Introduction to Time - Varying Electrical Networks
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Motivation to learn about time - varying circuits and systems: Part 2

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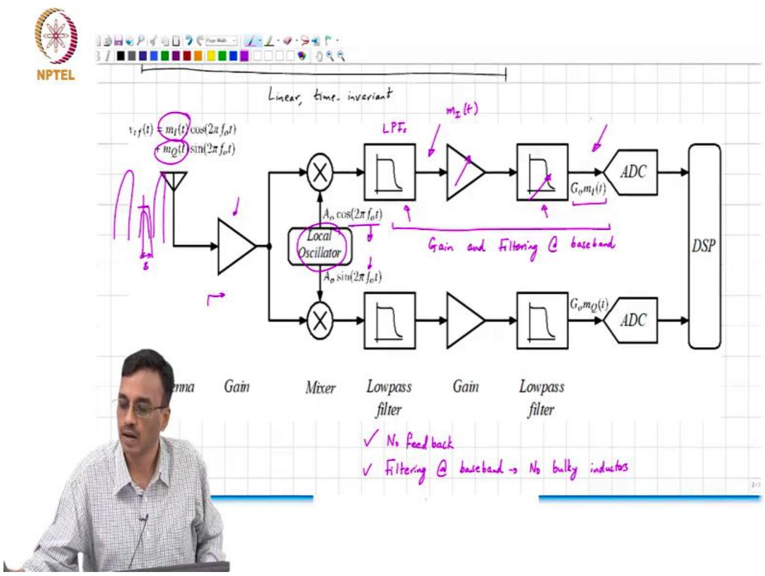

So, this is an example of a modern radio receiver. So, this is the rf signal and the rf signal is got signals modulated on both the sine on the cosine and the sine carriers and in other words this makes the spectrum asymmetric about or frequency about f naught and the idea behind this is to be able to efficiently use spectrum. Now, you have the antenna and some little bit of amplification. Then, as soon as possible you multiply the signal with what is called the called a local oscillator, which generates f naught which is the center frequency.

The moment you multiply this by cos and sin and these are low pass filters, so what comment can you make about the signal that you get here? Which of these baseband signals do you get there?

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Professor: Mi that is supposed to be i actually but never mind and then you have some more gain and some more filtering if necessary and so basically at the end of the baseband chain you basically get some gain times mi. Remember that the rf signal is still got you know desired signal and a whole bunch of interferer, where are the interferers getting knocked off here? What is happening, I mean so what is happening to the spectrum the moment you multiply it by a cosine what happens to the spectrum?

The desired I mean the whole rf spectrum shifts to this whole thing is moved to 0 and another one is move to 2 f naught. The 2 f naught anyway gets is gone because of the high frequency attenuation of this low pass filter, but even so if you make the low pass filter have only a bandwidth of the desired signal namely b, then all the interferers are basically eliminated by now eliminated by low pass filters rather than band pass filters. And do you think it is easier to make a low pass filter or a band pass filter?

Is it easier to make a sharp low pass filter or a sharp band pass filter? I mean you have seen this in fact in one of your assignments, basically as the frequency of the band pass filter becomes higher and if you want to receive a signal bandwidth, a certain signal bandwidth at a much higher frequency, you have a double whammy not only is the center frequency higher, so any band pass filter you make becomes more difficult to do. The q also becomes higher because center frequency is higher, bandwidth remains the same, so q is higher.

So, making something very high q is reliably is very difficult. So, it is much easier to filter using low pass filters and now these low pass filters as you already seen in this course can be realized very efficiently without the use of inductors by using op-amps r and c. So, basically this is the baseband chain so most of the gain and filtering is obtained at baseband. Now at low frequencies, the isolation you can get between circuits is between nodes is much much higher than the isolation you can get at rf.

So, the fact that you have a lot of gain at baseband is no longer problematic because well the isolation you can get is several orders of magnitude higher. And therefore, feedback is not a problem and filtering happens at baseband so no bulky inductors. Now what else what comment can we make about the center frequency, earlier remember in the tuned rf receiver, we had to tune the center frequency of each one of those band pass filters separately and make sure they are all kind of exactly the same.

Here what determines the center frequency of the effective band pass filter that you see from here, it is the center frequency of there is the oscillator frequency and it turns out that you can make the oscillator frequency very very precise by using phase lock techniques.

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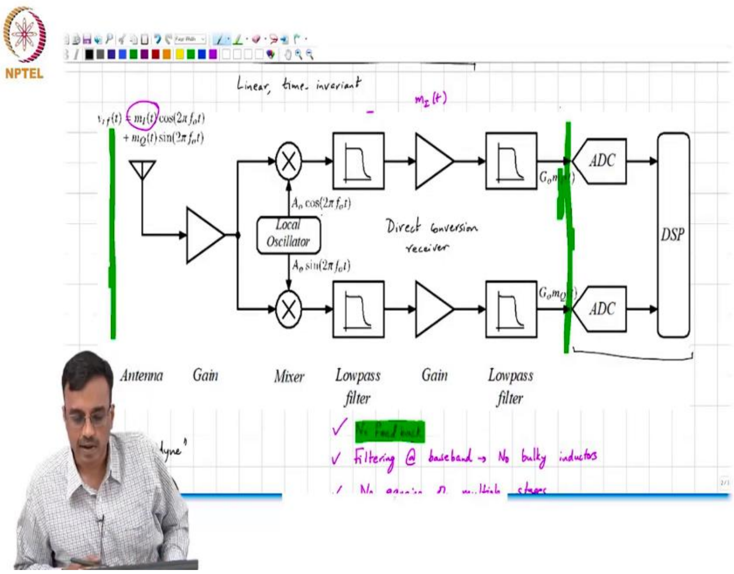

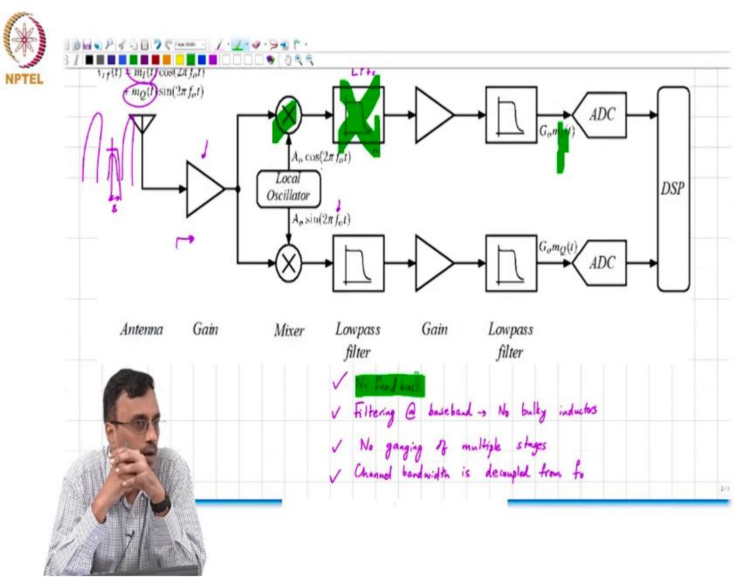

So, the center frequency basically this no need for ganging of multiple stages. What determines the and remember one more problem with the tuned rf receiver was when we changed the center frequency, the bandwidth of the band pass filter would change because q remains the same, omega naught changes. Here what determines the effective bandwidth of the band pass filter that we see? It is the bandwidth of the low pass filter which is responsible for selecting the channel that you want. So, the channel bandwidth is decoupled from the center frequency.

And the next thing is what happens if I mean you are right there is no the I mean feedback in the baseband chain is not a problem, what about feedback from here to here? Well this is a baseband signal, so even if it some of it couples to the antenna this is basically this will go through and you know get down converted. So, if this is baseband once you multiply it up by cos what happens it goes to f naught and then the low pass filter will go and kill it completely.

So, effectively because the baseband signal and rf signal are widely separated in frequency there is the feedback loop is essentially broken. So, there is no spurious feedback which can kill us because most of the gain and selectivity is obtained at baseband where isolation is fundamentally much much higher.

The center frequency of the effective band pass filter that you are making is set by the frequency of an oscillator it can be changed to whatever value you want without having to bother about, well I have 10 band pass filter stages you know how do I make sure that all those center frequencies are the same? And there is therefore all the problems associated with mechanical tolerances of components etcetera is all gone. The selectivity is obtained at baseband and it is much easier to make a sharp low pass filter than it is to make a sharp high frequency band pass filter and the channel bandwidth is decoupled from correct.

So, what is the key to the operation of the modern radio and what is the difference between what the tuned rf receiver and this receiver? The tuned rf receiver from the antenna to the final detector was a linear time invariant system. What comment can we make about the system from here to here?

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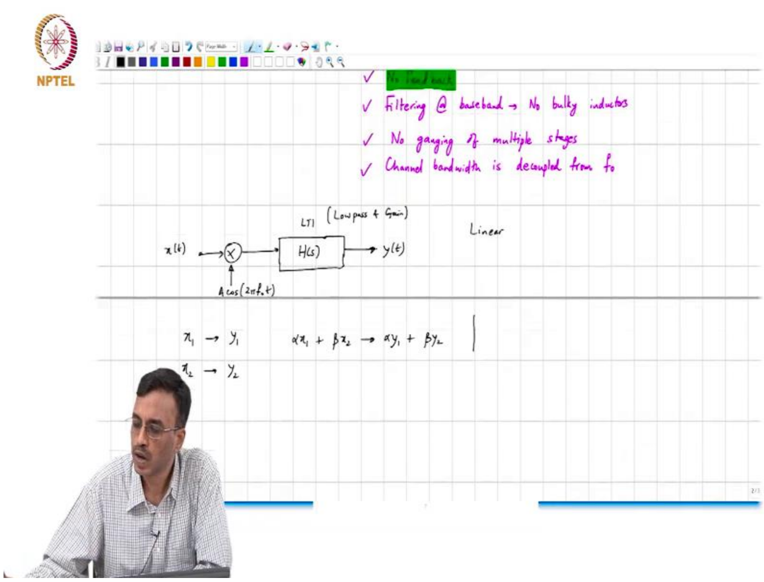

Well, if you kind of look at one of these chains you can see that it is you have the rf signal here, then we have some kind of multiplier where it gets multiplied by some a cos say 2 pi f naught t, then there is a lot of gain and filtering I am going to represent that by a single block diagram. So, I will call that H of s which is a linear time invariant system and this is the output. This is low pass and gain. So, what kind of system is this? Is this linear?

Well, even though you see the multiplier and you might first be tempted to think that it is non-linear. Remember that linearity just simply means that if you get if you put in x1 and you get y1 and you get a put in x2 and you get y2, when you put in alpha x1 plus beta x2 you should get alpha y1 plus beta y2. That is clearly satisfied in the case of a multiplier.

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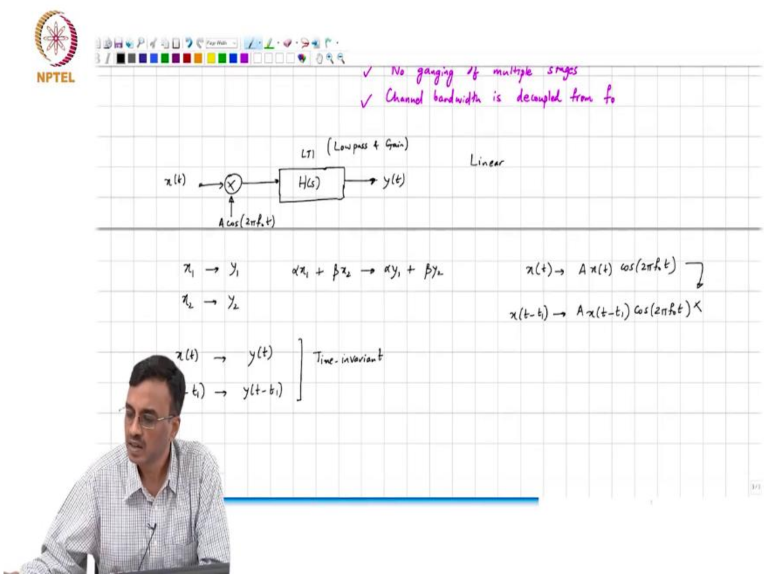

So, this is a linear for sure. What comment can we make about time invariance? What is let us remind ourselves what time invariance means? So, if you put x of t let us say you get y of t, if you put in an input x of t minus t1, you must get y of t minus t1 and then this is what you call a time invariant system and please note that time invariance has got nothing to do with linearity. These are two fundamentally different things.

Now is this system time invariant or time varying? Time varying, because if you put in x of t and you look at the signal here at the output of the multiplier you basically get x of t, a times x of t, times cos 2 pi f naught t, if you put x of t minus t1, then you get a times x of t minus t1 times cos

2 pi f naught t and clearly this is if you delay this by t1 you do not get that. So, this is multiplication is a linear but time varying operation.

So, as you can see the key point that makes the modern radio receiver work is there is a fundamental difference between the tuned rf receiver and the modern radio receiver and if you kind of want to think of one key feature that makes it work, it is the exploiting of time variance. So, these were called I mean so this whole multiplying by the local oscillator was this is called heterodyning and I mean this is not the only way you can make a radio exploiting time invariance.

I mean there is no reason for instance to go all the way to baseband you can go to what is called an intermediate frequency and from the intermediate frequency you can go to baseband, I mean each one has got its own trade-offs but if you open up you know your cell phone today you have two wireless lan receivers, two bluetooth receivers, multiple cellular band receivers and so on and all of them are pretty much based on this radio architecture which is called the direct conversion architecture.

This part is what you call I would call it more technology enabled in the sense that, well 100 years ago making A to D converters and DSPs was not possible, now it is and therefore you exploit that to use more and more complicated exotic signal processing algorithms to make sure that you are as spectrally efficient as possible. So, that the telecom companies can make money. So, the moral of this entire story is that well the key to making the modern radio receiver work is time variance.

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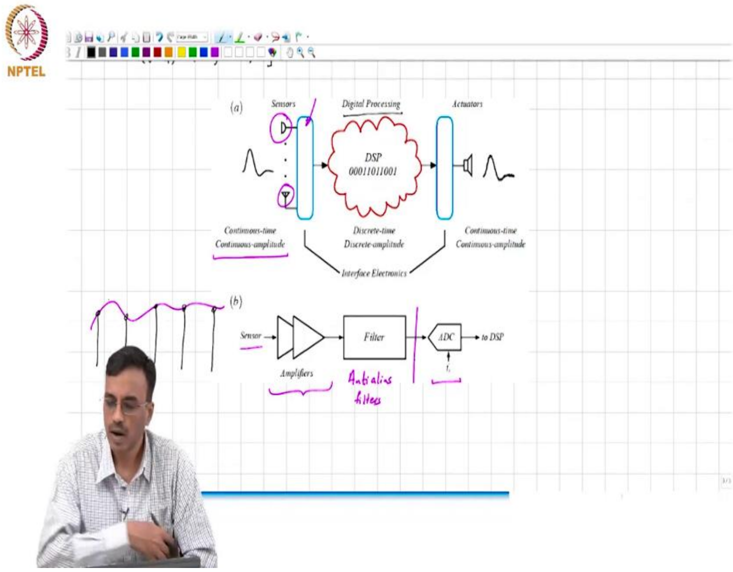

Now that apart, if you look at any electronic system today, so this is the block diagram of any modern electronic system, that most of the information is eventually stored and processed digitally but the real world signals that you were interested in namely I did not know maybe audio, maybe radio, whatever they are all in analog form.

So, these signals are all what you call signals that are continuous both in time and amplitude and the DSP only understands signals at discrete instance in time and discrete amplitude levels, so you need basically some electronics or some way to interface between the real world and the virtual world. Does it makes sense? And what enables you to do that, I mean how can you take a signal that is continuous in time and represent it losslessly as a sequence?

Sampling the shannon theorem basically is telling you that a signal is varying sufficiently slowly you did not need to keep looking at the signal all the time, you only look at it once in a while and then you are good enough and to prevent aliasing you need an anti ls filter. So, this is you have a sensor, the sensor signals are typically very small so you need to gain them up a little bit and then you need to have an anti-rays filter we already have seen what the features of this need to be, then you sample and quantize.

So, quantization of course is a process where you lose information but if you make this the quantization step sufficiently small, then it stands to reason well that the loss of information is so little that for all practical purposes you can ignore quantization. So, the a to d converter for all within code signal processing purposes is simply a is equivalent to a sampler, which continuous takes a continuous time input and gives you the discrete time.

In other words you have a continuous time input and then in the (())(17:00) book you basically, say, well this is my discrete time out. So, the a to d converter is basically taking this continuous time in and giving you the discrete time out. And if you look at what a to d conversion does, it is also what is sampling?

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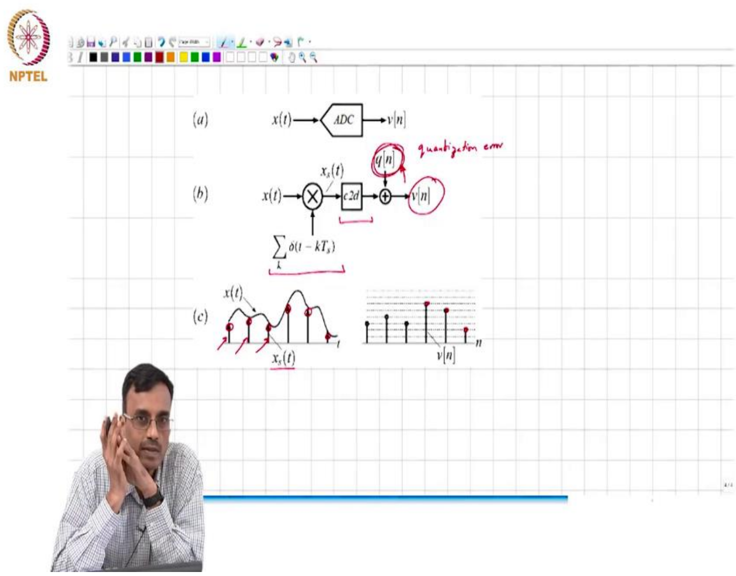

I did not know why the resolution is so poor but, so as you can see here this is what the a to d converter does but in mathematically what is happening you are sampling, so sampling in the continuous time domain is in principle taking x of t and multiplying it by an impulse train where these deltas are continuous time direct impulses. So, you now get excess of t which is something like that. Then you have some contraption in a signal processing model we do not worry about what that contraction is.

Now we convert these continuous time direct impulses into discrete time direct impulses, where the each discrete time impulse is basically corresponds to the area of that continuous time direct pulse and then once you get that you quantize it to the nearest level and therefore there is some quantization error and we get the quantize c. And as I was saying in your application you want to make sure that this q of n is sufficiently small so that for all practical purposes as far as the system is concerned you can think of it as being 0.

So, that is the job of the a to d converter designer we did not want to worry about it. So, again as you can see is this is sampling a time invariant operation or a time varying operation? It is a time varying operation because your again you can see that you are multiplying it not with the sinusoid now but with a periodic train of impulses.

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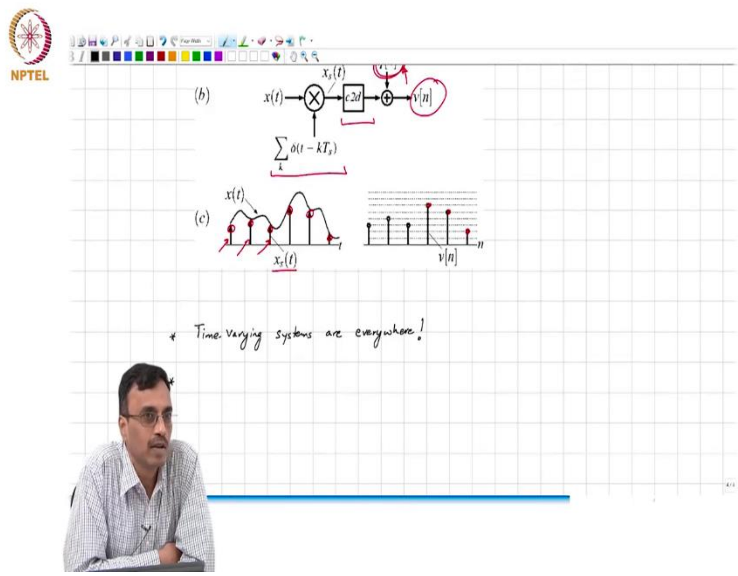

So, therefore to cut a long story short, whenever you pick up a phone to do anything you are pretty much harnessing the power of time variance and likewise whenever you build an electronic system with an A to D converter in it which basically today is all systems. Here also, I mean the key systems aspect of all these systems is time is that we are exploiting time variance to do whatever we are supposed to do. So, the bottom line therefore is that time varying systems are everywhere.

So, the question is not why should we study time varying systems, the question is how come we did not study we never bothered to study these important class of systems ever. So we spend in undergrad classes you probably spent taking I mean you have taken signals and systems, you have taken linear circuits where you basically learn Laplace and Fourier and so on. Then you have taken analog circuits and you have taken you know communication and so on.

Where every time you basically studying circuits that are and techniques that apply to systems that are linear and time invariant. Whereas the application of time variance is everywhere. So, in the lectures to come we will fill that shortcoming in your background. Does it make sense?