VLSI Data Conversion Circuits
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Lecture - 17
Quantization Noise – 2

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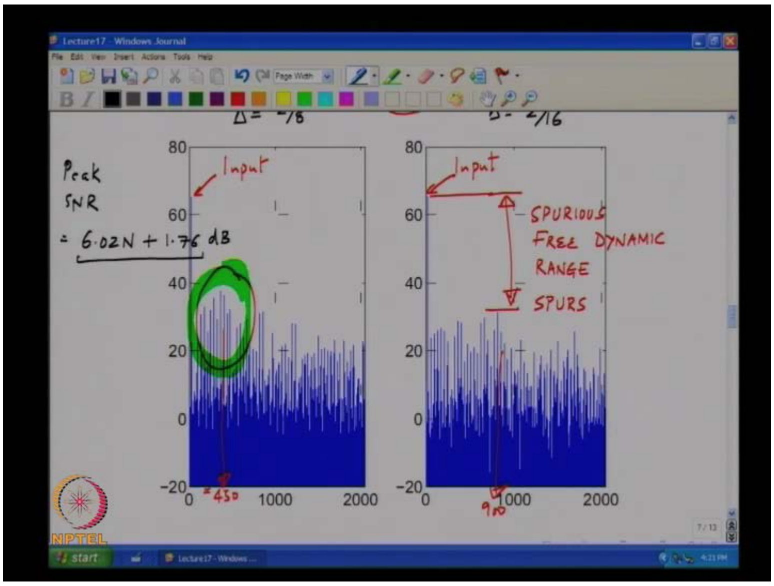

This is VLSI data conversion circuit’s lecture 17; the last class we were looking at the spectra of quantized signals. And, we were trying to get some intuition about or whatever little intuition is possible about these spectra one corresponds to a quantizer with a step size of 2 by 8. And, another 1 corresponds to a quantizer with a step size of 2 by 16. And, the input I told you is a very low frequency tone sitting at 19 by 4096 times f s, and the reason for this weird choice of numbers is we choose a 4096 point f f t, and you want the input to lie on a bin, and I chose 90.

So, these are the inputs in both cases the input is identical, the only change is the number of levels of the quantizer. We all identified that the picture on the right corresponds to a quantizer with a smaller L S B size, and clearly we see that in both cases the spectrum is not quite, the quantization noise spectrum is not quite white or not quite ok. Now, other things that we can discern, where does the peak of the quantization noise spectrum occur here on the left? Around here right, and that turns out corresponds to somewhere around the 450 bin or so, all right. Now, let us see why that makes sense, all right.

Any comments or suggestions on the intuition behind, why the spectrum kind of shows a peak at around the 450 bin, where does the input lie by the way, which bin?

Student: Nineteenth bin.

The input lies in the 19 bin very good. And, we see that there is a lot of energy concentrated around the 450 bin or so, right approximately. The question is this some random thing, or is there some intuition behind this ok.

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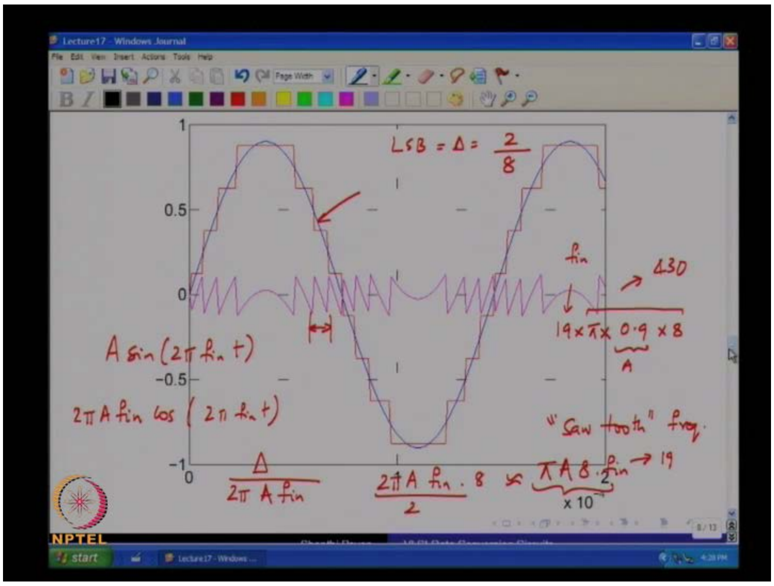

Let us try at this way; the quantization error waveform looks like this. So, as you can see this is a saw tooth waveform, and when you look at the waveform; if I did not shown you the red and the blue curves, and I had just shows you the magenta waveform, which is the quantization error waveform. Looking at it where do you think, it is dominant Fourier components would lay. In other words; where would most a lot of its energy lie intuitively.

Student: ((Refer Time: 04:12)).

Pardon

Student: one by fourth of the second.

1 by

Student: Fourth of the second.

No, why?

Student: Flat the flat portion, we have repeated it at a very high frequency rate.

Correct.

Student: And that rate is approximately equal to number of waves.

Ok.

So,

Student: It is periodic sir it is repeating at.

Yeah. So, very vaguely right we can say that this corresponds to a waveform, which is periodic and what do you think this period is on the average? Clearly the period is changing from tooth to tooth, right. But on the average what do you think that might be. So, these, the L S B size is 2 by 8 correct. So, how long will it take before you have a discontinuity here? How long in time will you have before this comes down?

Student: ((Refer time: 05:32)).

No.

Student: 8 by 1 by 8.

Why?

Student: 1 by 8 f s, 1 by 8 divided into.

No, I mean imagine this you have a sinusoid in continuous time right; you quantize it, all right. The error is this, the saw tooth type waveform, what is you know a good approximation to you know some average period of the saw tooth.

Student: 62 point ((Refer Time: 06:07))

How will you find it? No, I am not worried about the math, but how will you go about finding. Do you understand the question, the question is; this is the let us say for the time being assume the blue wave was a continuous time sinusoid right, am trying to figure out what this period will approximately be, can we get an approximation to that.

Student: 1 by 16 you can get.

How did you get that answer?

Student: 1 by f s by 2 into the number of.

Why?

Student: ((Refer Time: 07:13))

So,

Student: ((Refer Time: 07:17))

Can somebody give me a more carefully constructed argument?

Student: Number of the saw tooth in the wave of sin wave, how many numbers at that ((Refer Time: 07:33)).

Somebody else, I mean how long does the sin wave take to traverse delta, how will you figure that out. You need the slope, is not it. So, the slope is A sin 2 pi f in times t, right. So, the slope must be 2 pi A times cos 2 pi f in times t. So, if you have if the input amplitude is A then, the slope is; the maximum slope is 2 pi A right. Of course, the slope varies in magnitude all the way from 0 to 2 pi A. So, we say roughly I mean we are only interested in order of magnitude calculations. So, we say the slope is 2 pi A. Sorry, 2 pi A times f in correct, right. So, the maximum slope is 2 pi A times f in, which means; that the time taken to traverse 1 L S B will be of the order of delta divided by 2 pi a times f in correct, which means that the frequency of the saw tooth is approximately.

Student: 1 by this.

1 by this, which is 2 pi A times f in divided by delta, correct. And, what is delta it is 2 by 8. So, this must be, this is the saw tooth frequency; please note this is a very approximate calculation. So, you cannot expect to be write to within them, you know second decimal place, you understand. So, what must this tell us this is what it is pi times A times 8 f in. In other words: if f in is in the nineteenth bin right, a large part to the power of the

quantized sinusoid must reside around bin numbers. Nineteen times pi times A, which is 0.9 in this case, at this is A times 8 right. So, please note that this is the input bin, correct. And, what is this quantity can somebody calculates and tell me.

Student: 24

Pardon

Student: There 4500.

Please check carefully, somebody has a calculator or a.

Student: 136, nearly 23 sir 23.6 sir.

19 times pi times 8 times 0.9.

Student: 430.

How much?

Student: 430.

Yeah. So, this is about 430. So and, what did I say I said this 450 or 430

Student: 430.

Right, I will be tempted to say 430, but yeah it is 450. It is what this roughly is I am not surprised I mean now not quite surprisingly the p occurs at around, what we estimate right. So, that is our estimate was about 430 this peak occurs around 450 or so. This is a simple numerical experiment you can try out yourself and convince yourselves that it is indeed a case, you understand.

Student: ((Refer Time: 12:25))

Correct. So, clearly you know this is only an estimate, and we all quite lucky to get you know. So, close to the to the real thing right and so, you can see that some of the maximum slope is what we calculated of course, the slope goes on changing, right. So, it is not as if that waveform will have of you know fundamental frequency at 430, and know it is harmonics it is as you can see since the slope is changing the average frequency is also is not fixed, right. It keeps changing from the average I mean the period is not fixed it keeps changing from saw tooth to saw tooth. So, you can see a distribution around there by it seems to make intuitive sense that it peaks around whatever the four thirtieth or four fiftieth, all right.

Now, when I increase the number of steps by a factor of 2, in other words; if I increase the resolution of the converter by a factor of 2, what you would expect this f max to, you would expect it to double, and indeed that now, make sense right. We see that the peak kind of occurs I will say this is somewhere around 900 or so, right. We estimate 430 for the earlier one, we had estimate 900 for the second one, right. And, you can see that it is indeed the case, all right. So, then you build a practical converter, because of variation in step sizes and all sorts of things, it is again I mean a fact that you will not see a white spectrum with the quantization noise, you will see and all such a spikes they could be coming from distortion, they could also be coming from you know we had non linear interactions within the converter.

But as a person who characterizes whereas, the customer who uses an A to D converter. What one would do, would be to plot the f f t of the output sequence, right. And, we would be in many applications this is an important metric and that is the ratio of the input power to the highest spike that sticks out of the spectrum ok. So, these spikes you know which we interpret as not being part of the noise, but some deterministic tones right, are called spurs. And, this range which is the ratio of the input power to this the power of the spurs, this is called the spurious free dynamic range, all right. Let me again remind you that the peak S N R for an n bit converter is 6.0 2 n plus 1.76 d b beyond, which the S N R will start to fall quite rapidly, because you are over loading the quantizer, right. And, there will be a lot of distortion and stuff. So, the peak S N R is given by this quantity; specifically the dynamic range of a system is defined as the ratio of the largest signal that the system can process to the.

Student: Smallest signal.

To the smallest signal that the system can process, while still giving an acceptable signal to noise ratio.

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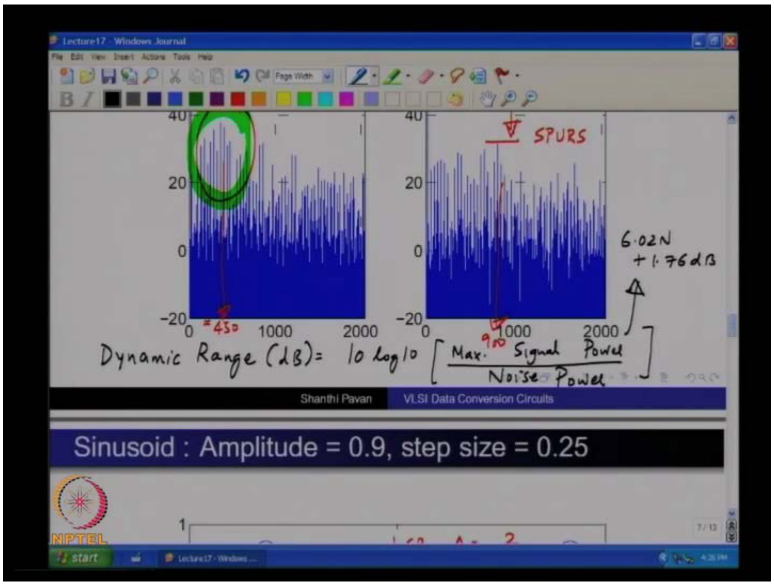

So, the dynamic range in d b is defined as the ratio to the log 10 of course, of the maximum signal power the system can process to the minimum signal power it can process. While being able to give an adequate signal to noise ratio, does it make sense? You understand I mean; a classic example of dynamic range is that of the human ear, right. So, we can hear if the signal acoustic input becomes too small we are not able to hear anything or you may not be able to listen with adequate precision, what the other person is saying, right. So, that gives the lowest that amount of acoustic power is the lowest input signal power that the ear can detect.

On the other hand; if I took to give close to one of the big speakers and rock concert. Obviously, you know your ear would break I mean so, too much of having too much acoustic input is also not good, because beyond a certain point you know things probably saturate inside and go for a toss. So, any system just like this will have a range of inputs over which it can figure out something about the signal, you understand. That range is called the dynamic range and so, again it does not make sense to say is the ratio of maximum signal to minimum signal, right. Because in principle you can put a very small signal also, and the system nothing will happen to it. But it is just that you the system cannot use that input signal in any fashion, because it is getting dominated by internal noise of the system, correct.

So, the dynamic range is defined as the ratio of the largest signal to the ratio of the smallest signal that a system can process with an adequate signal to noise ratio, you understand. So, that signal to noise ratio is often chosen to be 0 degree. In other words the assumption is that; even if the signal power is equal to the noise power, we can still make sense of the signal, this is quite an arbitrary definition, right. And, is commonly used in practice in which case the minimum signal power becomes the same as the noise power, right. In which case for the A to D converter the dynamic range in d b is simply nothing but.

Student: ((Refer Time: 20:58)).

If you have an ideal quantizer, what is the dynamic range of the quantizer in d b? It is 6.02 n plus 1.76 d b, all right. Because the noise power is nothing, but the quantization noise power, signal power is you know assuming a sinusoidal signal and signal with an amplitude equal to one of the quantizer range. And therefore; the dynamic range of the A to D converter is given by the standard peak S N R formula, all right. This spurious free dynamic range is defined as the largest, I mean separation between the input signal and.

Student: Spurious.

Spurious signals which can occur due to you know any number of reasons. And, this is another metric which is common to quote and this becomes important in some applications. For example; in communications it is important that not only is this signal to quantization noise ratio adequate, they must also not be tones sticking out of the spectrum too much. And, the reason is that in wireless for example, you know a narrow band tone is behaves like an input, you understand. Because the spectrum when you I mean sense the spectrum in the channel right, there will be a whole bunch of users and all of these are using small bandwidths. So, a spurs sticking out right, can be confused for a user, you understand.

So, while in some for example, in wireless it is a may not be too bad to have a slightly larger noise floor overall, but spurs sticking out could be a problem, right. For example; base station A to D converters are the classic case in point, all right. So, this is just first re concerned right now, this is just definition, all right. So, this is all the intuition I wanted to give you about a quantizer, and its spectral properties. So, to summarize the quantization error of an ideal quantizer is bounded between minus delta by 2 to delta by

2, increasing the resolution by 1 bit, increases the signal to noise ratio by a factor of 4 which is 6 d b ok.

The quantization error is approximated to be quite in practice, all right. And, when we actually plot the spectrum we see that it is far from white, but that need not detour us because we know that this assumption works. We saw little bit of intuition on why the characteristics of the spectrum makes sense, all right. And, we also said that when we build a real quantizer, in other words; when you build an A to D converter which consists of a sample and hold and I mean a quantizer, and then the true signal to quantization noise ratio will be smaller than what you actually designed it for. And, to characterize the goodness of a converter what you can do is define an effective number of bits.

In other words; you are not fooled by the number of digital outputs coming from the box, but you compute the spectrum, compute the signal to quantization noise ratio, and then compute the effective number of bits as the actual S N R minus 1.76 divided by 6.02. And, all the math and all the analysis and the intuition that went behind choosing the input and the sampling rate properly, for a sample and hold automatically hold for the entire micro state A to D converter ok. So, when you want to characterize the converter; if you can sink the clock and the sampling and the input signal together then, you would choose them such that their the ratio of input to sampling rate is rational number and most commonly the denominator is chosen to be at power of 2 for ease of calculating the f f t.

And, to calculate the signal to quantization noise ratio you then, calculate the signal power of the input as the power in that signal bin, because of the choice of the input signal there will be no leakage. Then, you can you know bunch up everything else and compute ratio of signal to everything else and compute the effective number of bits. If it is not possible to sink the input and clock then you have to use an appropriate spectral window, and repeat the whole process, all right.

Now, when we were discussing sampling and micros criterion and all that in the very beginning of this course; we said that there is a definite advantage to over sampling a signal. For example; if you have a signal with a bandwidth f b Nyquist says that you must only sample it at you only need to sample it at a rate.

Student: 2 f b.

2 f b right; however, we saw that we have sampled it at a rate much higher than 2 f b, there are some advantages. What are the advantages?

Student: ((Refer Time: 27:49))

Pardon,

Student: ((Refer Time: 27:51))

So, one of the big advantages of over sampling which is equal I mean which is sampling a signal over and above the Nyquist state is that the design of the anti alias filter becomes a lot simpler, right. Now, we take a look at another advantage of over sampling and to motivate that properly.

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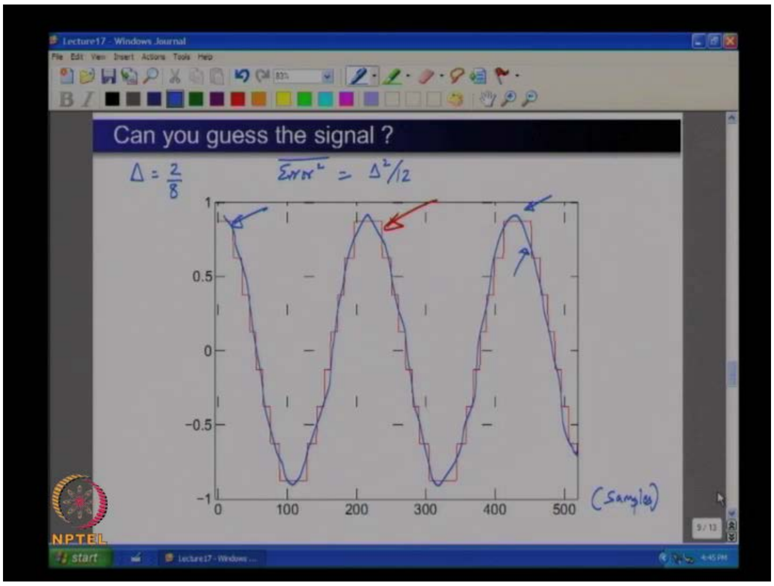

Let me show you this picture and ask you the question; this waveform in red is a sinusoid, which has been quantized or rather as the signal which has been quantized. Can you guess the actual signal, and tell me what all assumptions you have made in your guess work? Yes, can you draw an approximate waveform which you think will give you this output, when it is quantized? Yes, and sure enough if I mean if I gave you a pencil and a paper you had probably say these are likely signal which when quantized looks like this, all right. Clearly there is a big difference between the input and the quantized output, all right. So, I mean. So, visually what have you done, and how are you justified in doing what you have done.

Student: Soft current interpolating eliminating the.

See you have eliminated all the.

Student: Discontinuities.

You have eliminated all the discontinuities, in other words; you have removed all the sharp edges, correct. And, you are saying that the I mean the quantized output, even though the quantized output is has a whole bunch of steps correct, you are saying that the true input correct, is probably not that sharp waveform it is some smoothened version of the of the sharp waveform, please note that; what we are trying to do is to reconstruct, I mean; the eventual problem is we have a bunch of quantized samples, from which we want to figure out what the.

Student: Smoothened input signal.

Input signal is correct now; we are saying that if the quantized output samples they are like direct curve shown here then, the input signal is most likely like the blue curve correct. So, what is this mean and what is the big assumption behind this.

Student: We assume that signal is band limited no aliasing.

And what and of course, it is band limited.

Student: No aliasing sir.

No, we assume that the signal is band limited of course; I mean when we are building an A to D converter, we know that the signal is band limited, right. It is there something else beyond merely band limiting that you can say; on the x axis let me remind you are the sample values. So, these are samples and it looks like a continuous curve, because the samples are so, close by. So, what does this mean?

Student: ((Refer Time: 33:01))

Not quite because we there is this is quantized.

Student: Correlation between 1 sample and other sample is used.

Yes, which means that?

Student: Sampling rate is very high.

Sampling rate is very high in relation to the signal or we can get the signal rate is I mean or the signal is very.

Student: Slow.

Or the signal is varying very slowly with respect to the sampling rate; I mean unknowingly you made that assumption, when you do this curve correct, you understand. So, in other words what one can say is that if the input is varying very slowly with respect to the sampling rate. Or in other words; in more technical terms if the signal is highly over sampled correct then, looking at simply quantizing the quantized output right, will give you I mean what do you think will be the mean square error between the quantized output and the true input signal. In this example delta happens to be 2 by 8, do you follow the question. What do you think is the, I mean clearly the output of the quantizer is a signal is the sequence which is not the same as the input. So, between the 2 there is an error, what do you think the mean square value of this error is.

Student: ((Refer Time: 34:39))

How much you have what would that be.

It will be delta square by 12. So, the mean square error will be delta square by 12, correct. And, that mind you is the difference between this waveform in red, and the true input which produce this waveform, correct. However, by the arguments we just had you were telling me that you can give me a much better approximation to the input, and how did you give me a much better approximation to the input.

Student: ((Refer Time: 35:28)).

By,

Student: ((Refer Time: 35:31)) over sampling.

No, this is an over sampled signal lie we all understand that, correct.

No, do you do you understand what I am asking. So, we said that the signal is over sampled correct, and the output at I take this signal quantize it, with a quantizer with a delta of 2 by8. And, give me the samples and they look like the red curve shown here, and we all agree that the mean square error between the input and the quantized output is delta square by 12, where delta is 2 by 8. Now, you also said that given these samples clearly; you know the, I mean I can give you a much better approximation to the true input. Since, you have told me that the input is over sampled, you understand.

This is all what is implied when you when you do that curve correct, and why did that argument makes sense; we know that if the signal is over sampled it cannot vary rapidly from sample to sample, which means; that the true waveform is probably something like this, correct. It cannot be this jagged waveform. Does it make sense, all right please note that the blue curve here is not the true input it is.

Student: Approximate.

It is an approximate input that you come up with, all right. Based on the quantized samples and what have you done with the quantized samples to generate this blue wave this blue curve. You have taken a waveform with jagged edges and made it smooth. So, what does it I mean correspond to in and technical terms?

Student: eliminated the Nyquist.

Pardon, which is equivalent to?

Student: Low pass, interpolator integral into.

Now, no interpolation is not right, interpolation is where you increase the number of samples that is not what I am doing right, what have I done.

Student: Average.

Which is can you give me a more technical term for average.

Student: Filtering.

Averaging is in general.

Student: Integration.

No you are going backwards now right, I mean averaging is nothing but a special case of filtering. I mean what kind of filtering?

Student: Low pass filter.

Low pass filter, you understand. So, basically the visually what you have drawn right, can be expressed in technical terms as the following. Given an over sampled signal which has been quantized right. The error between my reconstructed signal given the quantized samples and the true signal can actually be made much smaller than delta square by 12 by simply the act of.

Student: Low pass filter.

Low pass filter is this clear, you understand. The aim of quantization is from the quantized samples go and figure out the input. If we did not know anything about the input signal then, you know we quantize and that is the end of the story. The mean square value of the quantization error will be delta square by 12. However, if I gave you the additional information that the signal is highly over sampled it means; that the signal is not changing much from sample to sample, which means; immediately while looking at the quantized samples you can say hey this does not make sense because the quantized output is jumping up correct, which cannot be the input signal. So, I can give you a better estimate of the input signal by simply smoothening out this.

Student: Discontinuities.

These discontinuities, which in technical terms is low pass filtering the quantized sequence, correct. And, if I give you a better approximation to the input in technical terms; the mean square error between the input and the output has.

Student: Decreased.

Has reduced, you understand. So, in other words, all right.

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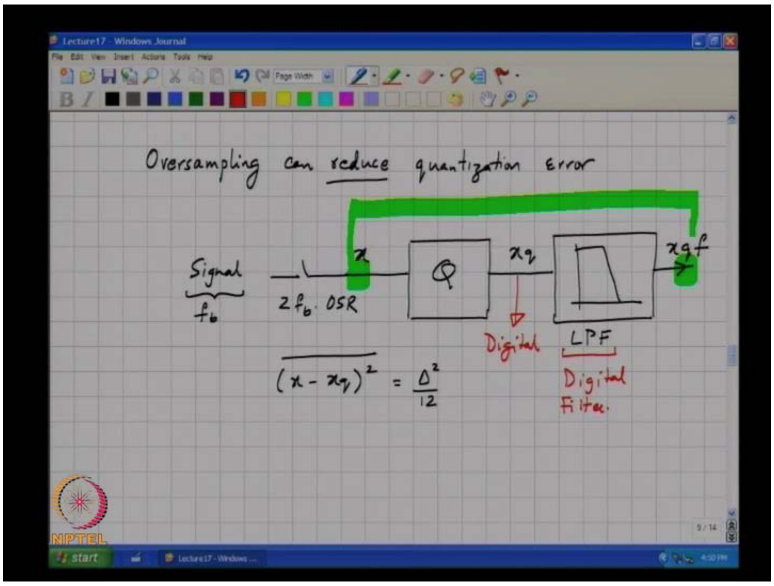

Does it make sense, and if we draw block diagram of what we did? Let us say the signal has a band width which is f b and I sample it not at the micro state which happens to be 2 f b, but a ratio much higher than this. T he ratio of the sampling rate to my signal band width or twice the signal band width is the ratio over the micro state at which I am sampling. So, this is called the.

Student: Over sampling.

Over sampling ratio of the O S R, correct. So, if I sample it at 2 f b times O S R and then, quantizes it, right. If I did not do anything, in other words; simply dump processing of the quantized samples will have a big error between. So, if I call this x and x q mean square value of x minus x q is delta square by 12. But by smoothening it out which is in general putting a low pass filter, the sequence I get out will be a lot better approximation to, in other words; if I call this x q filtered x q f and x will be very close to each other, right. In other words; the mean square error between x and x q f will be.

Student: 10 times smaller.

Much smaller than delta square by 12 correct, and this all common sense and intuition right, there is nothing new about this at all you knew this all along, right. It is just that now you added I mean you represented it as a block diagram, and there is jargon right, there is OSR, there is a digital low pass filter and all this stuff, you understand. By the way this sequence I mean; the signal is in what form here.

Student: Discrete time.

It is discrete time, but how many levels are there? Discrete number of levels 2 so, what signal is this?

Student: Digital

This is a digital signal, because you can represent it by a finite word length. So, what kind of filter is this? This is a digital filter. So, let us do some numerical experiments. And see what is going on.

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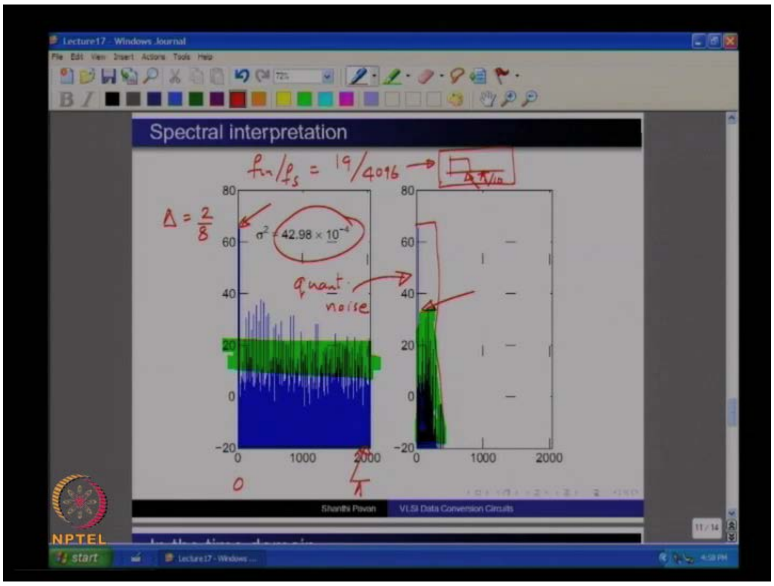

The spectrum on the left shows a low frequency signal again, f in by f s is 19 by 4096. And therefore; the input is lying somewhere here, and as you can see this is a case of definitely a lot of over sampling, correct. What do you think is the over sampling ratio roughly.

Student: 2200.

No,

Student: 22 that comprehends around 100.

Around 100 correct, because if the input tone was 2048 by 4096 here at Nyquist, correct. So, we are at 19, which basically means; that the sampling rate is roughly 2048 by the over sampling ratio is 2048 by 19, which is about 2000 by 20, which is about 100. Now, I have taken this and this is quantization noise, and I have taken a delta of 2 by 8, all right. Now, once I have this I mean this quantized output; I have gone it smoothed it, right. And, smoothing as we all agreed was is a special I mean is basically low pass filtering, I have taken this and pass this through a butter worth low pass filter, which is very sharp do not worry about the exact order and stuff.

All that I am saying is that we have taken a very sharp filter, and the filters band edge happens to be pi by 10, I mean f s by 2 corresponds to pi all right. So, f s by 20 corresponds to pi by 10. And, if I look at the out the spectrum at the output of the butter worth filter, sure enough it is gone in cut off all frequencies beyond this point. And, the resulting time domain I mean the resulting sequence has a spectrum like this. In the spectral domain, it is immediately obvious that the sequence after filtering will have a lot smaller.

Student: Noise.

Quantization noise, because it now corresponds to only this part. If we made a uniform I mean an assumption about quantization noise being white, how much do you think the quantization noise power would have gone down by? In other words; if this was assumed to be flat with frequency

Student: ((Refer Time: 49:18))

Pardon, earlier this was going all the way from 0 to pi correct. Now, you have taken a low pass filter and filtered of all stuff beyond p i by 10, correct. So, how much noise power you know remains? One tenth of the original noise power correct so, I mean this is immediately apparent when you talk about it in the frequency domain, right. In the time domain do we know that the noise power must reduce, we do not know how much it must reduce by ok.

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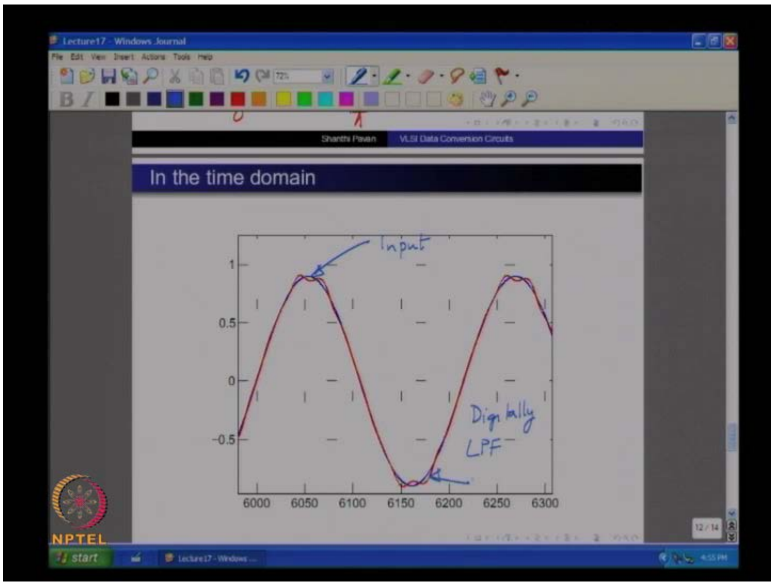

So, this for example; is the curve in blue is the input, and the curve in red is the, what do you think it is?

Student: Low pass filter.

Low pass filter, let me add the important statement digitally low pass filtered. Why because; the input is at digital sequence now, because there are finite numbers of levels in amplitude as well as in time. So, the digitally low pass filtered sequence when I plot looks like the red curve, and clearly the error between the true input and the digitally low pass filtered version of the quantized over sampled signal is lot smaller than delta square by 12, right. Which was what 1 would get we took that jagged waveform and then computed the mean square error, correct.

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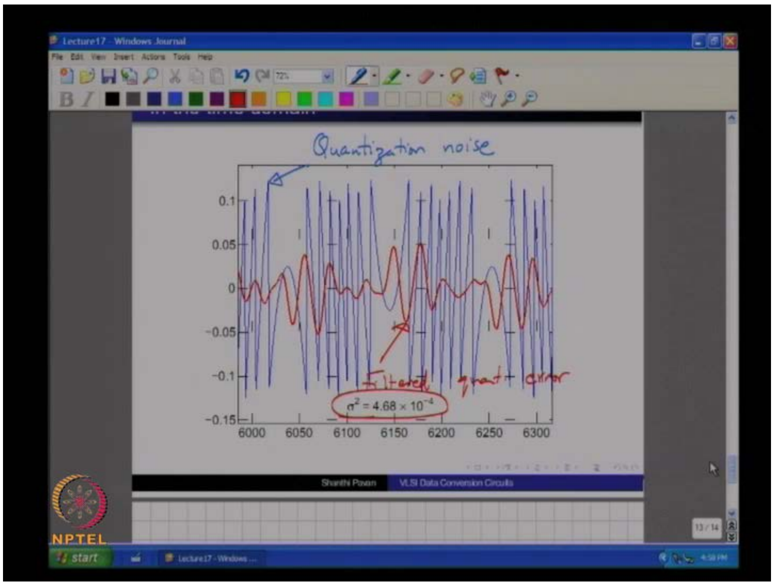

And, to give you an idea of the actual error waveform; the jagged waveform here is the quantization noise. And, there are big jumps, and that make sense the moment you smooth things out clearly you know coupling of things are apparent one the error is continuous, right. We call the low pass filtering or smoothing and more importantly what can you say about the amplitude of the error? The amplitude of the error is greatly reduced indicating that the power is the power of the mean square value of the error is, has reduced considerably, you understand.

So, this is the filtered quantization error, or you can think of it as the mean square error between the filtered waveform and the input, and that turns out to be 4.68 into I mean in this particular example it turns out to be 4.68 into 10 power minus 4. Whereas, in the unfiltered case; I computed it to be 43 into 10 power minus 4 the ratio is approximately 10. So, one is 4.3 into 10 power minus 4, the other one is sorry, 1.43 into 10 power minus 4, the other one is 4.6 into 10 power minus 4 is about a factor of 10. And, that make sense because you taken a filter with a band width of pi by 10, you understand.

So, the moral of the whole story is if we know that the signal is over sampled then, we can increase the effective resolution of the quantizer, correct. I mean; this is basically we have reduced the quantization noise by a factor of 10 here; you could have obtained the same effect by not, low pass filtering, but using a better quantizer instead correct. So, in other words; the act of low pass filtering the quantized signal after over sampling it has made has increased the effective resolution of the quantizer. In this particular case approximately we have how many bits? Please, recall that error going down by a factor of 4 is equivalent to.

Student: 1 bit

1 bit correct. So, this is gone down by.

Student: 10.

By 10 which is you know between 1 and 2 bits, correct. 1 bit improvement is 4 x lower, 2 bits is 16 x lower correct, you understand. So, this is10 so, it is somewhere between 1 bit and 2 bits. In other words: the effective resolution of the quantizer even though we have a delta which is you know 2 by 8, after digital low pass filtering the effective resolution is actually the L SB size is actually smaller, thereby improving the resolution of the quantizer. So, this is a way of taking a poor quantizer and making it look better. And, the trick that is doing this whole thing is over sampling, and you must remember of course, it does not help to simply over sample.

Student: Low pass filter.

The key thing to do is to low pass filter it after you.

Student: Quantize it.

After you quantize it, you understand. So, this is a prelude to the discussion that will follow from now on. Where we look at over sampling converters which work on this principle where you take a very cheesy basic quantizer and improve its resolution using over sampling techniques. And, it turns out that in the limit right, you can go on reducing the I mean if you look at this example; we have taken a 3 bit quantizer and made it look like a four and half let us say bit quantizer by over sampling. In other words; we contain the arguments upside down and say; if we wanted a 3 bit quantizer in the first place you could have started off with a poorer quantizer right, may be two and half bit, you know I mean so, one and half bit. So, quantizer, you understand.

So, you can take this to the limit, and say if I go on increasing the over sampling then, to get an effective quantizer resolution I can use a, you know lower and lower precision quantizer. I mean; it does not eventually mean that you go on over sampling and then, you can get it out of the quantizer all together, right. You need a quantizer only that can become poorer and poorer and what is the lowest resolution you can go.

Student: 1 bit.

1 bit. So, you will see that going forward that using over sampling we can make a 1 bit quantizer look like a 16 bit quantizer, using over sampling techniques. It seems amazing, but it is true and it is a, you know used all the time and these are the family of converters called over sampling.

Thank you.