VLSI Data Conversion Circuits
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Lecture - 36
Dynamic Range Scaling

This is VLSI Data Conversion Circuits, lecture 36. So, far we have seen several attributes of continuous time delta sigma modulators, in the problems that are associated with these modulators.

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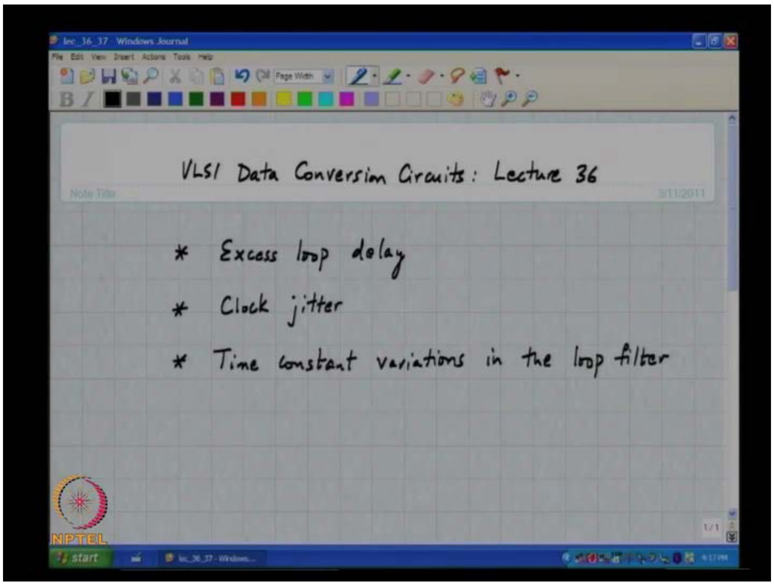

The first one we saw, as far as non idealities in these converters are concerned are excess loop delay, and we saw how to fix this two, this can be fixed by adding a Dirac path around the quantizer, or you can think of it as a feed forward path across the loop filter. The way this works is that, either case it looks like feed forward 0 is added into the transfer function, thereby stabilizing the loop, which would otherwise tend to become unstable, because of the excess delay which acts like extra phase in the loop.

The next problem that we discussed was clock jitter, and we say that clock jitter effects the performance of the modulator through the variation of the pulse widths of the feedback DAC. And we saw the effect of various pulse shapes on a modulator A the NRG DAC, B the returned 0 DAC and C the exponentially e decaying DAC. And we saw that, the returned 0 DAC is particularly bad for clock jitter, because there are two edges with an every symbol period; and each edge is twice as high as it would be in a NRG case.

So, the effect of jitter is significantly worse, the exponentially decaying DAC on the other hand will be less sensitive to jitter simply, because if the edge moves it does not matter the area under that pulse still remains the same. As long as the time constant is much smaller than the symbol period. Now, the last non ideality was time constant variations of the integrators, we saw the intuition why the noise transfer function can change.

And that is because, if the time constants become too small it means that the band width of the loop filter increases, if the band width of the loop filter increases. Then for in band signals the gain of the loop filter is high which means that, the signal to quantization noise ratio increases. Which must by a bode sensitivity be accompanied by some kind of degradation in the high frequency performance of the noise transfer function. So, most often there are not, you will end up with the NTF having a smaller maximum stable amplitude, if not becoming totally unstable, when the time constants become too small.

On the other hand, when the time constants become too large or rather increased compared to the nominal, then the gain of the loop filter within the signal band falls and you will find that the in band noise increases, which also means that the high frequency behavior is slightly better. So, you can at least expect a slight increase in the maximum stable amplitude, in general the variation of this time constants cannot be too large.

So, it is common to have some kind of system on the chip, which measures the time constant, and tweaks it in such a manner that, this product of the time constant times the sampling rate remains close to, what we intended this time constant to be. Any deviations can be in principle at least, can be addressed by tweaking the resistor values or the capacitor values. One way of varying the resistors is to have a whole bunch of resistors in parallel, and choose the one which is closest to the resistor you want.

Another way is to have a bank, where you have a fixed resistor plus a variable part and you are going to tweak the variable part such that, you after tweaking the resistor becomes close to what you wanted in the first place. So, today we will discuss some practical issues regarding the design and simulation of these converters.

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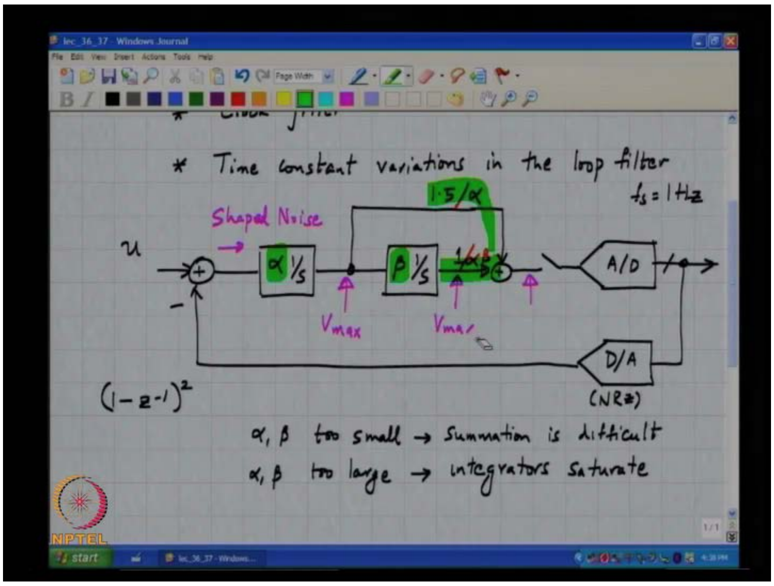

As a case in point I just want to start off with a second order example, let me for argument sake take a CIFF second order loop filter, where this is k 1 and this is k 2 and this is the quantizer, this is u. Now, if you want to realize a noise transfer function with the form 1 minus z inverse the whole square, and assuming as usual that are sampling rate f s is 1 Hertz, what should k 1 and k 2 be, if you a assume an NRG DAC.

Student: 1.5

1.5 and 1, now let us observe the following, let us say multiply this by alpha or rather let me make this, let us say I multiply this sub by alpha, what should I do in order to keep the noise transfer function the same, in spite of multiplying this sub by alpha. These coefficients mind you are 1.5 and 1 to begin with, now for some strange reason which we will come to in a couple of minutes, let us say I decide to multiply this by alpha. Clearly the noise transfer function is different, what should I do to make the noise transfer function the same.

Student: ((Refer Time: 08:16))

It is quite straight forward, I should this signal here is would be now alpha times larger, so if I did nothing the signal here would be alpha times larger, so but then I want the signal to be the same as I had before. So, all the signals that depend on this node here, must accommodate themselves in such a way that their outputs remain the same as before, so as all of you have indicated this must go down by alpha. And what about if I decide to change this coefficient what should happen, becomes again goes as 1 by alpha.

Some of you may say, you might as well add the instead of having the 1 by alpha in the coefficient, you might have it in the integrator, that is also a perfectly acceptable thing. Or you might do both and say, I am now going to modify this integrator, what was 1 by s before I am going to multiply this by say beta, what should I now do.

Student: 1by alpha beta

1 by alpha times beta, and alpha and beta can be any positive numbers and for all values of alpha and beta the noise transfer function is the same. The signal transfer function is also the same, because the loop filter transfer function is the same. Then one question that naturally arises is, if for all values of alpha and beta the STF and the NTF remain the same is there a smart way of choosing alpha and beta, is this clear.

So, for all values of alpha and beta, it appears as if the input output properties are the same, now what are the input output properties, that we are interested in the signal transfer function, as well as the NTF. And for all alpha beta we all by construction we know that, the STF and the NTF remain the same, so the question is are some choices of alpha and beta, better than other choices. And if so how would we find those choices, any comments.

Student: There is no

[FL]

Student: Small

What should be small

Student: Alpha and beta

Why?

Student: ((Refer Time: 11:41))

Ok

Student: It have should be larger to do noise

Well not quite, but I mean one thing to observe is that, if ((Refer Time: 12:02)) this box is given to you and only the input and output ports were accessible, then for all values of alpha and beta there is absolutely no difference in the input, output of the box. So, if the integrators were perfectly linear, and these coefficients were chosen this way, then there is no special value of alpha and beta, which is better than any other.

Because, as far as input output properties are concerned, the box behaves the same, however it turns out that in practice all these integrators are going to be realized using active circuits. So, these integrators will not be perfectly linear for all values of output swing, so every integrator output namely this ((Refer Time: 12:59)) and this will have a limit to how large those voltages can be, before the integrator becomes unacceptably non-linear.

I mean without getting into too many details at this point all that I had say is, all of you are familiar with how 1 makes integrators with opamp's, it is an r c, the capacitor is in feedback across the between the output and the inverting terminal of the opamp. Now, with the output of the integrator is basically the output of the opamp, so if the voltage swing at the output of the opamp becomes too high, you risk the chance of saturating the opamp. In which case the gain of the opamp is not large anymore which means that, you do not get ideal integration between the input and output.

So, to very crude approximation 1 can say as long as the output swing of the integrator is less than a certain value, or as long as the swing at the output of the integrator lies within some limits, one can think of it as a perfectly linear integrator. Beyond the swing one can think of it as a very bad integrator, because of all sorts of non-linearity is creeping into the integrator. So, the bottom line, therefore is that you want to make sure that the peak swings at the outputs of all the integrators, do not exceed a certain limit, in practice what do you think will impose this limit.

Student: Power supply

The supply voltage will impose this limit in practice, but at this stage all that one needs to be aware is that, it is quite reasonable to expect that if these voltages become too high, these integrators will start misbehaving, and will not act like ideal integrators. So, let us call the peak levels here, V max and minus V max, so in other words the absolute voltage, or the absolute value of the integrator output cannot exceed V max, does it make sense. Can somebody comment about the swing here, as a function of alpha and beta.

Student: C inverse

That remains the

Student: same

Same

Student: Independent of alpha

It is independent of alpha and beta, why

Student: Because, it is an a d converter adding like, the gain of the a d converter is there ((Refer Time: 16:21))

Ok

Then we know what is the output, then it goes to, the input has to be the signal plus some ((Refer Time: 16:29))

So, the input of the quantizer is nothing but, STF times u plus quantization noise times NTF of z minus 1, the STF and the NTF remain the same, so regardless of alpha and beta, the output of the loop filter must still be the same. Now, the next question is given now that the outputs of the integrators cannot exceed a maximum level, now are there smart values of alpha and beta. One logical choice is to say, I am going to choose alpha and beta to be very very small, so that the signal swings here are.

Student: Small

Definitely much smaller than V max, so what do you think is a possible disadvantage with this approach.

Student: ((Refer Time: 17:42))

No, it is a unchanged

So, one thing that is not immediately apparent is that, if we attenuate signals, here if the signal swings are very small finally, this swing is not changing, as we just agreed the voltage there is independent of alpha and beta. So, if you make this alpha and this beta very very small with the hope that, I will keep this swings very low and avoid saturation of the opamp all together. Then we are faced with the appeal task of amplifying this very small signals and bringing them up to a large value, you understand.

So, and as if you have to add two signals and multiply them up by a large number, in practice I mean this requires an amplifier, and the larger the gain of the amplifier is in general you can expect the band width of the amplifier to be small. Now, if you have finite band width inside in the summation, what do you think that will do to a loop, if one can expect that it behaves, a lot like excess delay. Because, finite band width in the loop means that, there is the output of the loop is coming out, a little later than it should ideally come out which means that, it kind of behaves like excess loop delay.

So, choosing alpha to be too small with the hope of keeping the integrator outputs very small, so that they would not become non-linear is not a particularly good idea. On the other hand, choosing alpha and beta to be very large saying that then, I will need very small gains, so my addition becomes easy is also not a good idea, because...

Student: ((Refer Time: 20:04))

No, we all discussed what the problem is with small alpha and small beta, what is the problem, the summation becomes difficult to do, because you are takes add two small signals and then, amplify that result by a large number. Now, let us go to the other extreme and then, say let me choose alpha and beta to be very large, so what do you think will be the problem then...

Student: Integrators

The integrators will

Student: Saturate

Will saturate, because the peak swings, I mean please recall that what is going into the loop filter is what, it is simply the shape noise and that is independent of alpha and beta. So, this signal is simply shaped noise, and if you choose alpha and beta to be very very large, then you stand the risk of saturating the integrators, so now what do you think is right thing to do.

Student: Beta less than outcome

Let me just put the conclusions of our discussion down, alpha beta too small summation is difficult, alpha beta is too large integrators saturate, so what should we do, if we choose alpha beta neither too large not too small. And what should we look out for, and what is the meaning of not too large or too small.

Student: ((Refer Time: 22:47))

So, simply from a view point of trying to implement this summation in a easy way, I would have like to have large alpha and beta, but that would saturate the integrators. So, I choose alpha and beta such that, the integrators peak outputs are limited to V max, limiting the output of the integrators to say V max by 10 is very straight forward, I would decrease alpha and beta by a factor of 10. But, that does not make sense from a practical point of view, because then I am face to the uphill task of amplifying these small voltages back again.

So, the best thing to do, therefore is to scale or choose alpha and beta in such a way, that the peak outputs of both integrators remain within V max. And this is what is called, so you will choose alpha and beta such that, the peak output swings of both the integrators is just a little bit less than V max. And what do you think you will do to determine the peak swings at the outputs of these integrators, how will you determine the peak swings.

Student: ((Refer Time: 25:22))

No, no see V max can be found once the details with the construction of the integrator are known, what I am asking you is given V max, how will you choose alpha and beta.

Student: ((Refer Time: 25:44))

No, correct as high as possible, so that the outputs do not, let us say you choose alpha equal to 1 and beta equal to 1, so what will you do to determine what the peak value is here for alpha equal to 1 and beta equal to 1, do you understand the question. And one way of doing this, is to say I will choose some random alpha and beta I will find the peak values of the outputs of the both the integrators. I know that I want the peaks to be V max, so I know the actual peak, I know the desired peak, so from that I can go and determine what alpha and beta must be is it not.

So, but how will I determine the peak signal output of the first integrator for instance

Student: ((Refer Time: 26:49))

So, now what I was trying to get at is to determine the maximum possible output, I mean swing at each of the integrator outputs, what one would do, would be to apply an input signal run simulation. Please note that the input to the loop filter in this particular example is shape noise plus a very small amount of input signal, we discussed this before. So, given that the input is a shape noise, it becomes difficult to analytically find the peak outputs of each of these integrators.

So, easiest thing is to run a long simulation and monitor this output quantity and the output in the second integrator, and from that we can compute the peak values, what peak value you desire for the output of that particular integrator. And therefore, once you know that you can always find alpha and beta is this clear, I mean one comment regarding this structure as is essence, this is a CIFF type structure, is the following. Can you comment on this the signal swing at the output of the second integrator, as a function of the input amplitude.

Student: ((Refer Time: 28:57))

I mean the output of the second integrator, which is the 1 by S square path of the loop filter will largely consist of the input plus some amount of shape noise whereas, the output of the first integrator will consist of very very little of the input plus...

Student: Lot of shape noise

Lot of shape noise, so when you want to do this scaling in this particular modulator example, one would need to apply the maximum input swing that, the modulator is likely to see, you understand. And then, check the peak outputs of all the integrators, so if the peak amplitude here was P 1 and here it was P 2, for alpha equal to 1 and beta equal to 1.

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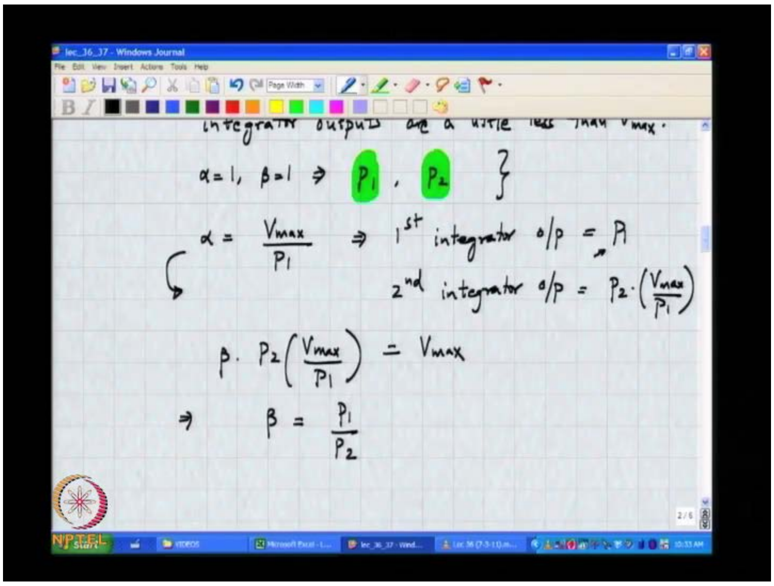

Then, one should choose alpha to be equal to, let us say for alpha equal to 1 and beta equal to 1, the peak value at the output of the first integrator was P 1 and the second integrator was P 2. What should I do now, I want the peak value at the output of the first integrator to be V max.

Student: ((Refer Time: 30:53))

So, alpha must be V max by P 1, if alpha has changed from unity to V max by P 1, then as we just said the peak value at the output of the first integrator will be P 1, or increasing in alpha causes this output to increase. And therefore, will cause the output of the second integrator also to increase by a factor of V max by P 1, so after this step the first integrator output maximum is P 1, therefore the output of the second integrator is now gone up from P 2 to P 2 times V max by P 1.

Now, you want the second integrator output to also have a maximum at V max and therefore, beta must be changed such that, beta times P 2 times V max by P 1 equals V max which means that, beta must be P 1 by P 2. This way the maximum outputs of both integrators will be V max and this is the best you can do, if you make these constants alpha and beta too small. As we just discussed the opamp swings or the integrator output swings become too small, which brings in challenges with respect to adding the integrator outputs.

If alpha and beta are too large, you tend to saturate the integrators thereby causing a whole bunch of other problems, so this choice seems like a good choice of coefficients alpha and beta. And all that is needed to figure out alpha and beta is one simulation with say alpha equal to 1 and beta equal to 1, and one can determine the maximum P 1 and P 2. From which one can go and compute what coefficients are necessary in order to keep the integrator peak maxima the same.

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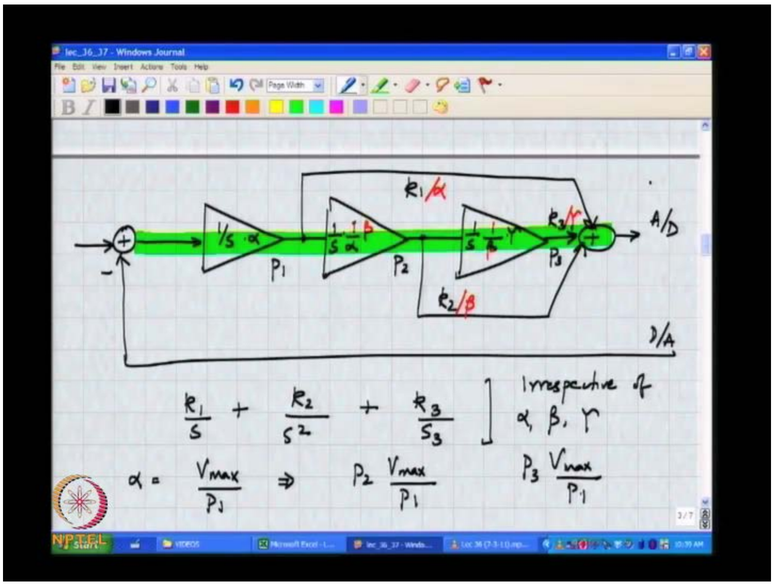

Now, in a more general case let us say we have integrators, we have three integrators, again arranged as a CIFF loop, and this is the rest of the modulator which consist of the A to D and D to A path, and I will only focus on the loop filter here. We assume the coefficients of the first order path to be k 1 by s, k 2 by s square and k 3 by s cube, now if the peak voltages for this particular choice of coefficients was P 1, P 2 and P 3. And let us say we introduce scaling factors in each of these integrators, if we multiplied this by alpha, one would have to divide this by alpha ((Refer Time: 35:42)) and this by alpha and this by alpha.

If I multiplied the second integrator by beta, then I would need to multiply k 2 with 1 over beta and k with 1 over beta, so we see that if I multiply the third integrator with a coefficient gamma, then I would have to divide this by gamma. So, we see that multiplying the integrator outputs or scaling the integrator coefficients, can be done without modifying the noise transfer function. Because, in spite of this scaling factors the loop filter transfer function still remains k 1 by s plus k 2 by s square plus k 3 upon s cube, this is irrespective of and gamma.

Now, what one does in practice is very similar to what we discussed with the second order case, if the swings at the outputs of the integrators are very small, then adding them up becomes a problem. If the swings are too large one risks, the prospect of saturating the integrators thereby causing a whole bunch of other problems, so one would like to keep the swings at the outputs of the integrators as large as possible without saturating them.

And the strategy is to choose for instance alpha such that, it is V max by P 1 and in this particular loop filter architecture P 2 and P 3 will go up by the same factor, the moment alpha goes up by a factor of V max by P 1. So, this means that the second integrator output maximum will become P 2 times V max by P 1, and the third integrator output peak will now become P 3 into V max by P 1.

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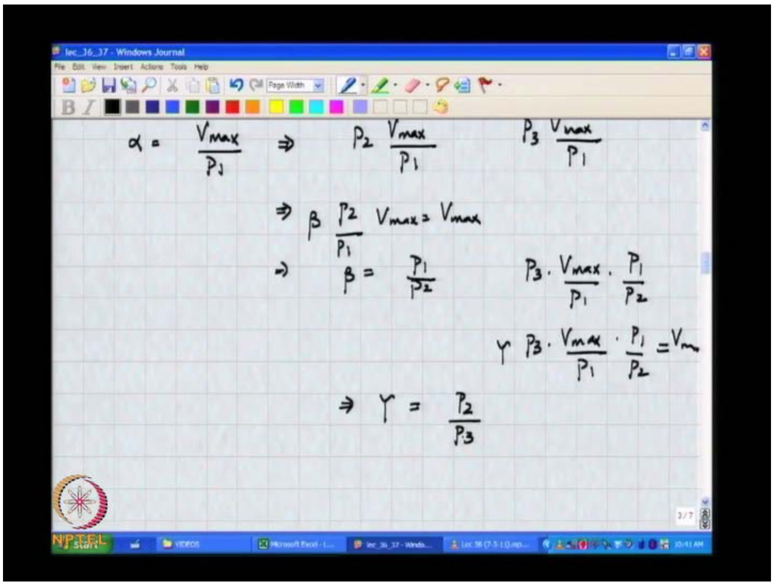

Now, if we like to keep the second integrator peak at V max, then what we would like to do is to keep P 2 by P 1 times V max equal to V max. And this means that beta must be chosen to be P 1 by P 2, which in turn means that P 3 the peak voltage with the output of the third integrator, is now scaled up by a factor of P 3 times V max by P 1, which was the original peak voltage multiplied by beta. Because, the output of the second integrator feeds with third integrator.

If you want to make this equal to V max, then gamma times P 3 times V max times P 1 by P 2 must be equal to V max, which means gamma equals P 2 by P 3, another practical thing to do is the following, as the input signals become larger and larger, the integrators will saturate. Regardless of how you scale the integrators, it is only a matter of make giving a large enough input to the modulator, that will cause the integrators to saturate. Now, one obvious question that arises is should we allow all the integrators to saturate at once, or should we allow integrators to saturate one by one in a gradual fashion.

And the answer for that question is the following, recall that the third order path 1 over s cube path has got the highest gain in the signal band; that means, as the input signal goes on increasing an amplitude; this output of the third integrator starts to become larger and larger. Now, if this integrator saturates first that is, if the third integrator saturates first, then it is effectively like a removing the third integrator from the feedback path, simply because the output of the integrator does not respond to the input any more.

On the other hand, if the first integrator saturates first, then what happens it is like breaking the entire feedback loop, because if the output of the first integrator saturates, none of the other integrators are able to make any sense of the feedback signal. Simply because, it is the equivalent to breaking open the feedback, this way the entire loop is operating in an open loop fashion, simply because the first integrator for all practical purposes is completely cut off.

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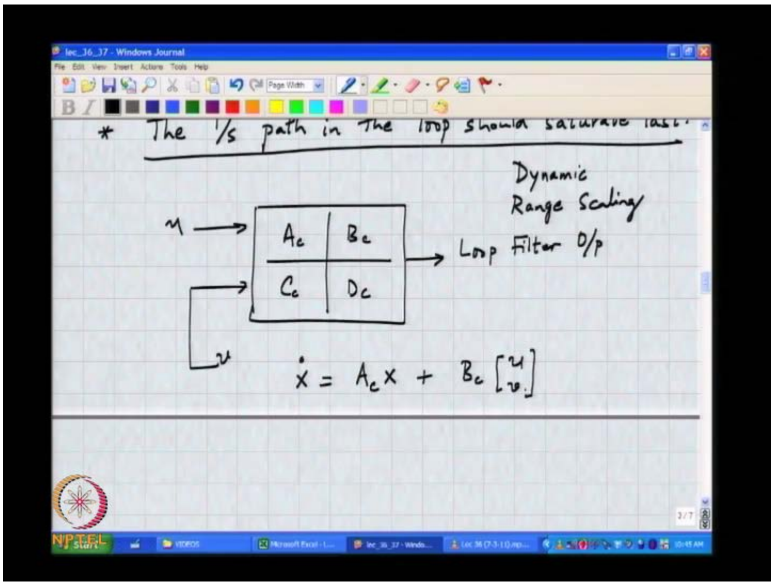

So, it make sense, therefore to make sure that the 1 over s path in the loop should saturate last, so this will then make sure that, as the input amplitude keeps increasing a modulator which was originally a third order modulator, will first become equivalent to a second order modulator. Because, the third integrator has saturated and will then, become a first order modulator, because the second and third integrators are saturated. On the other hand, if the 1 over s path was chosen to saturate early, then it is equivalent to cutting off the entire feedback loop, right at the beginning.

So, in general in a more mathematical way this whole concept of changing the coefficients of the loop filter, to make sure that the peak amplitudes at the outputs of the integrators are follow certain pattern, this process is called dynamic, range, scaling. And can be expressed mathematically as the following, let A c, B c, C c and D c represent the state space equations of the loop filter.

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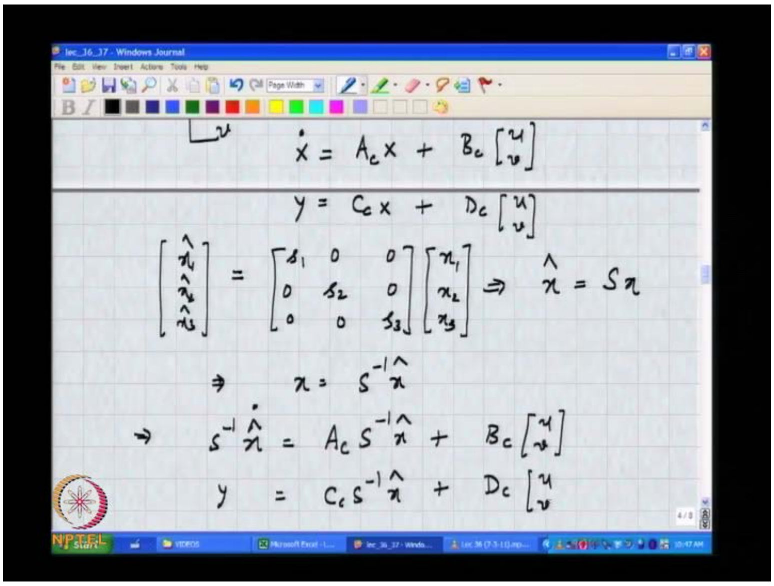

So, this is the loop filter output and these are the two inputs, the state variables follow or obey these equations x dot as A c x plus B c times u and v and y is C c x plus D c u and v. Now, we want the same input output transfer function, but we want the state variables to be scaled by certain values, so let us assume that the new state variables are denoted by x hat. And these x hats are related to x, so x 1 hat x 2 hat and x 3 hat, for example could be the new state variables in a third order loop filter; and all we want to make sure is that, the new state variables are some scaled versions of the old state variables.

In other words, the new state variables can be written as some scaling matrix S times x, the scaling matrix S, in our particular example will be a diagonal matrix. So, x can be written as S inverse times x hat which means that, S inverse x hat dot must be A c times s inverse x hat plus B c times u and v and y is C c times S inverse times x hat plus D c times u and v.

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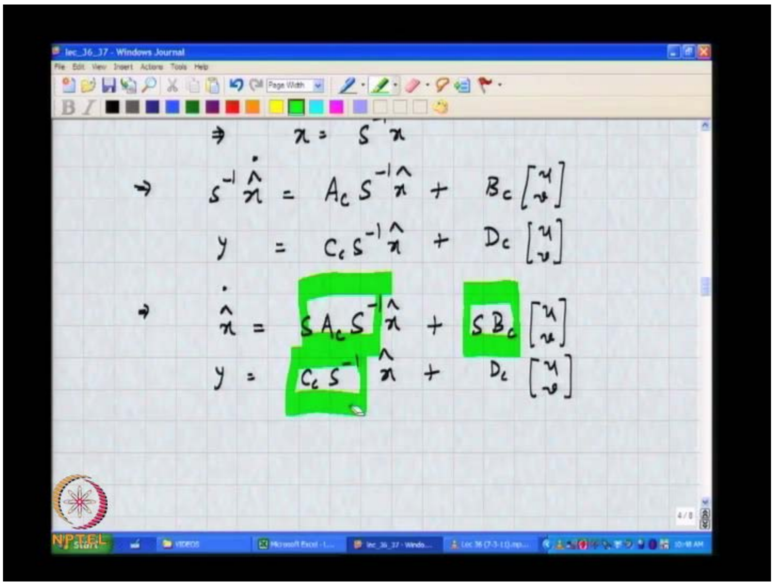

Which means x hat dot is S A c S inverse x hat plus S B c u and v and y is C c S inverse x hat plus D c times u and v. So, this is telling you that changing or modifying the coefficients, is equivalent to multiplying the A the B and the C matrices of the original loop filter. by matrices which depend on the scale factors. And this can be used to systematically go and derive the signal flow graphs for loop filters, where the modulators are scaled.

What we did in our third order example, was in effect use the same thing, but we did this for this particular topology and what we did was, whenever we scaled the output of an integrator by some factor alpha, all nodes or all signals that depend on this output of the integrator, must be scaled by a factor 1 over alpha. So, in our example the output of the first integrator is sensed by this path, which had a gain k 1 that had to be divided down by alpha. And similarly, the second integrator which also depended on the output of the first integrator had also to be scaled on by alpha.

And these equations ((Refer Time: 49:31)) simply represent a formal mathematical expression of the same bands. In the next class, we will look at another important issue namely how does one simulate this loop, for discrete time it is quite straight forward, the loop filter coefficients will describe a set of difference equations. And therefore, one takes the output of the loop filter, quantizes it and feeds it back, on the other hand when you have a continuous time loop filter, the output of the loop filter is...

Student: ((Refer Time: 50:19))

Is in principle a convolution of the input continuous time waveform, and the impulse response of the loop filter, but we are only interested in samples of the output of the loop filter, because that is what is being quantized and fed back. So, the question now is can we use simulators, which are been designed to simulate a discrete time delta sigma modulator, to simulate...

Student: Continuous time

Continuous time delta sigma modulators, I mean the basic idea being that, sure the loop filter is continuous time and all that, but all that I care about finally, is the quantizer output which only depends on samples of the loop filter output. So, since I am not really interested in predicting the entire waveform, may be it is possible to simply the simulation of a continuous time delta sigma modulator, where if I am only interested in samples taken at periodic intervals.

We will see that that is indeed possible and therefore, one can simply use a simulator designed to simulate a discrete time modulator, to also simulate a continuous time modulator. So, the shier tool box for instance can simply be used after a couple of simple manipulations on the, once you know the continuous time loop filter properties, one can derive an equivalent discrete time filter. And use the tool box to determine the properties of the loop, we will see this in the next class.