Digital Signal Processing
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Lecture 71:
Sampling (4)
- Is oversampling beneficial?

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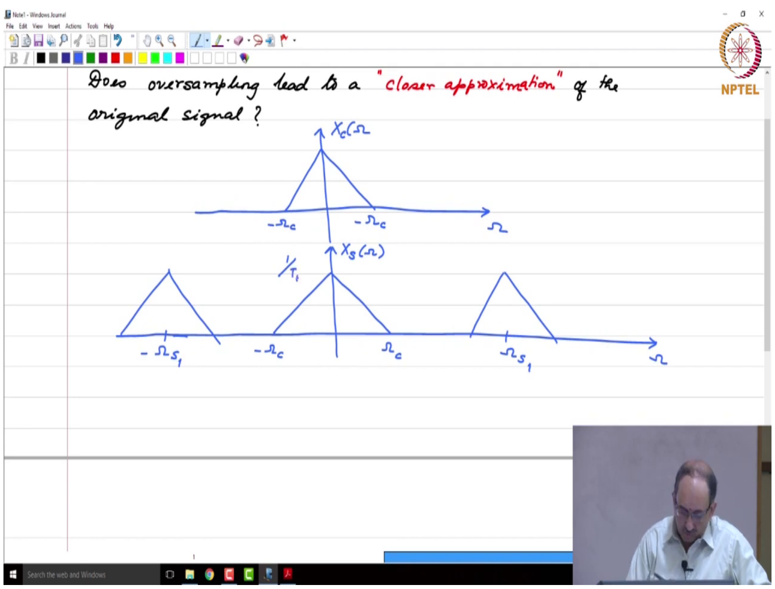

Let us continue to look at some more aspects of sampling. So, in practice, all signals are time limited and hence they are band unlimited. So, what we do is, we put an anti-aliasing filter to band limit the signal and then sample it. So, what we loose is, the components of the input beyond the band width of the band limiting filter.

And typically, since for all practical signals the spectrum falls off beyond a certain point, the information there, even if you cut them off, it is not going to matter much. And hence, we can afford to band limit the signal and yet not lose much information or perceptible information. But, if we do not look at this from this angle, if we just look at it from an intuitive and naive point of view, we may think that, if we sample at a higher and higher frequency, we may think that we are approaching the underlying continuous time signal better and better.

Or we may think, we are doing a better job of approximating the underlying continuous time signal better. So, this is the viewpoint that you may have if you just look at it naively. Let us see whether this is true and what the implications are of over sampling. So, the question that we want to ask is, does over sampling lead to a closer approximation of the original signal? So, this is the question that we want to ask. Now, let us look at the signal, its sampled spectrum for two different sampling rates. And then we look at the corresponding DTFT. So, we have this signal. So, this is cap omega.

So, this is $X_c(\Omega)$ and this is between $-\Omega_c$ and $+\Omega_c$. And what happens when we sample, we have seen this quite a few times now. This thing periodically repeats. So, this is the spectrum of $X_s(\Omega)$. So, this is still $-\Omega_c$ to Ω_c and this repetition occurs at intervals of Ω_s. Let me know call this as Ω_{s_1}, because we are going to compare this with a different sampling frequency, namely Ω_{s_2}. Therefore, this becomes $1/T_1$.

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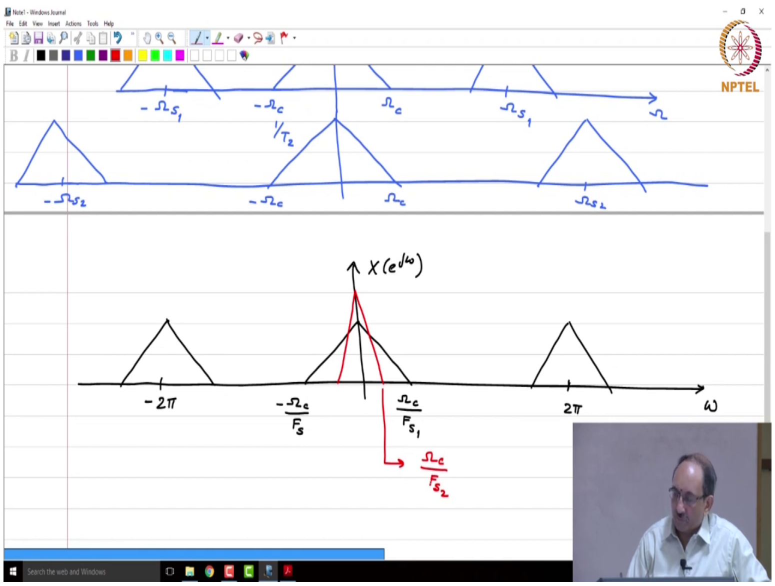

And, now if you compare this with a sampling frequency that is larger, so this is $-\Omega_c$ and $+\Omega_c$. So, this is Ω_{s_2} and this is $-\Omega_{s_2}$ and this is $1/T_2$. So, clearly T_2 is smaller because Ω_{s_2} is larger. Even though I have drawn them of the same height, these are of different heights because this is scaled by $1/T_1$; this is scaled by $1/T_2$ and T_2 is a smaller quantity.

Now, let us focus on the corresponding DTFT. Remember, the DTFT is 2π periodic. So, you have this periodic repetition , this of course is 2π because this is ω and this is $X(e^{j\omega})$. Now, this frequency is Ω_c/Ω_{s_1}, because any frequency in the analog domain when you map it to the DTFT, you have to scale it by F_{s_1} rather.

So, this is actually Ω_c/F_{s_1}, $-\Omega_c/F_{s_1}$. And now you have to compare this with the higher sampling frequency, namely Ω_{s_2}. Therefore, what happens in this case is two things. One the amplitude is going to be larger because it is $1/T_2$, which is a larger number. Second, you are going to divide Ω_c by F_{s_2} and F_{s_2} is a larger quantity. And hence, the spectrum looks like this. And this point is Ω_c/F_{s_2} and hence this is a smaller quantity.

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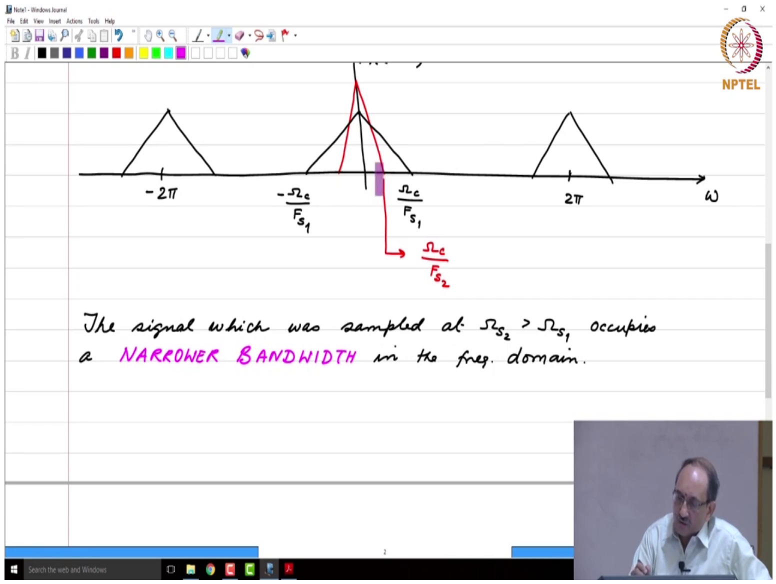

Now, if you look at this spectrum, the signal which was sampled at a higher rate, namely Ω_{s_2} which is greater than Ω_{s_1}, occupies a narrower bandwidth in the frequency domain. So, this is the key point, whereby frequency domain we mean the DTFT domain. And, hence higher sampling rates lead to signals with spectrum that are narrower.

So, what you might ask. Let us for example, consider the practical case where signal is say corrupted by noise. And, once you have received the signal, you want to clean it up. So, what you would do is, to clean up the received signal, you will put a low pass filter and you will put a low pass filter with a cut off frequency of Ω_c/F_{s_1} in the first case.

So, what it will do is, it will pass everything in between $-\Omega_c/F_{s_1}$ to Ω_c/F_{s_1} in the first case and cut out everything else. So, what the low pass filter does is, it eliminates what is called as the out of band noise. It passes everything between $-\Omega_c/F_{s_1}$ to Ω_c/F_{s_1}. So, in that region, both the signal is there and the noise is there, noise will occupy the entire frequency spectrum typically.

And by this low pass filter, you are eliminating the out of band noise and hence you are cleaning up the signal which is good. Now, in the second case, if you want to again clean up the signal, second case corresponds to the higher sampling frequency signal. If you did that, now you have to design a filter that has to cut out frequencies beyond $-\Omega_c/F_{s_2}$ to Ω_c/F_{s_2}, that is the in-band part.

And, you are going to reject everything that is out of band in the second case. But the low pass filter in the second case, do you need a narrower bandwidth low pass filter or a broader bandwidth low pass filter? You clearly need a narrower band low pass filter. Therefore, as far as the discrete time signal is concerned, the specifications on the filter are most stringent. You need to design a filter that is of much narrower bandwidth, because the higher frequency sampled signal occupies a low narrower bandwidth in the discrete time spectrum domain.

And hence the requirements on the filter are much larger. So, this is very very important. Not only that, remember, when you are implementing this on a DSP processor. If you thought that the, in the first case, samples are coming at you at a fast and furious space. In the second case, they are coming at you at a faster and more furious pace, Fast and Furious 1 and Fast and Furious 2.

So, your requirement on the processing speed is also more stringent, you need a processor with higher MIPS. So, you are only making things on you much more difficult by over sampling. Therefore, typically, what you would want to do in the discrete time case when you are sampling signals is, you want to sample them just adequately.

By over sampling, you are not going to gain anything. In fact, you are going to make things on yourself much more difficult, by requiring to design filters that are more stringent and hardware that has more processing power. So, you should almost always never over sample, if over sampling is not needed. Yes, question.

Student: (Refer Time: 12:31) transition bandwidth (Refer Time: 12:33) much wider

Student: (Refer Time: 12:35).

Yeah, the transition bandwidth can be wider, but then you will have more out of band noise because noise is going to occupy the full spectrum typically. Therefore, if you have the same filter that you use as in the first case, sure the signal will pass through. But what will happen is, he used the filter in the first case, noise that was occurring between this frequency and this frequency will also pass through.

If you use the filter that you use for the lower sample signal therefore, you will have more outer band noise. The signal is only between this frequency and this frequency. By relaxing the constraints of the filter, you are going to pass noise in this range. And hence, your output SNR will be poorer. That is you are accommodating more out of band noise than what you would want to do.

Student: (Refer Time: 13:37).

Yes.

Student: (Refer Time: 13:39) noise (Refer Time: 13:39).

Typically noise will have a white spectrum.

Student: (Refer Time: 13:45) higher sampling frequency (Refer Time: 13:49).

So, always you will put a band limiting filter. So, even in the continuous time case, noise occupies all frequencies, I mean thermal noise goes to very high frequency. Therefore, moment you put in band limiting filter, the noise also is band limited and then you sample. And in the discrete time domain, noise occupies the whole frequency range between -2π to 2π which is the most typical case that is encountered. So, once that is there, as the typical case we are talking what happens in these instances.

Student: Sir, we will go for (Refer Time: 14:31).

No, remember, you should not talk about aliasing here as far as noise is concerned, because noise typically has a very wide bandwidth. A moment you put in an anti-aliasing filter which is part of any ADC, then out of band noise will get rejected. That is, noise outside that band limiting filter will get rejected.

So, noise also is band limitted and hence if you sample at, what frequency we are going to sample? We are not going to cost aliasing as far as noise is concerned. And you need to keep this in mind, because if you are not aware of these things or if you are not a technical person who knows all of this, marketing people will say somebody will say I will give you 48 kilo Hertz sampling.

If it is voice, you do not need 48 kilo Hertz at all, 16 kilo Hertz is plenty as far as voice is concerned.

Because, beyond 8 kilo Hertz there is nothing much in terms of signal content information and telephone speech is 4 kilo Hertz. And even that is except for high frequencies like s and f, which you cannot distinguish most of the informations there that is present.

And 8 kilo Hertz is what is called as wideband speech, everything is perfectly fine. And hence 16 kilo Hertz sampling is plenty. Only if it comes in music, you need higher sampling frequency. Anyway, remember, the final arbiter is the ear and we cannot hear beyond 20 kilo Hertz. So, this is assuming ideal upper limit. So, this is assuming your ear is not glued to an iPod speaker all the time.

So, this is assuming normal ears. So, I always tell, if you get your ears checked by an audiologist, you will be shocked at the kinds of hearing loss many have especially those who have the iPod things stuck in the ears all the time. If you read the fine print there, you will see the kind of damage it can cause. So, they put all those fine print, so that they are clear legally.

They put it in fine print because you will not read them anyway, right. So, you will very quickly starts to suffer hearing loss. So, what you will do is you will crank up the volume, it will cause damage even more. And then in response to that, you will cause the increase in volume even more and it is a vicious cycle. So, what has been shown is, continuous exposure to even 75 dB of sound pressure level can cause damage to the ears.

And 75 dB is loud conversation, normal conversation is 60 to 65, 75 d B is really loud conversation. And, this was at least thinking some time back. Now, I think researchers are thinking even continuous exposure to sound at 65 dB sound pressure level can cause damage. So, my ears for example, are very sensitive, I can keep the TV volume really low and still here.

Because, I do not listen to, I do not have iPod earplugs stuck in my ear all the time. But, may be in so many occasions when I am here next person is wearing a earphone and I can hear what that person is hearing. It is so loud that even I can hear person who is sitting next. I cannot imagine the kind of sound pressure level some of you are exposing yourself to by using this.

And of course, we always have Sarang and I always say there you go to these rock festivals, you are not really hear the music but also feel the music. So, that is like 100 120 dB, which is close to threshold of pain, but you are so much enjoying the music you do not feel the pain I guess. So, the reason I am saying all this is the final arbiter of all this is the ear.

And beyond a certain point, you cannot hear any of this. And hence if your highest frequency content as far as voice is concerned is 8 or 10 kilo Hertz, 20 kilo Hertz should be more than fine. And 16 kilo Hertz sampling is very adequate for voice and slightly more for music. But somebody might say I will give you 48 kilo Hertz sampling frequency.

Then the competitor will say he gives you only 48, I will give you 96; other person will come and say I will give you 192 kilo Hertz. So, if 96 is better than 48, 192 has to be four times as better, right. So, but all of this is nonsense, this extreme over sampling apart from increasing the storage space of the recorded sound does not convey anything at all and it only makes processing more difficult, more stringent.

So, once you know basics of sampling, any over sampling is not good at all in most cases. We have delta sigma converters over sample those are all special cases, where the number of bits is less and there you over sample for a reason. But if you are assigning adequate bits to these samples, over sampling is very bad and almost always should be avoided. So, you should let the signal after sampling to occupy almost the full range.

Because, if you over sample and compress the spectrum, you are only creating kind of empty space, there is no information here. And, making this narrower and narrower is absolutely no good at all, other than causing more storage space most process requiring more processing power and so on.

So, after sampling, if the signal occupies the full bandwidth or nearly so, that contains all the information. Because you have band limited the signal to a certain frequency say $F_s/2$, and then you are sampling it at F_s, which means there is no loss of information. There is no benefit in over sampling. So, this is very important to understand.