that may exceed the natural resistance of a practical spread-spectrum system, which has a limited processing gain.
- There are a wide variety of techniques that supplement the inherent ability of a direct-sequence system to reject narrowband interference.
- All of the techniques directly or indirectly exploit the spectral disparity between the narrowband interference and the wideband direct-sequence signal.

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So, we understood that spread spectrum communication is having a inherent processing gain, that comes because of spreading of the waveforms spreading of the band width or the signal, transmitting signal, and which helps which is directly related with the length of the code sequence utilize for spreading the transmitted signal, and finally, this is the processing gain that helps us also, to get a gain against a narrowband interference, and it helps us to have a cancellation, inherent cancellation property for this kind of communication system against a narrowband interferences.

But certain in practical implementation we have seen actually that the bandwidth is a scarce resource, is very scarce resource we understand, and that is why there are several communication systems, so, who are actually sharing the same almost same bandwidth and we call actually this systems are the overlays systems, and if some spread spectrum communication is a overlays system, where actually there exist narrowband communication system, a priori, then this spread spectrum communication we have a huge interference problem, from that narrowband interference.

Let us take an example mine is suppose there was a narrowband interference there was a narrowband communication system within the bandwidth, this, and we are planning to deploy a spread spectrum communication system, which also incorporates this narrowband interference zone. If this is the situation and these are very common example or very good example is ultra wide band technology. If you see the ultra wide technology we have a bandwidth of 7.1 gigahertz and it starts from 3.1 gigahertz to 10.6 gigahertz and within this, around 5giga hertz, you will see actually there is a Wi-Fi operation going on and then Wi-Fi is a narrowband signal, whereas, ultra wide band signals are have really wide band not only wide band they are the ultra wide band signals who need to be deployed using the incorporating the 5 gigahertz band.

So, now the signals who will be in this zone who will be within this 5 gigahertz zone ultra wide band signal who are inside this wide five gigahertz zone. If UWB is having a, option to use the spread spectrum communication system, and you if you use a spread spectrum communication then the signal who is close to the zone this 5 gigahertz, then we will have an effect severe effect from the Wi-Fi, and. So, Wi-Fi interference as the power transmission or power transmission level of the Wi-Fi signals are much higher compared to this ultra wideband signals, according to the FCC norms and the rules and then the intend ultra wideband signals will be completely buried under the interference of the Wi-Fi.

So, this is the one example that, where interference narrow band interference can be a very, say can have the very severe effect on the performance of the spread spectrum communication, and for such example, even if you are processing gain for this direct sequence spectrum is a pretty large, then also compared to the processing gain you are having and the power level of the transmission power level of the Wi-Fi and the transmission power reception receive power level of the Wi-Fi and receive power level

of the, UWB is having so big difference that the process is to meet up this interference level only by the processing gain, will be will be typically will typically very harder and also which requires very long length of the sequence and it will be finally, hard to implement in the in the hardware.

So, this is one example where the narrow band interference can actually is severe, can have the severe effect on the deployment of the other spread spectrum overlays system. And, so, we call this arrangement that overlays system because UWB is the over lay, over lay over the existing narrow band systems of the Wi-Fi. Another example is that jamming against the tactical spread spectrum communications, it is another instance where the narrowband interference matcan exceed the natural resistance power of a spread spectrum communication; that means, the power of a interference is much higher than the processing gain achievable by the spread spectrum communications system. And that is why in both the situation the example one and example two, you will have a severe effect from the narrow band interference on the intended spread spectrum communications.

So, there are wide verities of technique. So, it is important first of all to understand that cancellation of narrow band interference is a very very critical issue in the spread spectrum communication. We should not only derive on the processing gain and the capacity inherent capacity or spread spectrum communication, which is actually coming from the processing gain to cancel out the narrow band interference. We have to develop some technique to cancel the interference inside the receiver architecture.

In context of that there are wide verity of the techniques available that supplement the inherent ability of these interference cancellation (Refer Time: 07:07) interference cancellation, and frankly speaking all this techniques, usually they exploit the disparity between the spectral, spectral disparity between the narrow band signal the wide band signal under spread spectrum wideband signal. We will utilize that fact that we understand that the narrow band signal band width and the wide band signal band width are in spread spectrum signal band width there is much difference between the these two bandwidths.

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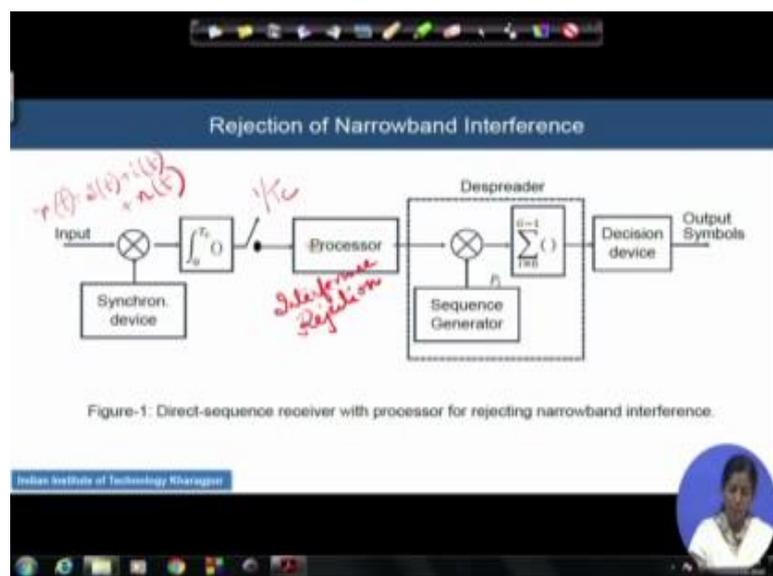
Rejection of Narrowband Interference

- The most useful methods can be classified as:
 - Time-domain adaptive filtering
 - Transform-domain processing
 - Nonlinear filtering or Code-aided techniques.
- The general form of a receiver that rejects narrowband interference and demodulates a direct-sequence signal with binary PSK is shown in Figure-1
- The processor, which follows the chip-rate sampling of the baseband signal, implements one of the rejection methods.

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So, we will try to exploit it to divide the rejection algorithm narrow band interference rejection algorithm. The most useful methods that discussed here is, the time domain adaptive filtering, the transform domain processing, the non-linear filtering or the code aided techniques. Out of that today we will discuss the time domain adaptive filtering, and remember the where this filter will seat in the receiver architecture we will see quickly in the next slide.

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First we understand this block diagram several times we have referred this block diagram to do lot of receiver performance study. And here we understand that the incoming signal $r(t)$ is consisting of our intended signal $s(t)$ plus the, $I(t)$ the interference signal and the noise portion $n(t)$.

So, after the synchronization after the synchronization device is multiplied with this incoming signal, you are down to the base band where the match filter is operated and after that the sampler is sampling at the chip rate. So, the sampling is going at the rate of T_c , and here is the processor here we have introduced a new block which were actually the interference rejection block. This processor is basically due to this interference rejection. And the output of this interference rejection block is fair to the despreader and the decision device is they are connected on.

Earlier we did not show this block, we have done the whole receiver architecture study receiver architecture by directly connecting the output of the sampler, to the input of the despreader. So, here comes the interference rejection filter.

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Time-Domain Adaptive Filtering

- A time-domain adaptive filter for interference suppression processes the baseband sample values of a received signal to adaptively estimate the interference.
- This estimate is subtracted from the sample values, thereby canceling the interference.
- The adaptive filter is primarily a predictive system that exploits the inherent predictability of a narrowband signal to form an accurate replica of it for the subtraction.
- Since the wideband desired signal is largely unpredictable, it does not significantly impede the prediction of a narrowband signal.
- When adaptive filtering is used, the processor in Figure-1 has the form of Figure-2. The adaptive filter may be a one-sided or two-sided transversal filter.

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So, if we have understood that this is time. So, that block that interference rejection block either can be the time domain adaptive filter, or it can be the transform domain processor, or it can be a non-linear filter or code aided technique in deployed in that. The processor which gives a which takes actually the chip rate sample output from the sampler, basically it get is getting the base band signal at an input and implement sides

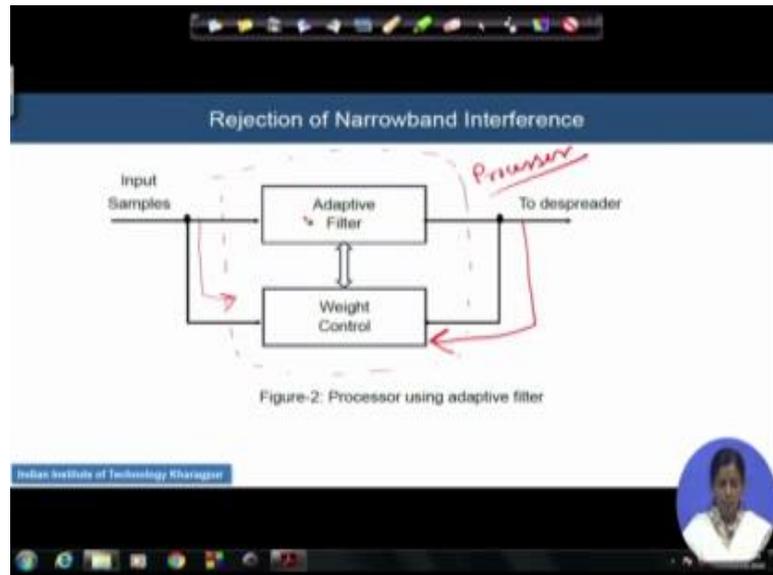
one of these rejection mechanism, either time domain by time domain adaptive filtering or by transform domain processing or by non-linear filtering.

We will refer then figure again and again. So, now, it is a time to enter inside the time domain and adaptive filtering. Time domain adaptive filter for the interference separation process, it takes the base band sampled values as I have mentioned and shown in the earlier diagram and it adaptively estimate the interference when the name goes. The adaptive filter it is adaptively estimating the interference from the incoming signal, and once the estimate is done it subtracts the interference is estimated interference from the received signal and here by cancelling the interference. Basically if you (Refer Time: 11:12) see the structure of this filter it is basically a predictive system, and that system if have exploits the inherent predictability in a narrow band signal, by exploiting the code relation property of the signal, and it forms actually the accurate replica of this narrow band signal. So, that finally, it can be subtracted from the receiving signal.

So, the in the received signal $r(t)$ you are getting the combined spectrum of a narrow band plus a wide band signal. Narrow band signal is coming from the interference and a wide band signal is a intended direct sequence spread spectrum signal, and noise is definitely wide and it is having the total signal bandwidth hence per our assumption. So, this filter will have the effect only on the narrow band signal it is not having any effect on the noise spectrum and, definitely it will not have any other effect on the intended direct sequence spectrum also.

Since wide band desired signal it is highly unpredictable. So, it does not it significantly impaired on the prediction on these narrow bands. So, it is completely unpredictable one that why. So, if I use that adaptive filter is the then a figure that we have shown in the last, here the filter that I have told earlier that it will be replacing their processor.

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Basically will have the form will look like this. So, this, the whole processor to me, now the processor will be, this whole, in the figure one, the processor means here the filter as well as the weight control.

Please mention please notice here that, the filter is taking an input not only from the input samples, he also taking from the precious filtered output do the prediction, and to control the all weight to predict the weight to be adjusted for the newly entered received symbols. So, the filter is a taking the present as well as the part values.

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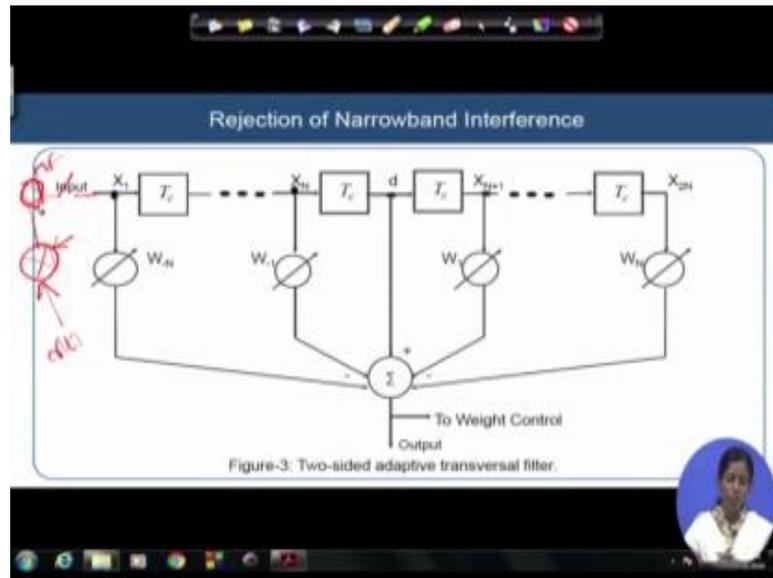
The slide is titled "Time-Domain Adaptive Filtering" and contains a list of five bullet points. The text is as follows:

- The two-sided adaptive transversal filter multiplies each tap output by a weight except for the central tap output, as diagrammed in Figure-3.
- This filter is an interpolator in that it uses both past and future samples to estimate the value to be subtracted.
- The two-sided filter provides a better performance than the one-sided filter, which is a predictor.
- The adaptive algorithm of the weight-control mechanism is designed to adjust the weights so that the power in the filter output is minimized.
- The direct-sequence components of the tap outputs, which are delayed by integer multiples of a chip duration, are largely uncorrelated with each other, but the narrowband interference components are strongly correlated.
- As a result, the adaptive algorithm causes the interference cancellation in the filter output, but the direct-sequence signal is largely unaffected.

There are red handwritten annotations on the slide, including "Predictor" and "Interpolater" with arrows pointing to the relevant terms in the text.

And then it is processing gain for the filter for the processing gain for the interference cancellation option. Remember if we are having the selective filters can have actually several kind of the architecture if it is two sided and then we call it is an it is symmetric, is that sense we will call it two sided a transversal filter and the structure of such transversal filter will look like.

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This like figure three where there is a central point where there is a central point and the filter is a having mirror image on both the side, and this is a transversal filter the way you see the vectors one is also actually reflected as weight vectors minus one. So, you are having coefficients weight coefficients going from minus n to plus n, I mean two n two n number of the weights and you are having and you will be dealing with, we will come in details later on.

So, if it is one sided structure. So, remember it does not have a central point if it is one sided structure then we will call that filter structure as a predictor and this filter is basically an interpolator. So, it uses both as I told it uses showed that it will uses both the past and the future samples to estimate the value to be subtracted from the incoming signal. So, this adaptive algorithm that will be implemented inside, that adaptive algorithm will be basically to give the adaptive weight. So, it is a, algorithm is basically a weight control mechanisms this nothing, but an weight control mechanisms this is designed to adjust the weights in such a way, is very important that you have designed

this algorithm this weight control mechanisms in such a way that the total power of the filter should be minimized.

So, the target the objective function over which actually the algorithm runs is to minimize the filter output power. The direct sequence components of this tap outputs which are delayed by the integer multiple of this chip duration will be largely uncorrelated as we understand. So, see the direct sequence components. So, filter means we are having the several taps and, the direct sequence components that are coming as an input, you we actually basically inside the filter. We are having the direct sequence components and we are having the narrow band interfering components. See these direct band components who are delayed by chip. So, one chip to the next chip inside the direct sequence components you will not find any correlation to them.

The samples are uncorrelated to each other, but the narrow band interference the narrow band signal they are able to see here with the chat with the typical chip delays, then you will be able to see that they are largely uncorrelated with each other, but the narrow band interference components, as they are heavily uncorrelated. So, filter will be work on this fact that he will try to find out the correlated samples and subtract those correlated samples from the uncorrelated from the total one to find out the; so that you are left with the correlated uncorrelated samples.

So, filtered targets to estimate the correlated samples and then subtract it from the total one received samples to get with to stay with the, uncorrelated samples. So, that is a main idea, and. So, hence the interference is cancelled by estimating the correlated sample and subtracting it from the total input signal, and by that method actually once we are doing this the direct sequence signal we is the desired one and hence this become actually this remains largely unaffected in the whole analysis.

This is the structure all the taps are the filter taps are delayed by the chip all the filtered taps are have are (Refer Time: 18:23) the output of each and every filter tap they are multiplied with some weight we are having a total $2n$ number of the taps and $2n$ number of the weight vectors and they are summed up such a way that, they are summed up such a way. So, that you get the total output power of the filter such a way that the total power of the filtered output will be minimized. So, the filter will have how many taps now that

you are having n plus n two n taps plus one. So, two n plus 1 number of the taps where the central point is never is never controlled this is not weight controlled.

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Time-Domain Adaptive Filtering

- An adaptive filter with $2N + 1$ taps and $2N$ weights, as shown in Figure-3, has input vector at iteration k given by

$$\mathbf{x}(k) = [x_1(k) \ x_2(k) \ \dots \ x_{2N}(k)]^T$$
 (1-1) *real valued*
- and weight vector

$$\mathbf{W}(k) = [W_{-N}(k) \ W_{-N+1}(k) \ \dots \ W_{-1}(k) \ W_1(k) \ \dots \ W_N(k)]^T$$
 (1-2) *real valued*

where T denotes the transpose and the central tap output, which is denoted by d has been excluded from \mathbf{x} .

- Since coherent demodulation produces real valued inputs to the adaptive filter, $\mathbf{x}(k)$ and $\mathbf{W}(k)$ are assumed to have realvalued components.
- The symmetric correlation matrix of \mathbf{x} is defined as

$$R_{xx} = E[\mathbf{x} \mathbf{x}^T]$$
- The cross-correlation vector is defined as

$$R_{xd} = E[\mathbf{x} \ d]$$
 E[xd]

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And if I write the input vector, to that filter, to be $\mathbf{x}(k)$, where $\mathbf{x}(k)$ will be having $2n$ number of the samples. The corresponding weight vector will move from w of minus n k say $2w - n$ k, and where t transpose nothing, but the transpose in both the cases, and the central tap output, that is denoted by the d , in the last, figure in the last figure, in the last figure we have denoted the central tap as a d . It is excluded, this d is excluded from our expression $\mathbf{x}(k)$. And since this a coherent demodulation is going on a priory with its architecture.

I mean to say that at the input of this filter we have seen that this is coming from, this input to the filter is basically coming from the sampler and before the sampler some synchronous modulation, the match filter was there, and a priory match filter we had the synchronous the synchronizer who were actually the $r(t)$ was fed, and here our t to the power $j 2 \pi f_c t$, that frequency was coming to bring it down. So, if the synchronous there is the synchronous modulation, if I think that some synchronous modulation is going on a priory, and then we can definitely think of the fact that, if the synchronous modulation is going on and a priory, then the symmetric, then all the, n components that the $\mathbf{x}(k)$ and $\mathbf{w}(k)$, that we are concerned here, they are all the real value components because at the

end of the modulation synchronous modulation you will not get any, you will not, you will be left only with the real valued complaint come real valued samples.

And both are the both the samples at you are dealing with here that is why actually we are keeping it considering that all are the $x(k)$ and all the correlation functions everything they are of real valued only. The symmetric correlation matrix if we are now defining by $r \times r$, that will be given specially by the expected value of x and x transpose, and the cross correlation vector will be definitely given by the expected values of the x into with the d , because this d is not included inside this vector the input vector. That is the definition of ours one is the symmetric correlation matrix, another is cross correlation vector.

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Time-Domain Adaptive Filtering

- According to the Weiner-Hopf equation, the optimal weight vector is

$$W_0 = R_{xx}^{-1} R_{xd} \quad (1-3)$$
- The least-mean-square (LMS) algorithm computes the weight vector at iteration k as

$$W(k) = W(k-1) + \mu e_k x(k) \quad (1-4)$$

where

- $e_k = d - y_k$ is the estimation error,
- $y_k = W^T(k)x(k)$ is the filter output,
- μ is the adaptation constant, which controls the rate of convergence of the algorithm.

- The output of the adaptive filter is applied to the despreader.
- Under certain conditions, the mean weight vector converges to W_0 after a number of iterations of the adaptive algorithm.

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Remember one thing; this optimal weight vector the target is to find out the optimal weight vector, what would be our optimal value of the weight.

So, the optimal weight vector will be such the way the finding of this optimal weight vector will be, simply found out by the multiplication of this $r \times r$ inverse with the $r \times d$; that means, the correlation of the; that means, the multiplication of this two matrices the symmetric correlation as well as the cross correlation vector and where actually we are trying to find out. And this relation that it will be give the optimal weight, that there is a fundamental proof of it and we got we get this proof by this Weinerhopf equation.

And once actually this optimal weight vector is found the least mean square error algorithm by applying the least mean square error algorithm, we can form recurring relation between the k th value of the weight with the k minus 1 of the last instant value with this weight, which is basically the current value of the weight is the last value of the weight last factor given by the μ epsac k and x_k , where remember this epsac k is basically the estimation error vector which is nothing, but y where y_k is my output of the filter.

So, filter output and the central portion, that difference you are trying to trying to find out, and this is the estimation vector estimation error, and last this is the estimation error then you are trying to find out this $w^T x_k$ and x_k , this is the filtered output given by. So, μ is that adaptation constant, which controls the rate of the conversation of the algorithm and it is a selected. So, multiply by this adaptation constant your error vector and the current, value of the input adaptive input vector, this is this is controlling the choice of this second portion of the equation and finally, adjusting adding up that factor, with the old weight vector you are getting the current weight vector.

So, the way this is the this least mean square the weight vector is done such a way that the, you are choosing the least value of the mean square errors, of the mean square error of the estimation, and then the output of this adaptive filter, that you are getting with by adjusting this one. I mean adjusting this new weight vectors we do the filtration of the incoming signal and then it will be applied the filtered signal to the despreaders. So, remember under certain condition it will happen that this mean weight vector the mean weight vector converges to w zero of certain number of the iterations of the adaptive filter.

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Time-Domain Adaptive Filtering

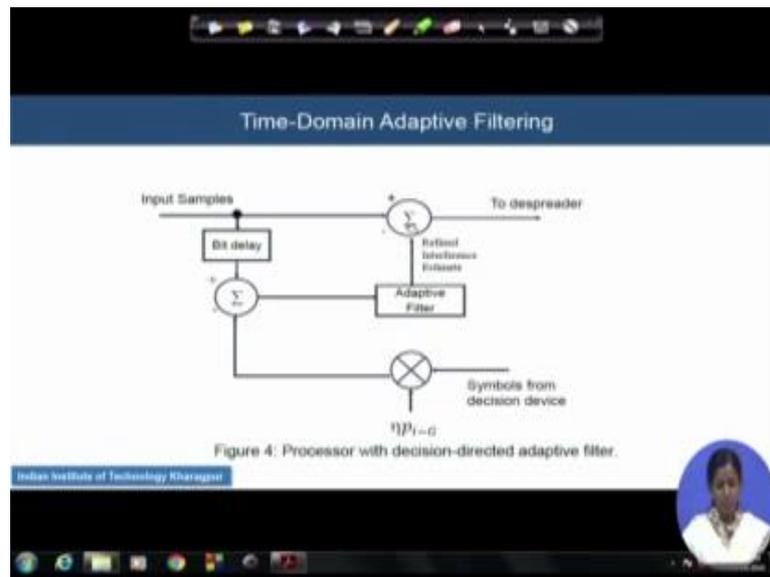
- If it is assumed that $\mathbf{W} = \mathbf{W}_0$, then a straightforward analysis indicates that the adaptive transversal filter provides a substantial suppression of narrowband interference.
- Although the interference suppression increases with the number of taps, it is always incomplete if the interference has a nonzero bandwidth.
 - Because a finite-impulse-response filter can only place a finite number of zeros in the frequency domain.
- The adaptive transversal filter is inhibited by the presence of direct-sequence components in the filter input vector $\mathbf{x}(k)$.
- These components can be suppressed by using decision-directed feedback, as shown in Figure-4.

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So, if it is assured that w will be equal to w_0 , I mean you are converging then a straightforward analysis indicate that the adaptive transversal filter, will provide the substantial suppression of this narrow band interference. So, this is the target where actually you have to reach by adjusting the μ and checking the value of the ϵ such that actually you can suppress an interference reasonably. This interference suppression though increases if we increase the number of the taps, but sometimes it always incomplete also. If the interference as a non zero bandwidth because we understand that a finite impulse response filter can only place a finite number of the zeros in the frequency domain.

The adverse the adaptive transversal filter is inhibited by the presence of the direct sequence components, in the input vector $\mathbf{x}(k)$, and those components also can be suppressed by using a decision directed feedback. This decision directed feedback is shown in the next slide.

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So, what we are trying to do here is, the, decision directed adaptive filter is a next stage of whatever we have seen in the figure number three. Here the input samples with the big delays they are separate feed in the adder and basically the symbols that were already, this that are already obtained at the output of the decision device our decision device they already brought and the with the delay of actually with the weight adjusted with them by the η , and, the and also the spreading also the spreading code which with the delay of the processing gain g or with the delay of the length of the actual code spreading code utilized.

They are actually regenerated suppose this is another spread signal that you have got now, and input samples are also obtained here, and if we are having some addition share and then that signal is fed to the adaptive filter. So, one level of the rejection of the interference can is possible, and the final refinement is done by the adaptive filter itself and again, this guy is fed to the output of this error is fed to the despreader.

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Time-Domain Adaptive Filtering

- Previously detected symbols remodulate the spreading sequence delayed by G chips (long sequence) or one period of the spreading sequence (short sequence).
- After an amplitude compensation by a factor η , the resulting sequence provides estimates of the direct-sequence components of previous input samples.
- A subtraction then provides estimated sample values of the interference plus noise that are largely free of direct-sequence contamination.
- These samples are then applied to an adaptive transversal filter that has the form of Figure 2&3 except that it has no central tap.
- The transversal filter output consists of refined interference estimates that are subtracted from the input samples to produce samples that have relatively small interference components.

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So, in this cases the previously detected symbols which are remodulated, once again the which remodulate, the which remodulate, which remodulate, the spreading sequence once again which are delayed by this g , that we have shown earlier in the last figure. They are used here to help out the filtration process again, but remember the amplitude compensation is done by a factor of η , and the resulting sequence provides then the estimate of this direct sequence component, or the component of the sequence itself actually that represent may be inside the input samples.

So, this subtraction can of the can help you actually to get rid of the direct sequence combination and, the samples are then applied to the transversal filter, and that transversal filter will the structure of this transversal filter, that we have seen in this figure, this adaptive filter is basically having the structure like this bigger one. And, so, this samples are then apply to this adaptive filter when we are will apply structure, the output will be, the output of this will be fed to an adaptive filter, but this adaptive filter is having exactly the structure like except the central tap. And then the transversal filter output will be refined interference estimate that are subtracted from the incoming signal.

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Time-Domain Adaptive Filtering

- An erroneous symbol from the decision device causes an enhanced direct-sequence component in samples applied to the transversal filter, and error propagation is possible.
- However, for moderate values of the signal-to-interference ratio at the input, the performance is not degraded significantly.
- Adaptive filtering is only effective after the convergence of the adaptive algorithm, which may not be able to track time-varying interference.
- In contrast, transform-domain processing suppresses interference almost instantaneously.

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Now, if we are having some erroneous symbol at the output of the detector. So, definitely, means that if that is used to remodulate the, locally generated direct sequence sequence direct sequence, then definitely and if it is used to reject the effect of the direct sequence is the incoming signal then this erroneous signal will actually escalate the problem of the interference in the incoming in the escalate the process of of the escalate the interference amount and it will disturb the total process of the interference rejection in the adaptive filter.

So, we need to be very careful of using the symbol already detected symbols. And the moreover if you are having the moderate values of the signal to interference ratio in high $s_i r$ zone, when the signal interference ratio is moderate then we can realize that this process will what at mean tap in the decision directed adaptive filter will really be very very helpful in addition to the our original adaptive structure that we have discussed. But adaptive filtering is only effective after the convergence of the adaptive algorithm, which may not be able to track the time varying interference definitely we should be remembering that fact, that it is a hard actually to track any time varying interference. So, it will be effective after the only the convergence of the adaptive algorithm, and the in contrast actually the other one which we are understand as a transform domain processing that we will discuss the that we may discuss actually sometimes later, that will suppress the interference on most instantaneously.

So, if I compare the pros and cons of this adaptive filter. So, this is a very common and fast stage of the filter and most popular also, in the receiver architecture to they have to implement also, and to utilize and to get rid of the additive interference rather than interference in the spread spectrum communication systems. And it is a supplement of the existing inherent capacity of the direct sequence system, to reject the narrow band interferences by means of the code gain, and which is also call the processing by means of the processing gain which is actually directly related with the length of the codes.

But this filter remember that adaptive filter is, not able to track the time varying interference in that sense and if we wish to instantaneously suppresses the interference, then adaptive filter will not be a good choice.