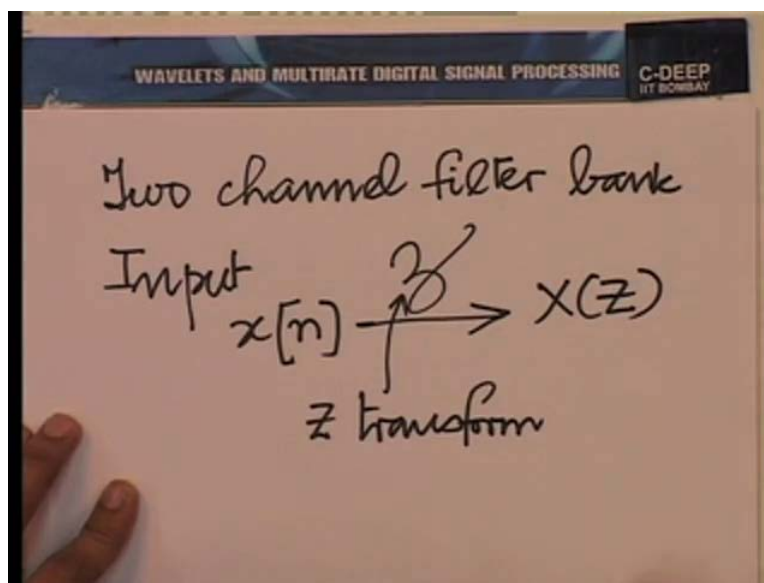


Advanced Digital Signal Processing- Wavelets and Multirate
Prof. V. M. Gadre
Department of Electrical Engineering
Indian Institute of Technology, Bombay

Lecture No. # 12
Perfect Reconstruction: Conjugate Quadrature

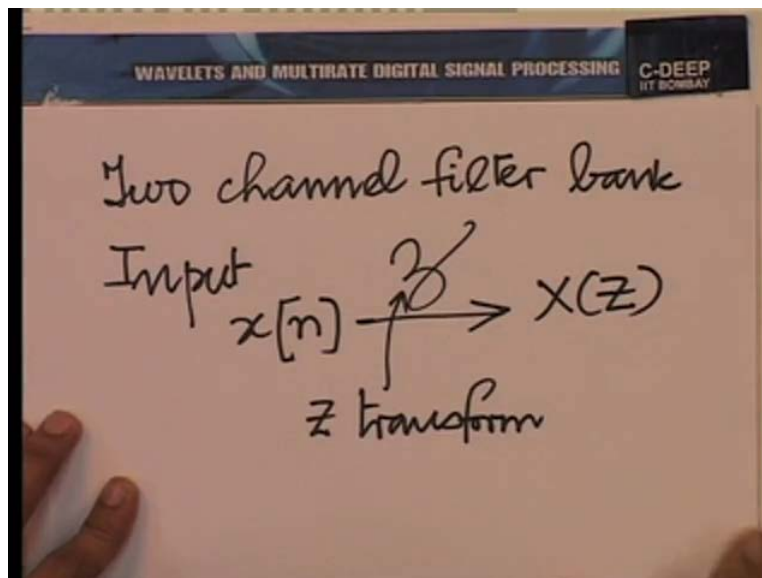
A very warm welcome to the twelfth lecture; on the subject of Wavelets and Multirate Digital Signal Processing. Let us spend a minute on what we had done in the previous lecture. We had looked at the two band filter bank in the previous lecture. And, we had written down a set of conditions based on what happens when you go past a down sampler and an up sampler in the z domain. So, let us put down the conditions once again or let us in effect, put down the relation between the input and the output in the z domain; where it is valid to use the z domain, which is true in many circumstances.

(Refer Slide Time: 01:29)



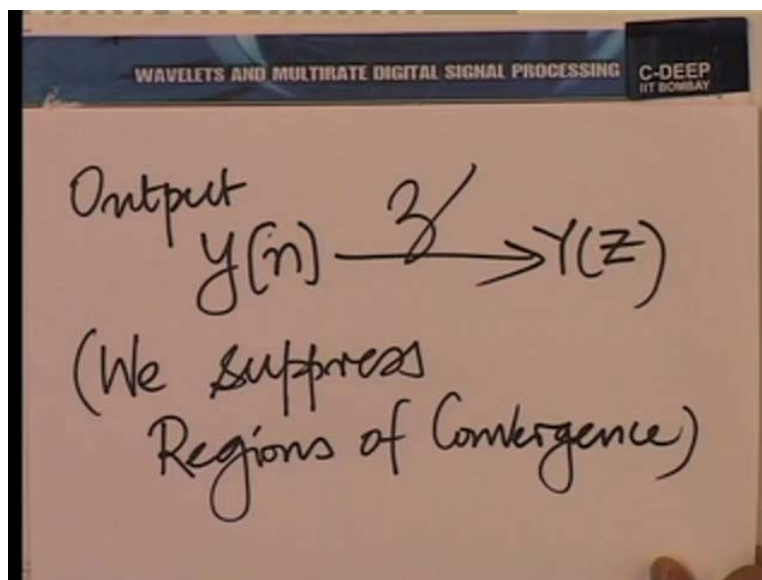
So, let us summarize what we had derived the last time. We said 2 at two channel filter bank, input $x[n]$; it is z transform.

(Refer Slide Time: 02:07)



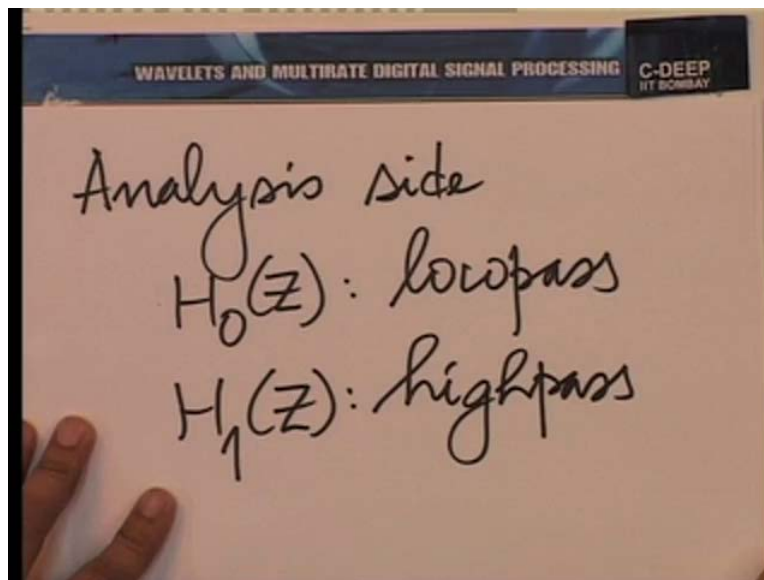
Now, use this script z to denote the z transform, capital X z.

(Refer Slide Time: 02:07)



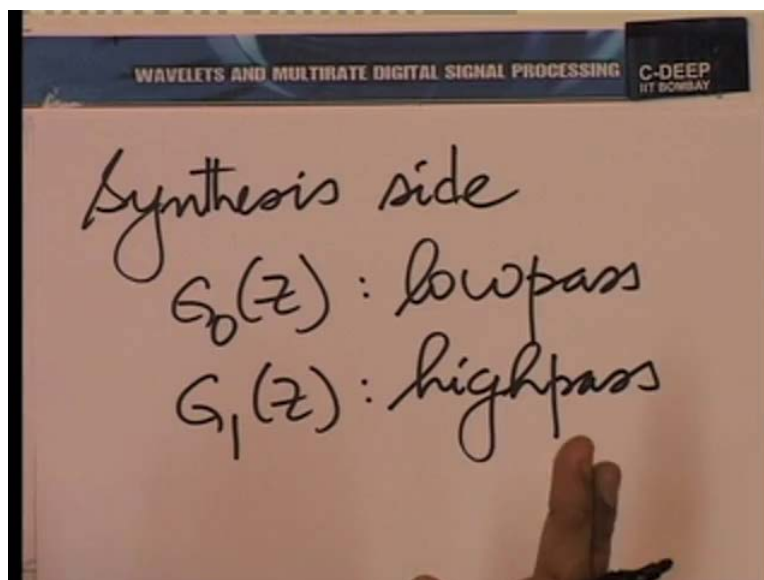
Output $y[n]$; which z transform capital Y z. Now, incidentally we suppress the regions of convergence. So, we do not explicitly mention the regions of convergence.

(Refer Slide Time: 02:43)



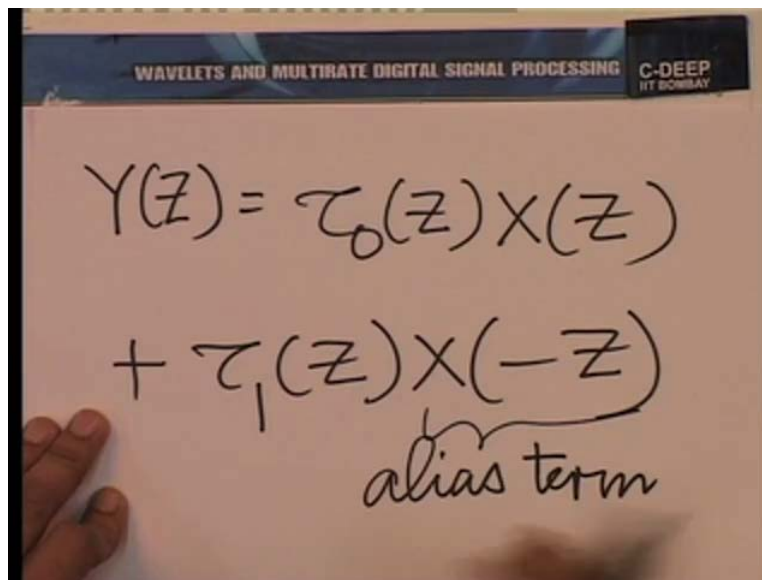
Analysis side: $H_0(z)$, so-called low pass filter and $H_1(z)$, the high pass filter.

(Refer Slide Time: 03:09)



Synthesis side: $G_0(z)$, low pass filter; $G_1(z)$, high pass filter. So, this is the circumstance. Of course, you know where the down samplers and up samplers are. The relation between $Y(z)$ and $X(z)$ is as follows.

(Refer Slide Time: 03:41)



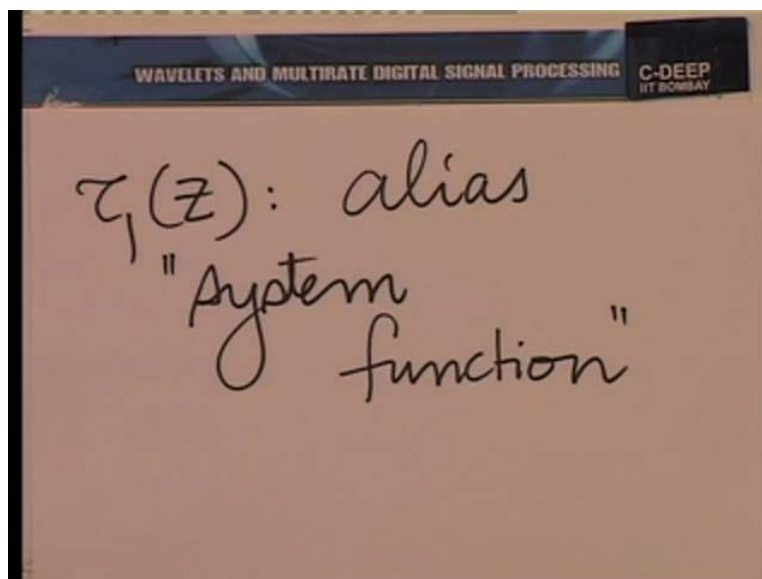
WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

$$Y(z) = \tau_0(z)X(z) + \tau_1(z)X(-z)$$

alias term

We have $Y(z)$ is $\tau_0(z)X(z)$ plus $\tau_1(z)X(-z)$. And, we call that, we call this the alias term.

(Refer Slide Time: 04:13)

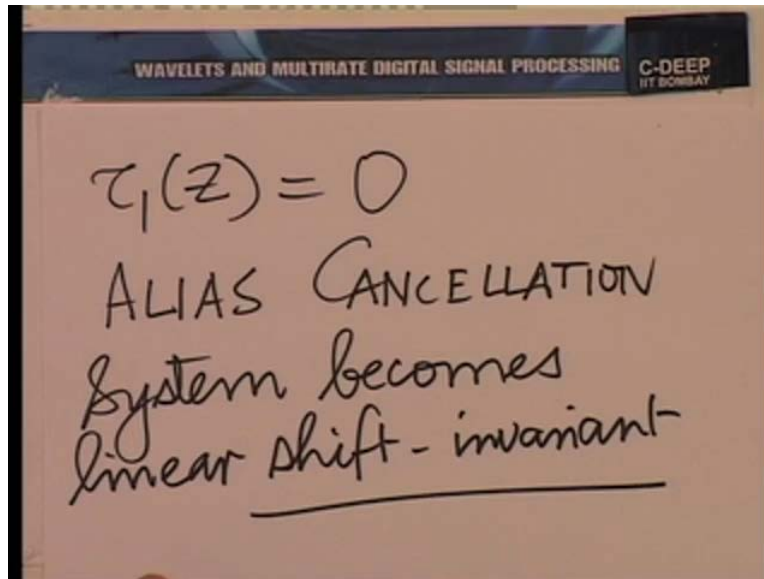


WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

$\tau_1(z)$: alias
"system function"

And therefore, $\tau_1(z)$ was called the alias "system function". And, put "system function" in inverted commas and it is synthesized here.

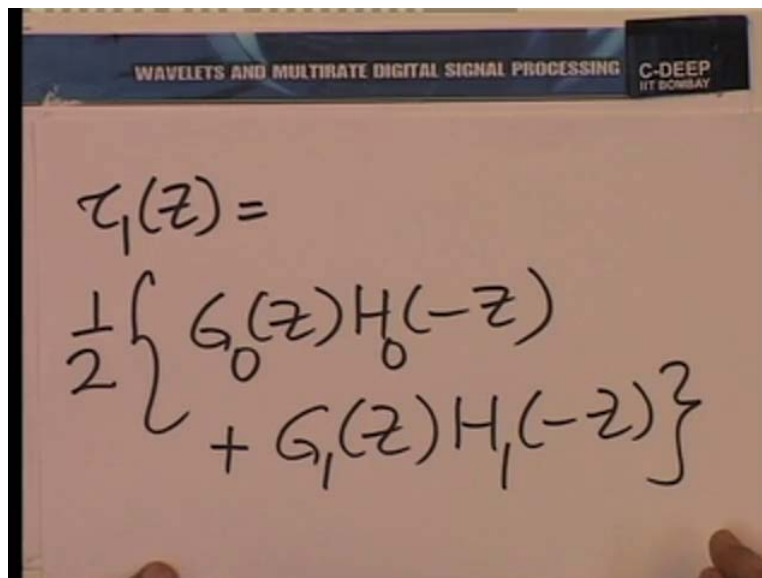
(Refer Slide Time: 05:02)



You know, when there is an alias term, the words “system function” is actually a misnomer. One should not use the term system function because the system is not linear and shift-invariant. On the other hand, if $\tau_1(z)$ is 0 the system becomes linear and shift-invariant.

So, this is something that we must now take note of. $\tau_1(z)$ equal to 0, is essentially what is called the condition for alias cancellation. And, we note the system becomes linear and shift-invariant. Now, we have to look at one possibility under which $\tau_1(z)$ could be zero.

(Refer Slide Time: 05:59)

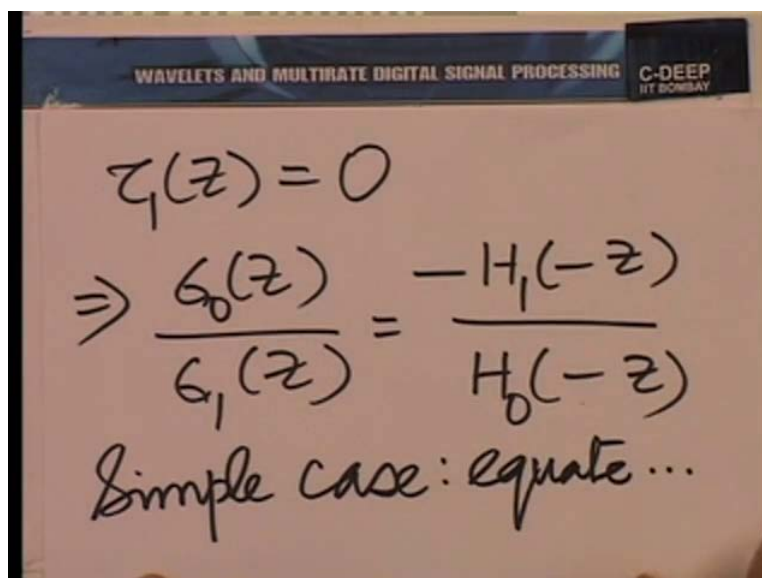


WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

$$\tau_1(z) = \frac{1}{2} \left\{ G_0(z)H_0(-z) + G_1(z)H_1(-z) \right\}$$

And, we had said the most general possibility can be accommodated by what is called a cancelling term Rz . let us put that down again. We said that $\tau_1 z$ is essentially the function or the expression half $G_0 z H_0$ minus z plus $G_1 z H_1$ minus z .

(Refer Slide Time: 06:21)



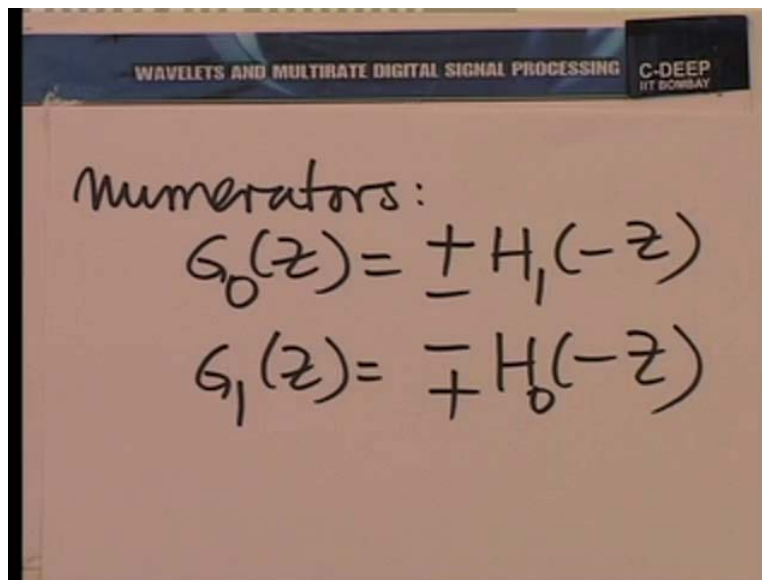
WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

$$\tau_1(z) = 0$$
$$\Rightarrow \frac{G_0(z)}{G_1(z)} = \frac{-H_1(-z)}{H_0(-z)}$$

Simple case: equate ...

And, $\tau_1 z$ equal to 0 means essentially that $G_0 z$ by $G_1 z$ must be minus H_1 of minus z divided by H_0 of minus z . And, the simple case is, equate the numerator and denominator.

(Refer Slide Time: 06:56)



Numerators:

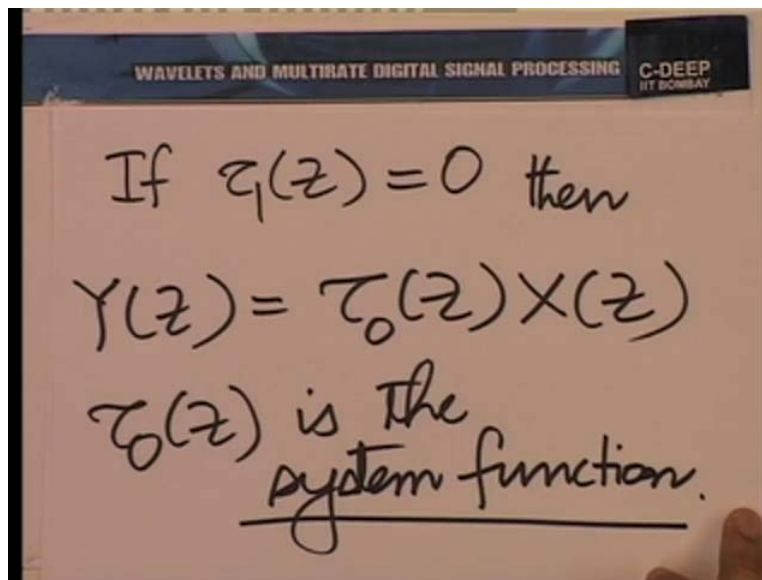
$$G_0(z) = \pm H_1(-z)$$
$$G_1(z) = \mp H_0(-z)$$

So, **equate** if you equate the numerator you get $G_0(z)$ is plus minus $H_1(-z)$ and $G_1(z)$ is correspondingly minus or respectively plus $H_0(-z)$. We also interpreted these expressions in the ideal case. If $H_0(z)$, for example, was the ideal low pass filter, then this would become an ideal high pass filter as we expect. And, if H_1 is the ideal high pass filter, this would become ideal low pass filter. All with the cut-off of $\pi/2$.

So, in that sense, we have a **neat** interpretation for the ideal case. Even if the filter is not ideal, we have a reasonable interpretation in the sense that **it** could always take this to be a non-ideal low pass filter with the cut-off of $\pi/2$. And, this would again become a reasonably close high pass filter with cut-off $\pi/2$ and vice versa with this. If this is high pass filter with cut-off $\pi/2$, then this becomes reasonable low pass filter with cut-off $\pi/2$, whatever it be.

Let us now, of course consider the second condition. You see, this is alias cancellation. With alias cancellation, we are assuming that there is linear linearity and shift-invariants in the system. There is the linear shift-invariant system there. What is the system function there?

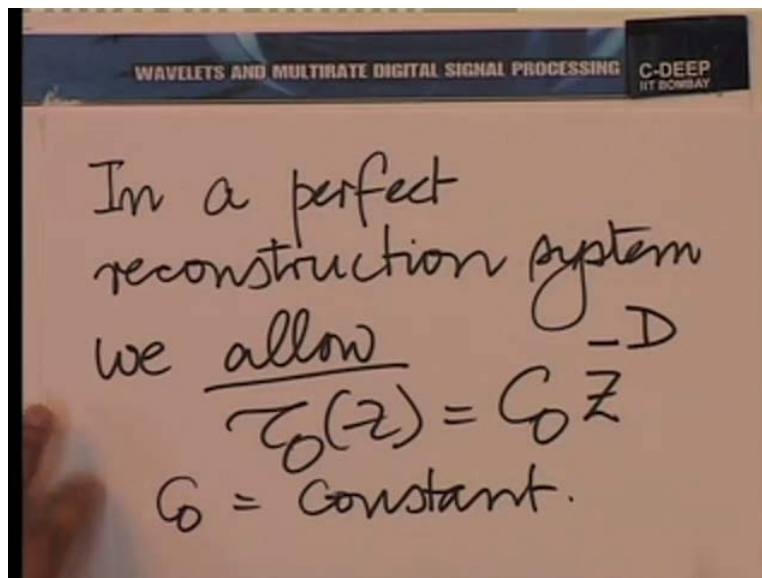
(Refer Slide Time: 08:40)



If $\tau_1(z)$ is equal to 0, then you have $Y(z)$ is equal to $\tau_0(z)X(z)$. And, we have seen effectively that $\tau_0(z)$ is the system function in the true sense. So, this is an alias system with system function $\tau_0(z)$. Now, you see one of the things that one needs to worry about is what should $\tau_0(z)$ be. $\tau_0(z)$ is also modifying experience for the input. And, ultimately we want decomposition and reconstruction. So, in reconstruction, we want exist to be almost same as $Y(z)$. If not quite, if not identity, at least there must be tolerable changes.

What are these tolerable changes that we can allow? Or, more appropriately what can we and what should we tolerate here. That is what we need to think about. What can we and what should we tolerate. Well, if you are talking about time systems, then we have to tolerate a delay. First, let me explain this intuitively and then let me put it down mathematically. The filter does some processing. It takes some time to process at the analysis side or the synthesis side. Now, you need finite time to process at analysis side. You need finite time to reconstruct at the synthesis side.

(Refer Slide Time: 11:48)



So, if you want to know other change between the input and the output, at least you have to accept the change of a delay to allow for some time to process. So, if we do not want any other change, at least we should allow for a term of the form z to the power minus D in $\tau_0(z)$. The other thing that we do not mind allowing is an overall multiplicative constant. After all, we do not mind if the whole input sequences multiplied by some constant c because we can always multiply by $1/c$ at the output. It is a simple operation to do; a simple amplifier or attenuator. That is not very difficult to do. So, we do not mind if the whole alias system that we had here after alias cancellation multiplies by a constant and delays by D .

And, that is exactly what we shall now put down mathematically. Assume you are saying in a perfect reconstruction system, we allow, we should say we allow $\tau_0(z)$ to be the form some constant times z raised to the power minus D ; is a constant.

(Refer Slide Time: 12:21)

WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

Ideally we would have:

$$\tau_0(z) = 1 \forall z$$

Ideally, we would have **write** $\tau_0(z)$ is equal to 1 for all z , essentially an identity system. Now, that would, as I explained before make the system non-constant.

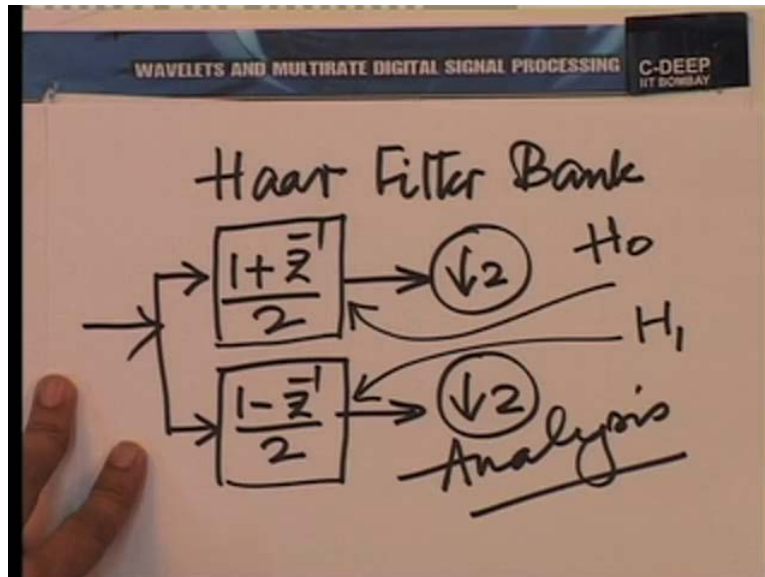
(Refer Slide Time: 12:48)

WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

z^{-D} (D positive)
is "allowed"
because of causality

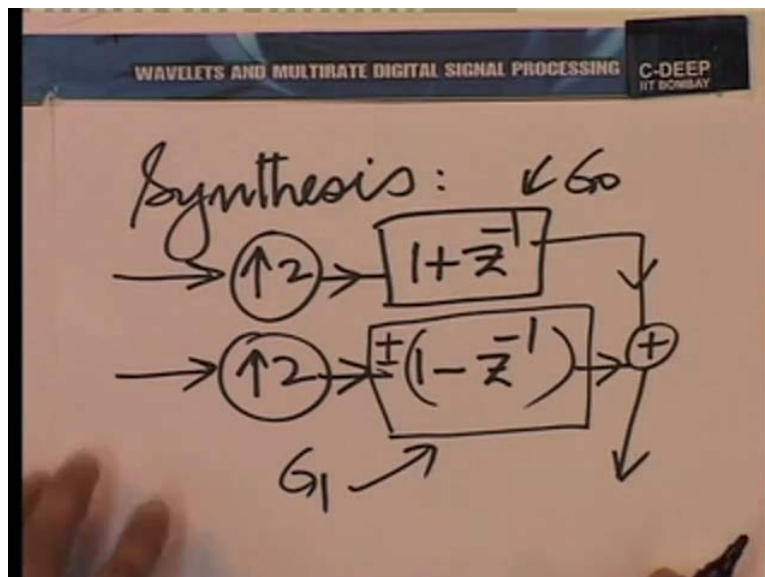
So, this allowance of the z raised to the power minus D , of course **of course** D is positive here; is allowed because of causality. Now, we will take the example again. as I said of the Haar MRA and the filter bank corresponding to the Haar MRA.

(Refer Slide Time: 13:44)



once again we will see, if we understand the haar case, we understand lot of things at once. So, next put down the filters for the Haar case. So, in the haar case, we had the following filters analysis.

(Refer Slide Time: 14:38)



So, this is H_0 and this is H_1 . Synthesis: You know, you remember, on the synthesis side, at that time **if z will** allow for a plus minus ambiguity here. Let us keep the ambiguity **and you will see** why the ambiguity is needed. This is G_0 and this is G_1 .

(Refer Slide Time: 15:25)

WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

$$\tau_1(z) = \frac{1}{2} \left\{ G_0(z)H_0(-z) + G_1(z)H_1(-z) \right\}$$

Let us try **to 1** $\tau_1 z$ here, $\tau_1 z$ by definition is, of course $G_0 z H_0$ minus z plus $G_1 z H_1$ minus z .

(Refer Slide Time: 15:54)

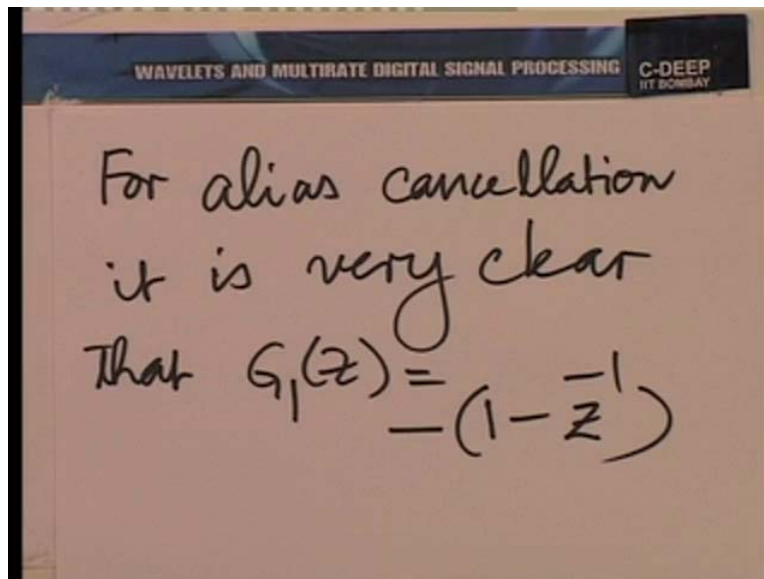
WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

$$\text{RHS} = \frac{1}{2} \left\{ (1+z^{-1}) \left(\frac{1-z^{-1}}{2} \right) + (1-z^{-1}) \left(\frac{1+z^{-1}}{2} \right) \right\}$$

We want = 0

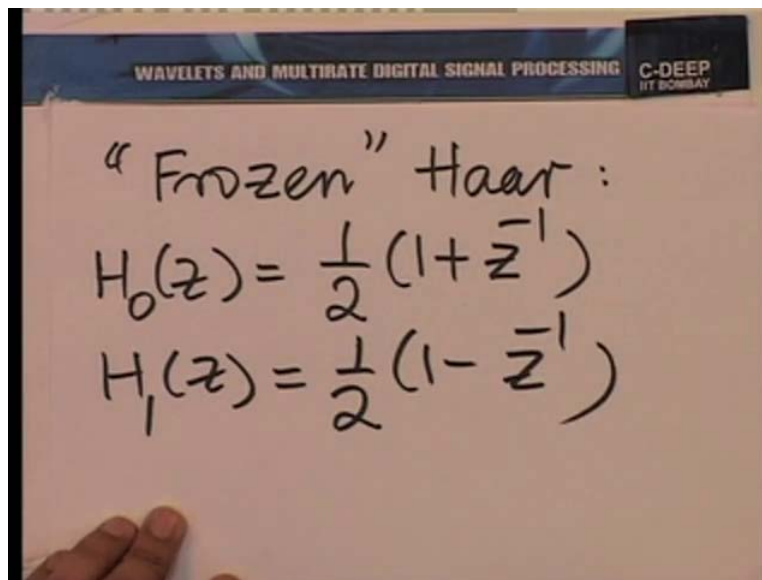
And, with that definition of G_0 , H_0 , G_1 and H_1 , we have the right hand side becoming simply half G_0 is $1 + z^{-1}$. H_0 minus z is $1 - z^{-1}$ by 2. Now, here we have a plus minus ambiguity. G_1 z is, of course $1 - z^{-1}$ and H_1 minus z is $1 + z^{-1}$ by 2. Now, you know, when you look at this carefully, you noticed why we want this ambiguity there. We want this to become zero. And, therefore it is obvious that the minus sign should be chosen, the plus sign will not give us the zero.

(Refer Slide Time: 16:56)



So, for alias cancellation, it is very clear that G_1 z must be equal to minus $1 - z^{-1}$ and not plus.

(Refer Slide Time: 17:31)



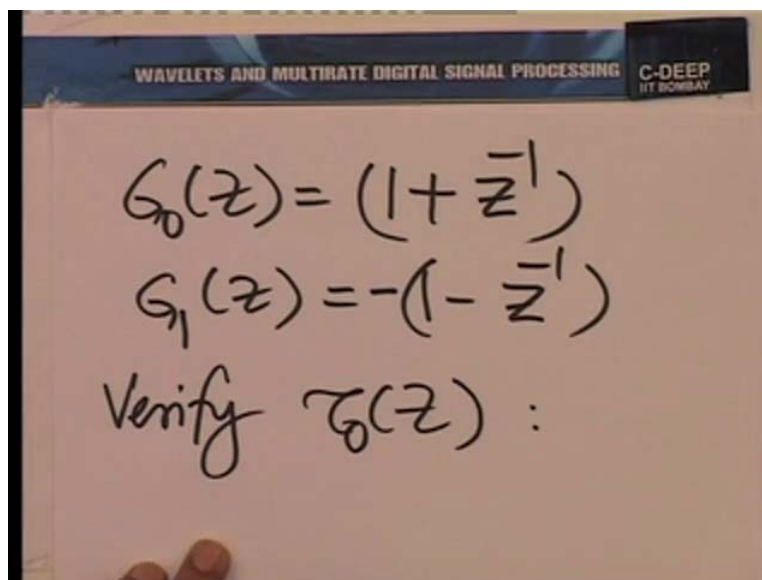
WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

"Frozen" Haar:

$$H_0(z) = \frac{1}{2} (1 + z^{-1})$$
$$H_1(z) = \frac{1}{2} (1 - z^{-1})$$

So, therefore, now let us freeze our G_0 , G_1 , H_0 and H_1 for the Haar case.

(Refer Slide Time: 17:59)



WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

$$G_0(z) = (1 + z^{-1})$$
$$G_1(z) = -(1 - z^{-1})$$

Verify $\tau_0(z)$:

And, let us verify the perfect reconstruction condition or verify $\tau_0(z)$.

(Refer Slide Time: 18:34)

$$\begin{aligned} \tau_0(z) &= \\ & \frac{1}{2} \left\{ G_0(z)H_0(z) + G_1(z)H_1(z) \right\} \\ &= \frac{1}{2} \left\{ \frac{(1+z^{-1})^2}{2} - \frac{(1-z^{-1})^2}{2} \right\} \end{aligned}$$

Indeed, $\tau_0 z$ is obviously, $G_0 z H_0 z$ plus $G_1 z H_1 z$. And, when we expand this we get 1 plus z inverse the whole square by 2 minus 1 minus z inverse the whole square by 2 . And, this is easy to evaluate.

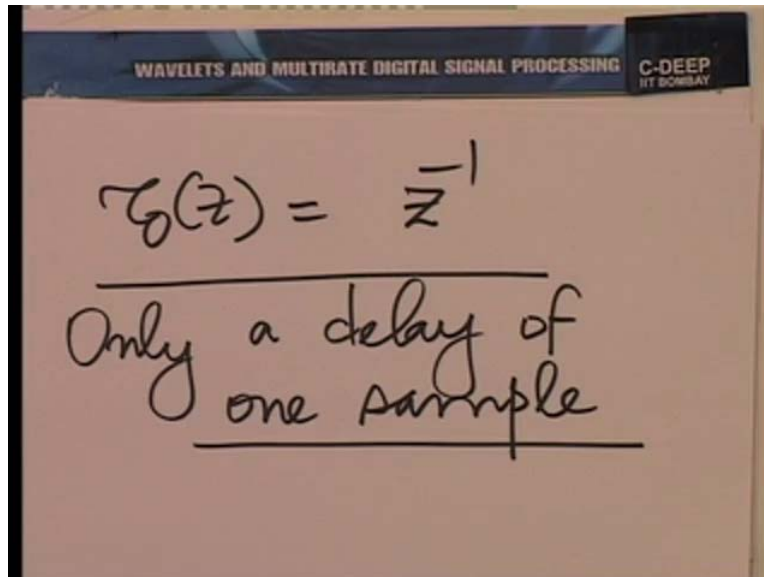
(Refer Slide Time: 19:19)

$$\begin{aligned} &= \frac{1}{2} \cdot \frac{1}{2} \cdot \left\{ (1+z^{-1})^2 - (1-z^{-1})^2 \right\} \\ &= \frac{1}{4} \cdot \frac{(1+z^{-1} + 1 - z^{-1})}{(2z^{-1})} = \frac{1}{2} \end{aligned}$$

Essentially gives us half into half. And, we can use the a plus b into a minus b kind of expression and we have 1 by 4 a plus b is 1 plus z inverse plus 1 minus z inverse. And, a minus b is 1 plus z

inverse minus 1 plus z inverse; so, two z inverse. And here, of course z inverse cancels and here we have z inverse surviving. And, all in all, this is equal to 1. Simple and elegant. In fact it is 1, but with a factor of z inverse. So, c_0 is equal to 1 and you have a z inverse there.

(Refer Slide Time: 20:42)

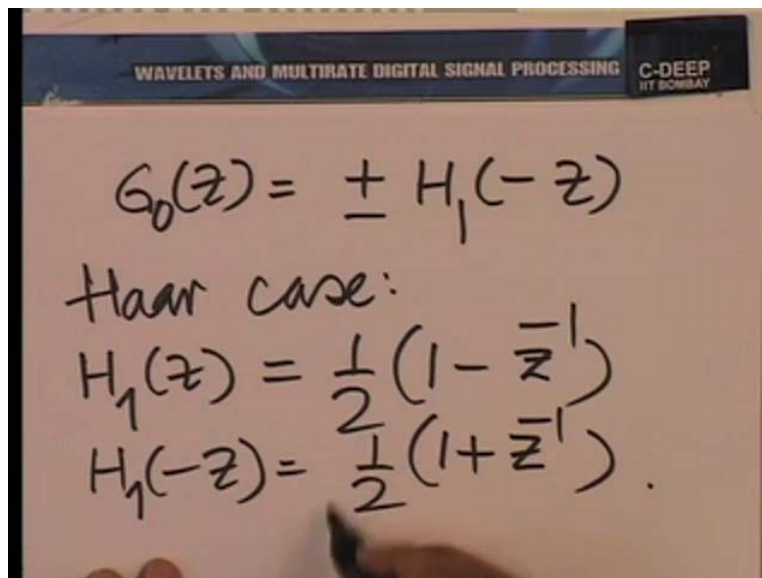


So, the $T_0(z)$, all in all is z inverse. What does this mean essentially? Only a delay of one sample. The constants have already been accommodated. So, c_0 becomes 1. Now, why was this delay required? As I said, this delay is required **on a count** of causality. If we did not **want** this delay to be there, we would need non-causality either on the analysis or in the synthesis side.

So, for example, **for example**, if I do not want this z inverse term, I must multiply the output by z. In other words, I must shift the output backward by one sample. That means, G_0 and G_1 would now become non-causal filters; wherever causality is not an issue.

So, for example, if we are dealing with spatial data, then this is not a problem. We can get $T_0(z)$ exactly equal to 1 without z inverse term. But, where causality is an issue, **as** it is when you are dealing with time data, then we cannot do this. Now, in fact in this case, let us also **digest** the situation and understand a little better. Let us put down the condition that we have written for alias cancellation in the specific and simplest case and see if it **holds** here.

(Refer Slide Time: 22:30)



WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

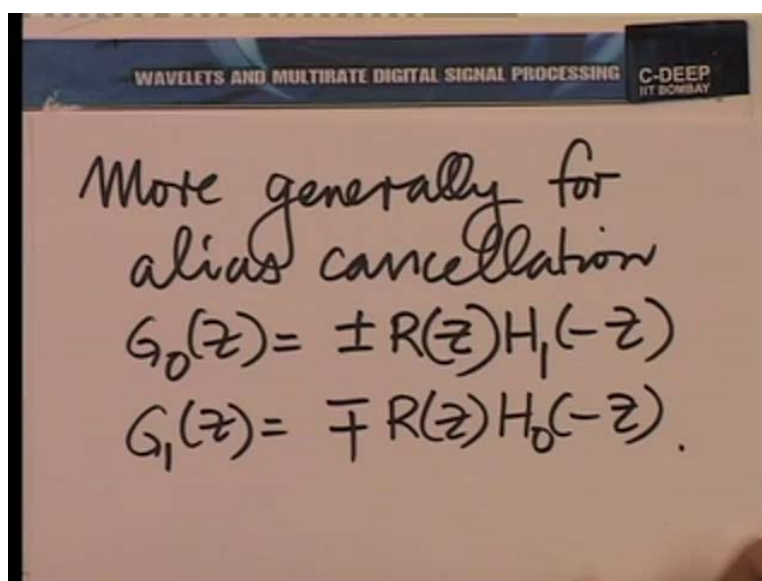
$$G_0(z) = \pm H_1(-z)$$

Haar case:

$$H_1(z) = \frac{1}{2}(1 - z^{-1})$$
$$H_1(-z) = \frac{1}{2}(1 + z^{-1})$$

So, indeed that the simplest possibility of alias cancellation is when $G_0(z)$ is either plus or minus H_1 of minus z . And, in the Haar case, we have $H_1(z)$ is essentially half 1 plus or 1 minus z inverse rather. And therefore, H_1 of minus z would be half 1 plus z inverse. So, you notice that $G_0(z)$ is indeed plus H_1 of minus z , but without this factor of half. Now, you know that factor of half is not an issue at all.

(Refer Slide Time: 23:29)



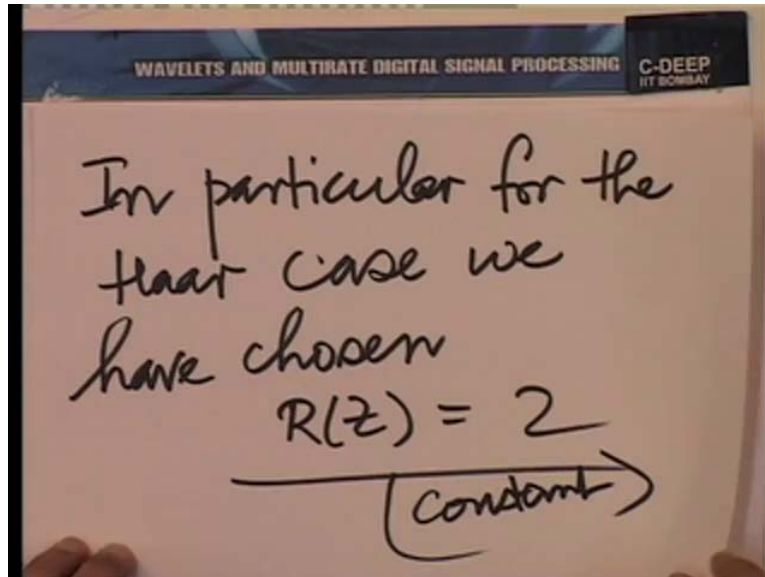
WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

More generally for alias cancellation

$$G_0(z) = \pm R(z)H_1(-z)$$
$$G_1(z) = \mp R(z)H_0(-z)$$

You remember, that more generally, we have written down the following requirements. We had said that more generally for alias cancellation, we need $G_0(z)$ to be plus or minus, some $R(z)$ times $H_1(z)$ and $G_1(z)$ to be correspondingly minus or plus $R(z)$ times $H_0(z)$. And, in particular, you could choose $R(z)$ to be a constant.

(Refer Slide Time: 24:14)



So, in particular for the Haar case, we have chosen $R(z)$ to be equal to 2, the constant. And, in fact I can also check for the second expression.

(Refer Slide Time: 24:52)

The image shows a handwritten derivation on a slide. At the top, the slide title is "WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING" and the logo "C-DEEP IIT BOMBAY" is visible. The handwritten text reads:

$$G_1(z) = -2H_0(-z)$$

Indeed

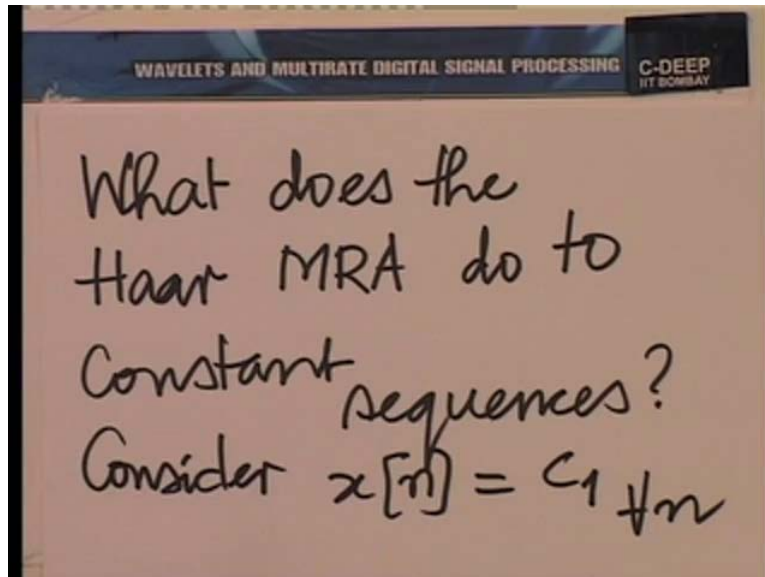
$$-2H_0(-z) = (-2) \frac{1}{2} (1 - z^{-1})$$

Correct

$G_1(z)$ in the haar case should then be minus H_0 of minus z or rather with a factor of 2; so, two times. And, indeed minus 2 times H_0 minus z is minus 2 times half into $1 - z^{-1}$, which is correct. So, things have all fallen into place. It is convenient.

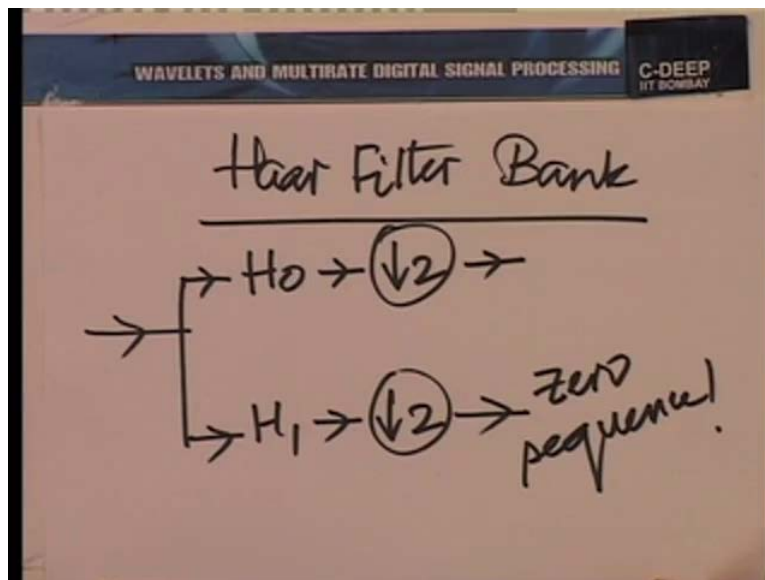
I once again, point out how beautifully one can understand several concepts at once, when one takes the specific example of the Haar. The Haar MRA embeds in it. Several concepts explained in a simple way. But, of course we cannot be content with the Haar and we shall slowly understand why. The first step in understanding this is to understand where the Haar is the baby and where **we** need to grow further. **Why is the Haar**, just the beginning of family of multi resolution analysis. In what sense is it in the best case. Towards that objective, let us look at that low pass filter and that high pass filter from a slightly different perspective. What does it do to a certain class of sequences, let us see that.

(Refer Slide Time: 26:42)



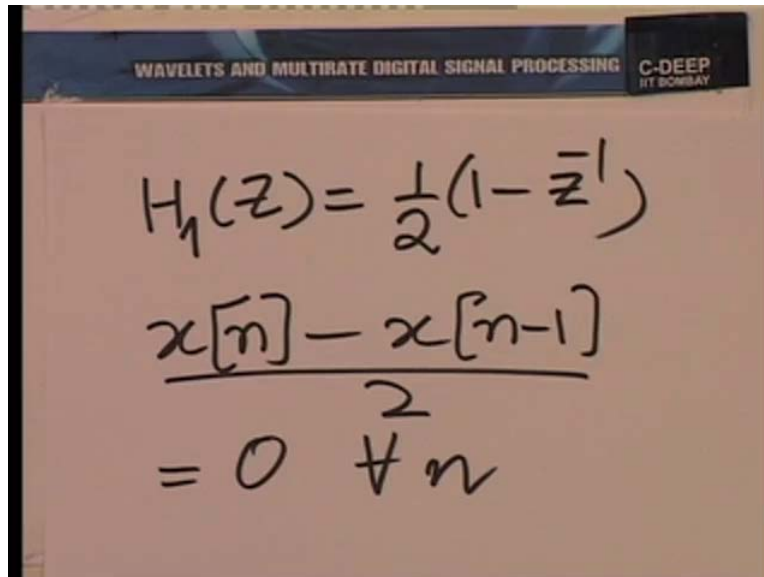
So, let us put the following question. What does the Haar do to constant sequences? In other words, consider $x[n]$ equal to some constant say c_1 , for all n ; extreme case. How would the outputs of the various points in the Haar filter bank look?

(Refer Slide Time: 27:36)



So, it is very easy to see that if you take the haar MRA, I will not keep writing the filters again. I will just show them symbolically. I have H 0 here; I have H 1 there. And, if I take just the analysis side, it is very easy to verify that the output here is going to be the zero sequence.

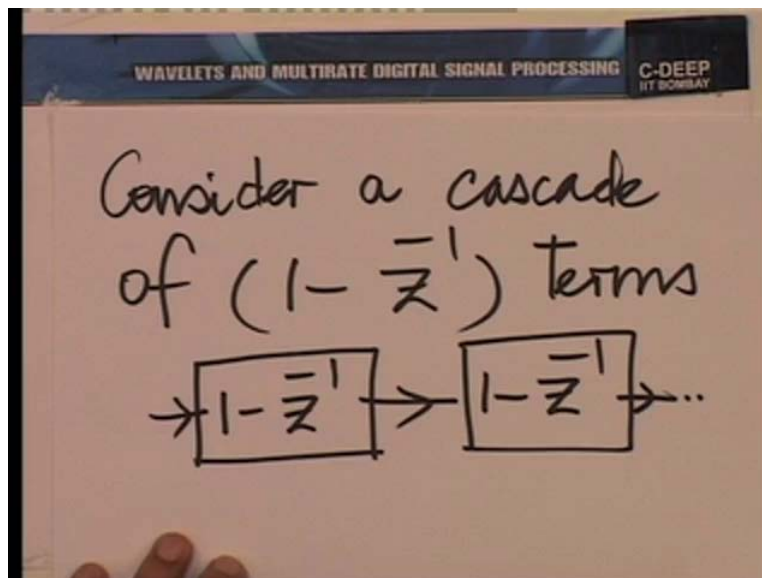
(Refer Slide Time: 28:19)



The image shows a whiteboard with handwritten mathematical expressions. At the top, there is a header that reads "WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING" and "C-DEEP IIT BOMBAY". The main content of the whiteboard is a derivation of the output of the H1 filter. It starts with the transfer function $H_1(z) = \frac{1}{2}(1 - z^{-1})$. Below this, the expression $\frac{x[n] - x[n-1]}{2}$ is written, followed by the conclusion $= 0 \quad \forall n$.

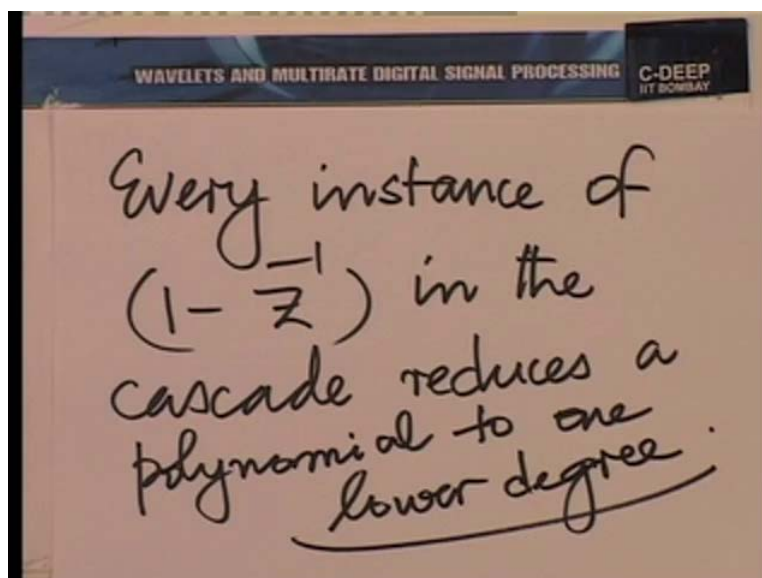
In fact, I will take just a minute and verify it. Essentially, H 1 z operates 1 minus z inverse. And, this essentially means your operation x n minus x n minus 1 by 2, which is identically zero for all n. So, this is a very significant observation we are making. We saying on the haar filter bank if there is any constant component in the input, it is destroyed on the high pass branch. This is a slightly different way of working with the haar filter bank. In fact, now we will go one step further. In the haar filter bank, I had one term of the form one minus z inverse. Suppose, I had two such terms what would happen? So, let us put that down.

(Refer Slide Time: 29:24)



So, we consider a cascade of 1 minus z inverse terms. So, you know you have a system like this; 1 minus z inverse fit into 1 minus z inverse and so on. We shall now proof a very simple and a very elegant result.

(Refer Slide Time: 30:06)

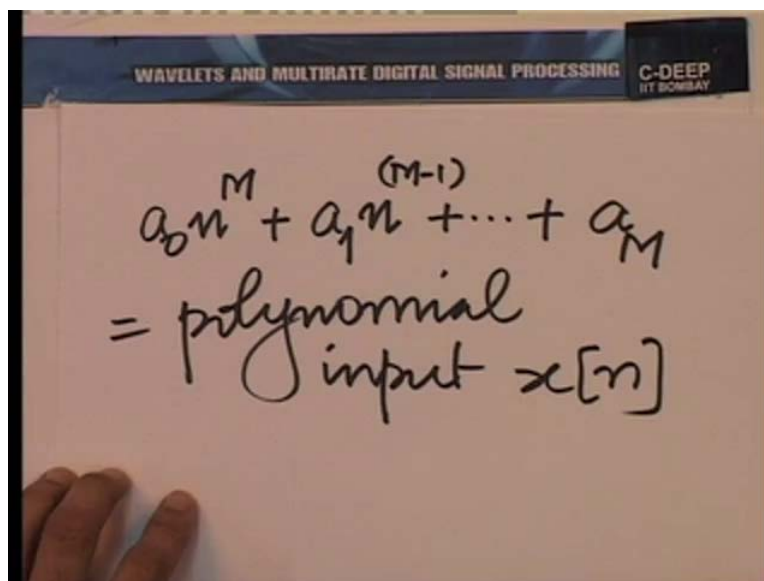


We shall show that every **cascade**, every instance of 1 minus z inverse in the cascade reduces a polynomial sequence to one degree lower. So, you know I am looking at the situation from a

slightly different perspective. Now, I am not talking about frequencies or sinusoids anymore. I am saying, suppose you think often input sequence as having polynomial components, now, **where on earth** to an encounter polynomial kind of expansion valued. We know about the Taylor series. After all, the Taylor series is essentially a polynomial expansion of an input. And, when we make a polynomial expansion of the input and we subject a few terms in this polynomial expansion to the action of $1 - z^{-1}$, we have an interpretation. That we are talking about here. So, you know visualize a region in which you are talking about the sequence... **and you know.**

So, let the sequence, for example, come from an analytic continuous function and let, then that function be expanded in a Taylor series around a certain point, which means **you have** polynomial terms. Now, let those polynomial terms be subjected to the action of this cascade of $1 - z^{-1}$. That is the situation in which we should visualize **aspects**. It is a different way of expanding an input. Anyway, so putting that context in **perspective coming back to the polynomial.**

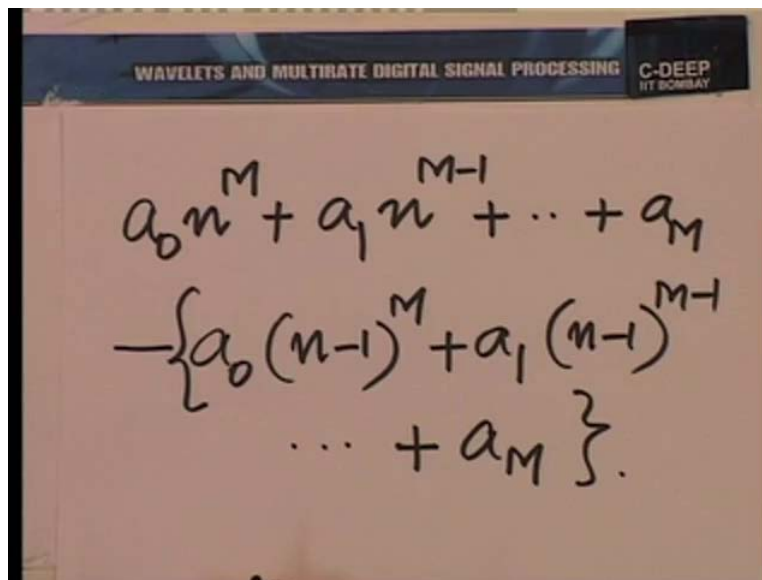
(Refer Slide Time: 32:20)


$$a_0 n^M + a_1 n^{(M-1)} + \dots + a_M$$

= polynomial input $x[n]$

So, we show that if I feed any polynomial of the form say $a_0 n^M$ plus $a_1 n^{M-1}$ and so on, up to a_M ; which is the polynomial input sequence.

(Refer Slide Time: 32:57)

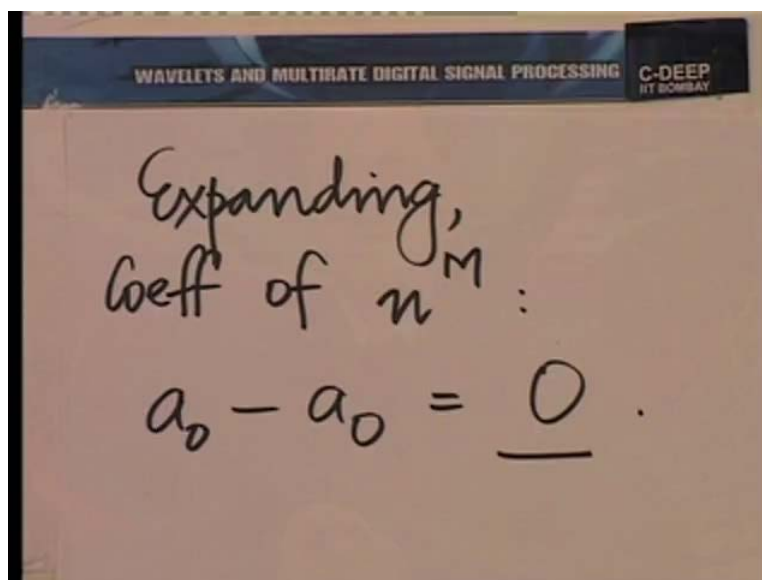


The slide shows a handwritten mathematical expression on a light-colored background. At the top, there is a dark blue header with the text "WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING" and "C-DEEP IIT BOMBAY". The expression is:

$$a_0 n^M + a_1 n^{M-1} + \dots + a_M$$
$$- \left\{ a_0 (n-1)^M + a_1 (n-1)^{M-1} \right.$$
$$\left. \dots + a_M \right\}.$$

Every time, we subject this polynomial to 1 minus z inverse, what is going to happen? So, let us subject it to 1. The first time we subject it, we are doing this. Now, what is happening in this process?

(Refer Slide Time: 33:50)



The slide shows handwritten text and an equation on a light-colored background. At the top, there is a dark blue header with the text "WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING" and "C-DEEP IIT BOMBAY". The text and equation are:

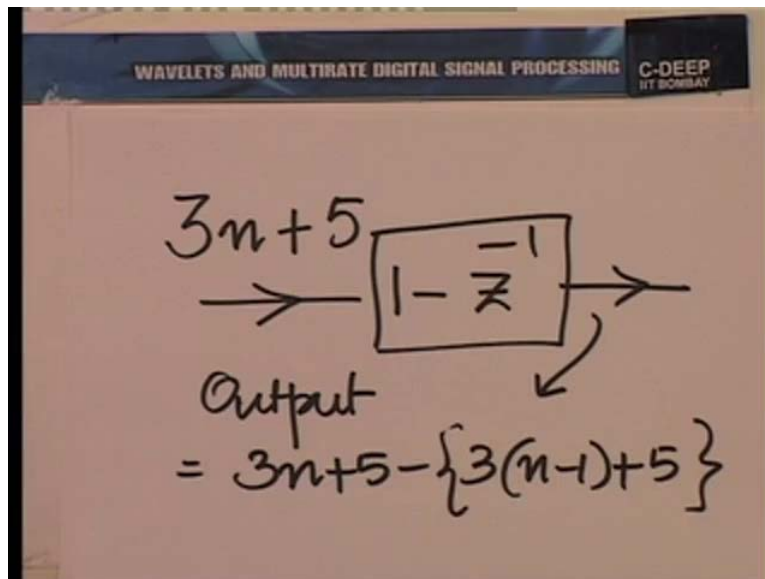
Expanding,
Coeff of n^M :

$$a_0 - a_0 = \underline{0}.$$

It is very clear that when we expand this, the coefficient of n to the power of M is easy to evaluate. It is essentially a 0, I mean, you know you have the term $a_0 n$ raised to the power of M

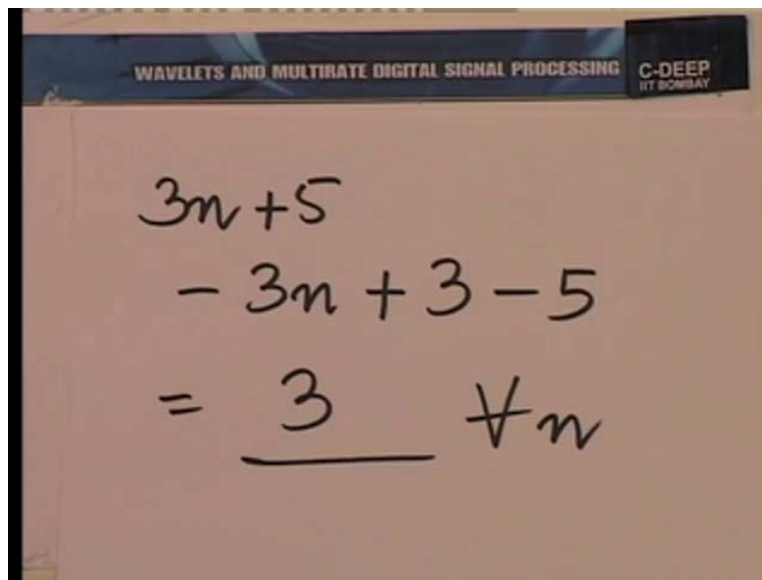
coming from here and the term $a_0 n^{-1}$ to the power of M , which contributes the coefficient of n raised to the power of M here. And, that coefficient is again a 0; so, a 0 minus a 0, which is 0. That is interesting. Each time, we subject this polynomial sequence to the action of $1 - z^{-1}$, we are reducing the degree of the polynomial by one. In fact, let me illustrate this. If I am taking a sequence which is polynomial of degree 1 and let us subject it to the action of this filter. So, you have essentially something like set...

(Refer Slide Time: 35:18)



So, let us take a concrete example. So, let us take 3 times n plus 5 and let us subject it to action of $1 - z^{-1}$ inverse to fix our ideals.

(Refer Slide Time: 35:51)



The image shows a whiteboard with handwritten mathematical work. At the top, there is a blue header with the text "WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING" and "C-DEEP IIT BOMBAY". The main content is a derivation of a polynomial expression:

$$\begin{aligned} &3n + 5 \\ &- 3n + 3 - 5 \\ &= \underline{3} \quad \forall n \end{aligned}$$

What **emerges** here is essentially $3n + 5 - 3n - 1 + 5$; that is easy to evaluate. It is essentially, $3n + 5 - 3n + 3 - 5$ and that is just 3, for all n . See, you have brought the degree of the polynomial down by one. You have a degree 1 polynomial; now you have a degree 0 polynomial. That is exactly, what happens for any degree polynomial. So, what we just showed a minute ago, and I will put back that discussion is that the coefficient of the highest power n to the power M vanishes. And therefore, when this sequence goes into z raised to the power minus 1 or $1 - z$ inverse, so to speak, you only have n raised to the power M minus 1 and lower degree terms left. So, now we just prove to simple lemma. Each instance of $1 - z$ inverse brings the polynomial degree down by one.

(Refer Slide Time: 37:36)

WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

$$(1 - z^{-1})$$

MUST BE HIGHPASS

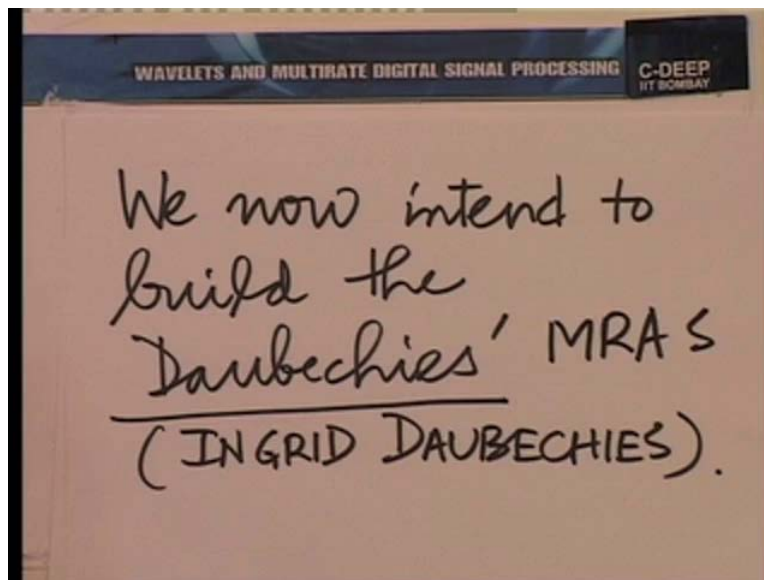
$$= 1 - e^{-j\omega}$$
$$z = e^{j\omega}$$

At $\omega = 0$ $= 0$

So, in other words, in a certain sense, the more 1 minus z inverse terms you have and by the way, we soon see these terms are going to be on the high pass branch; is not on the low pass branch. You know 1 minus z inverse, when we substitute z equal to e raised to the power of j ω , vanishes at ω equal to 0. It is verified.

So, when we take 1 minus z inverse and substitute z equal to e raised to the power of j ω , we get 1 minus e raised to the power of j ω , and then we put ω equal to 0. We get this is equal to 0. So, in other words, this is 0 d c. So, to speak 0 at 0 frequency, 0 response at 0 frequency, this cannot possibly be low pass in its behaviour. So, it must be high pass. In other words, if we do want terms of the 1 minus z inverse, they must only be present in the high pass filter. They cannot be the terms present in the low pass filter.

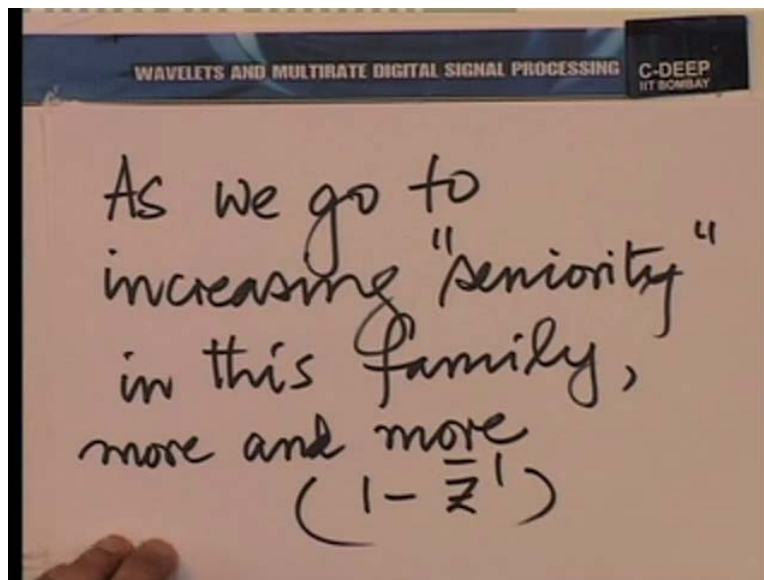
(Refer Slide Time: 39:10)



Otherwise, you know you would have a 0 response at zero frequency... for a low pass filter. In fact, what we are now going to build up is a whole family of multi resolution analysis in which you have more and more $1 - z^{-1}$ terms in the high pass branch. And, that is, in fact very well known as what is called the Daubechies' family in multi resolution analysis. Let us write that down.

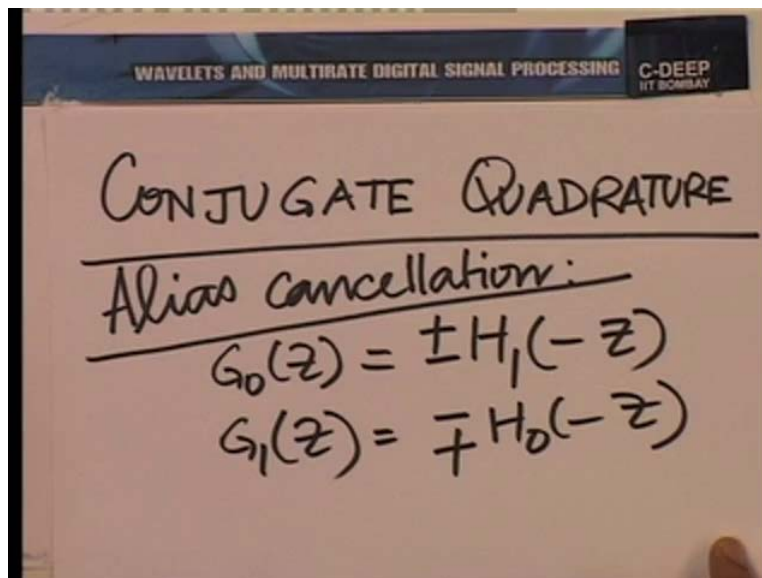
We now intend to build what is called the Daubechies' family. You know, this name is actually the name of a mathematician, scientist, whatever you might want to call her and the full name is this. I believe this is correctly pronounced as Daubechies, but I could be wrong. I think, we could just **published Daubecheis and the content.** So, anyway we now intend to build the Daubechies filter bank. So, Daubechies' MRAS.

(Refer Slide Time: 40:14)



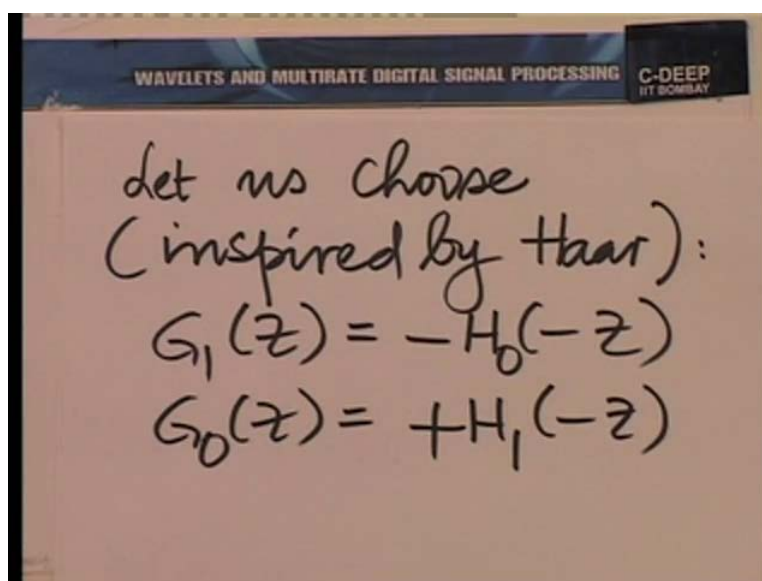
And, the features of these Daubechies MRA s is that as we go to seniors members of the family, as we go to increasing "seniority", there are more and more $1 - z^{-1}$ terms. So, on the high pass branch, we are effectively cancelling or killing higher and higher order polynomials. So, the other way of looking it is, if you are cancelling or killing them on the high pass branch, they must go to the low pass branch. So, we are retaining more smoothness on the low pass branch. That is the other way... And addition to doing this, we also want the same kind of analysis and synthesis filters. So, all these together leads us to a specific class of filter banks, which we shall now put down explicitly. And, these filter banks are called conjugate quadrature filter bank.

(Refer Slide Time: 41:39)



So, we are looking at one class of what are called conjugate quadrature filter banks. Now, we describe the equations of these conjugate quadrature filter banks are very simple. We will start from the alias cancellation condition. The alias cancellation condition says $G_0(z)$ needs to be essentially H_1 of minus z . And, we have the freedom to put plus or minus here. Similarly, $G_1(z)$ needs to be correspondingly minus or plus H_0 minus z .

(Refer Slide Time: 42:48)



Now, taking inspiration from the haar, let us take the following choice. $G_1(z)$ is minus $H_0(z)$. And therefore, $G_0(z)$ is then, plus $H_1(z)$. Now, you know, we will keep away the factor of two for the moment because after all, that factor can be observed in the $G_0(z)$ that we allowed.

(Refer Slide Time: 43:41)

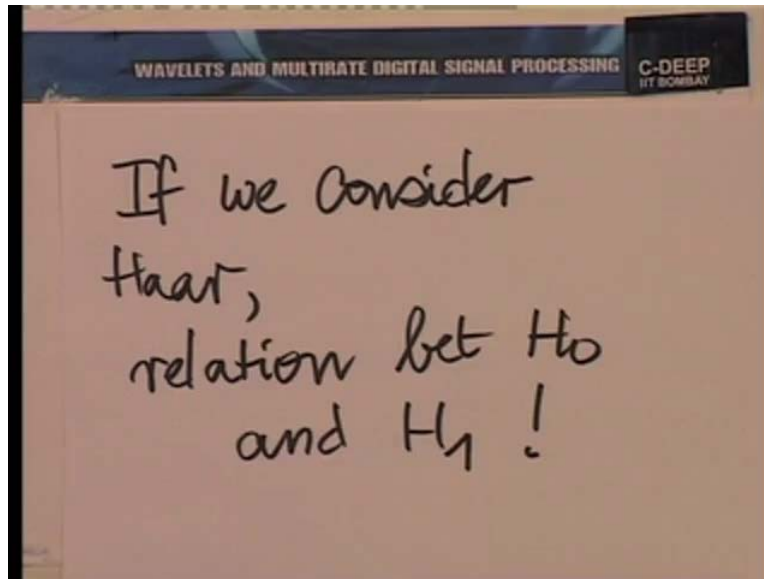
WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

$$\tau_1(z) = 0 \text{ by construction}$$

$$\tau_0(z) = \frac{1}{2} \{ H_1(-z)H_0(z) + (-H_0(-z))(H_1(z)) \}$$

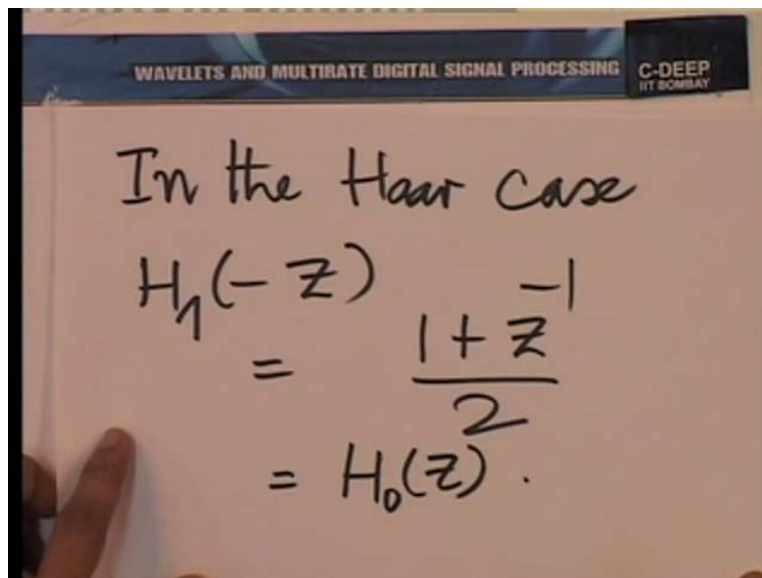
So, with this substitution, what do we get? The $\tau_0(z)$, then of course $\tau_1(z)$ is identically 0 by construction. But $\tau_0(z)$ takes the following form then. $\tau_0(z)$ takes the form half $G_0(z)$, which is essentially $H_1(z)$ times $H_0(z)$ plus minus $H_0(z)$ times $H_1(z)$. So, we have an interesting situation here. We have this $H_0(z)$ $H_1(z)$ product. You know, $H_1(z)$ is essentially a high pass filter; $H_1(z)$ therefore essentially becomes or aspires to become low pass filter with a cut-off of $\pi/2$. So, here you essentially have a cascade of two low pass filters with cut-off $\pi/2$. And, correspondingly this becomes a cascade of high pass filters with cut-off $\pi/2$. And, you are effectively saying that the overall system functions with this cascade of low pass filters of cut-off $\pi/2$ and this pair of high pass filters with cut-off $\pi/2$ must go towards a perfect frequency fraction situation. That is the interpretation of the equation here. So, we now focus on $H_1(z)$.

(Refer Slide Time: 45:34)



You see, if we look at the haar, once again there is the relation between H_0 and H_1 . In fact, what we trying to say is, we of course rate the synthesis to the analysis for the purpose of alias cancellation. But, now we have a perfect frequency structure requirement. So, we want this $\tau_0 z$ to essentially go to a delay and multiplying constant. Now, that means you need a relation between H_1 and H_0 . And, one simple thing to do is to make $H_0 z$ related to H_1 minus z . That is what we have actually done in the haar.

(Refer Slide Time: 46:37)



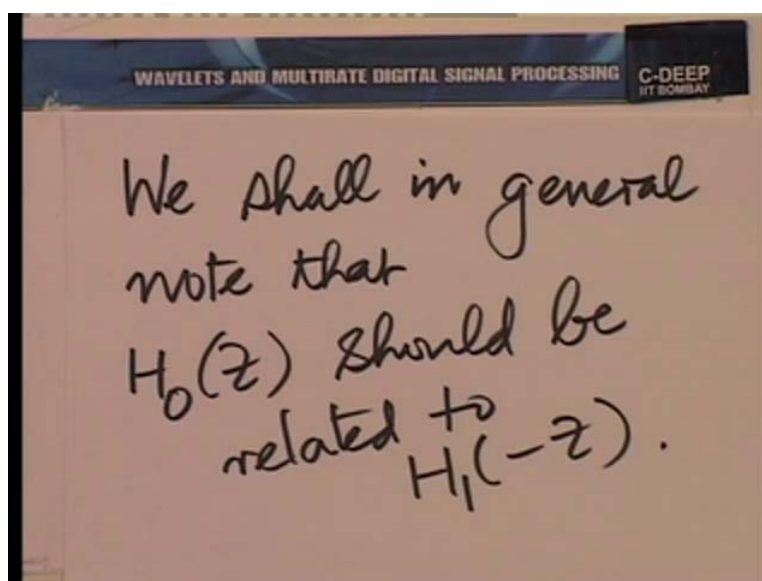
WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

In the Haar case

$$H_1(-z) = \frac{1+z}{2} = H_0(z).$$

If you look at it carefully in the Haar case, $H_1(-z)$ is essentially $1+z$ inverse by 2. So, you know $H_1(-z)$ and $H_0(z)$ are very closely related. Now, let us generalize this. In fact, the **only catch is you know** we will later on need to make a little adjustment here. So, at this moment, in which simply aspect $H_1(-z)$ to be equal to, see in this the Haar case, this is equal to $H_0(z)$, but we may need to make a little adjustment here.

(Refer Slide Time: 47:34)

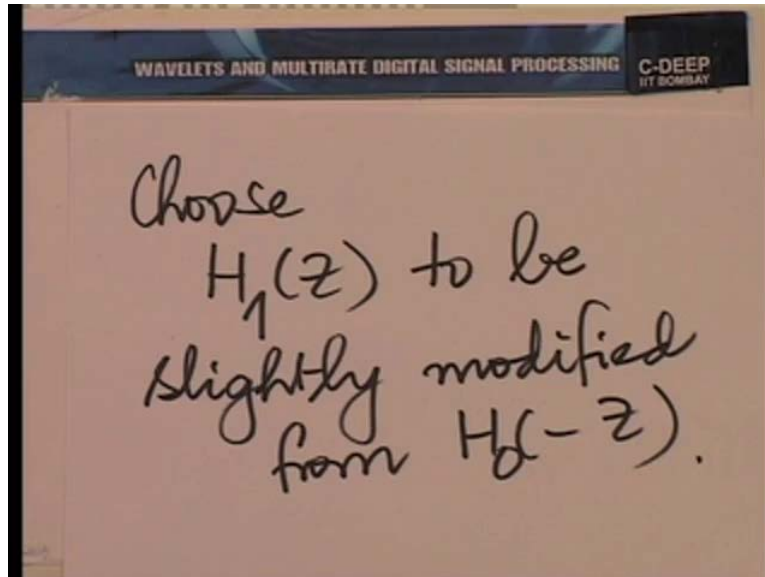


WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

We shall in general note that $H_0(z)$ should be related to $H_1(-z)$.

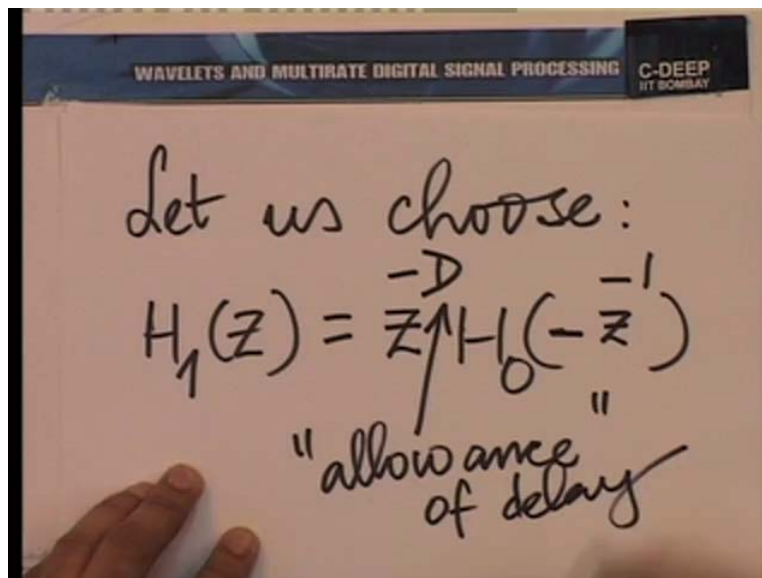
So, what we will do is, we shall in general, note that H_1 or rather H_0 should be related to H_1 minus z . So, in the Haar case **here** equal, but in general we ask for a relation, a very close relationship. The other way of saying it is that if you replace z by minus z in H_0 , we should get the H_1 .

(Refer Slide Time: 48:20)



So, what we are saying is choose H_1 to be **derived form** or to be slightly modified form H_0 of minus z . Modified in what way? So, in fact you know here again there is a little bit of an issue. So, what I am now going to do is to put down a choice for H_1 by knowledge or by an exposure to the filter banks **s** that I have and justify it later. So, you will have to bear with me for a little while, not too long, may be just about a lecture. I shall put down the choice here; I shall partially justify the choice in this lecture and completely justify the choice in the next lecture, where we once again look at the whole system in **total**.

(Refer Slide Time: 49:32)



WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

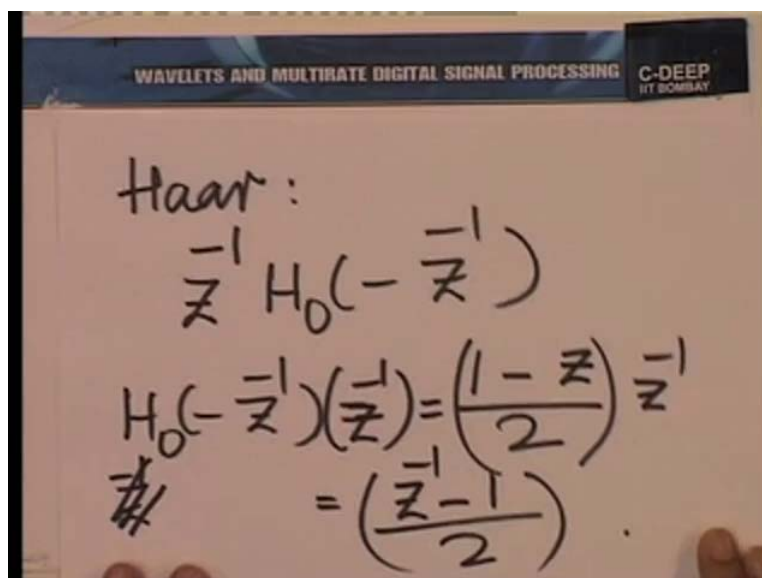
Let us choose:

$$H_1(z) = z^{-D} H_0(-z^{-1})$$

"allowance of delay"

So, let us choose $H_1(z)$, not to be H_0 of minus z , but H_0 of minus z inverse. And, we will also allow for the possibility of z raised to the power D here. So, we will say minus D . We will allow for this possibility.

(Refer Slide Time: 50:32)



WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

Haar:

$$z^{-1} H_0(-z^{-1})$$
$$H_0(-z^{-1}) \left(\frac{-1}{z}\right) = \left(\frac{1-z}{2}\right) z^{-1}$$
$$\neq \left(\frac{z^{-1}-1}{2}\right)$$

If we do that, in fact we can verify for the case of Haar, it is very easy. For the haar case, let us take z inverse times H_0 of minus z inverse and indeed H_0 of minus z inverse for the Haar case

is essentially 1 minus z by 2. And then, if I take z inverse times H 0, so if I multiply this by z inverse, multiply both sides of z inverse.

(Refer Slide Time: 51:33)

WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

Consider

$$H_1(z) = z^{-D} H_0(-z^{-1})$$

I have essentially z inverse minus 1 by 2. So, we are doing well. Now, let us make this the most general case. So, we will consider H 1 z to be of this form; z to the power minus D H 0 minus z inverse. And, write down the tau 0 z for this.

(Refer Slide Time: 51:49)

WAVELETS AND MULTIRATE DIGITAL SIGNAL PROCESSING C-DEEP IIT BOMBAY

$$\tau_0(z) = \frac{1}{2} \left\{ H_0(z)(-1) H_0(z^{-1}) z^{-D} - H_0(-z) z^{-D} H_0(-z^{-1}) \right\}$$

Tau 0 z would become half, then $H_0 z H_1$ of minus z. So, H_1 of minus z becomes minus 1 raised to the power minus D times $H_0 z$ inverse. Now minus, you see you have $G_0 z$. Again, now we have $G_0 z$ and let me put back the expression for you just for convenience.

We want tau 0 z to be this; H_1 minus z times $H_0 z$, for which we have this term $H_0 z$ into H_1 minus z. All right. We have z raised to the power of D and minus H_0 minus z times $H_1 z$. Now $H_1 z$, it is accepted to be z raised to the power minus D times H_0 minus z inverse. So, we have z raised to the power minus D here.

Now, what we intend to do in the next lecture is essentially to look at this expression. So, we have $H_0 z$, $H_0 z$ inverse here, H_0 minus z, H_0 minus z inverse there. The z raised to the power minus D terms there. And, of course you have the minus 1 raised to the power minus D here and minus 1 here. And, we need to choose these strategically. It is clear. And then, we also need to put some conditions on H_0 . So, this indeed becomes a perfectly reconstructions situation. We shall complete this in the next lecture and take it further from there. Thank you.