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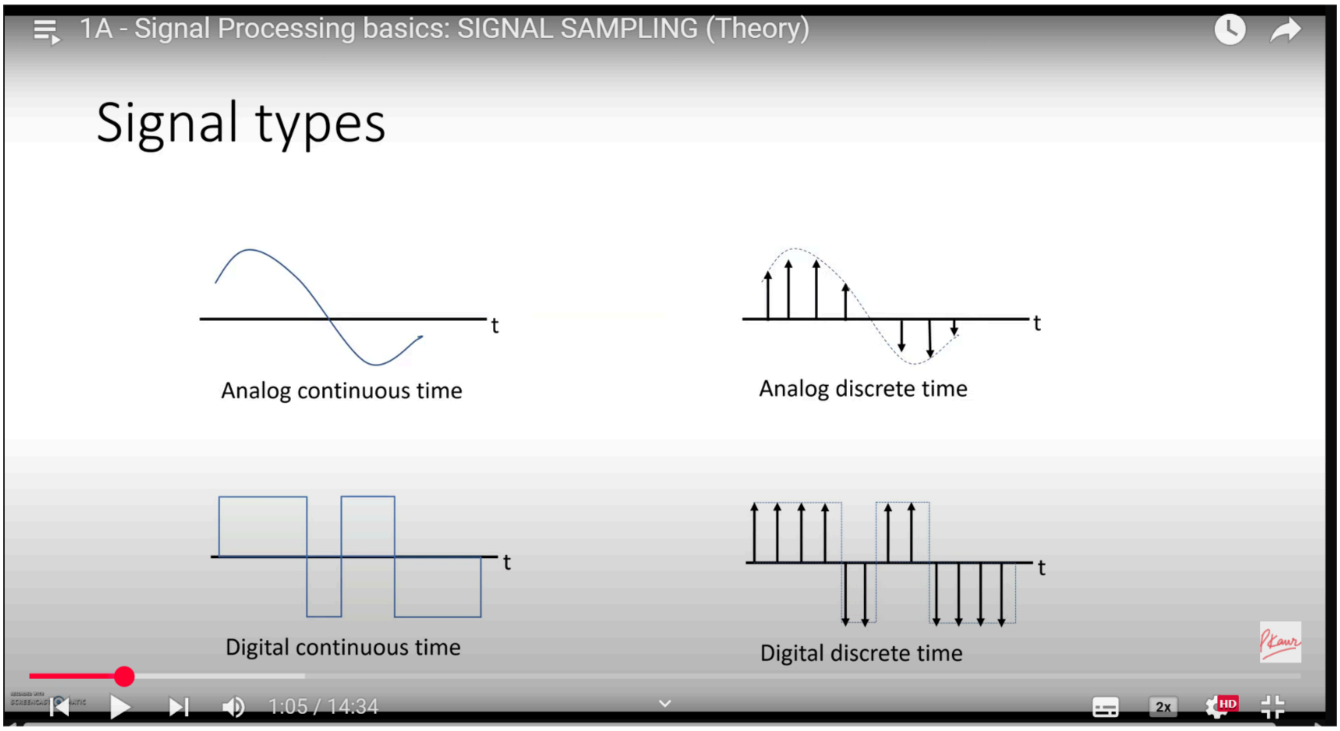
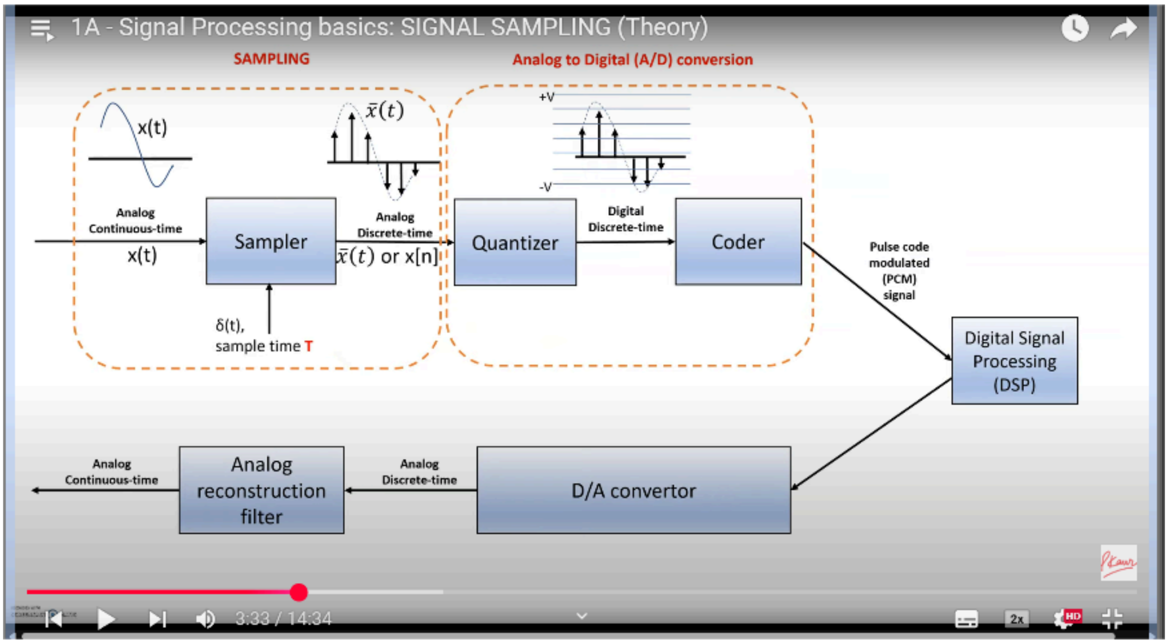
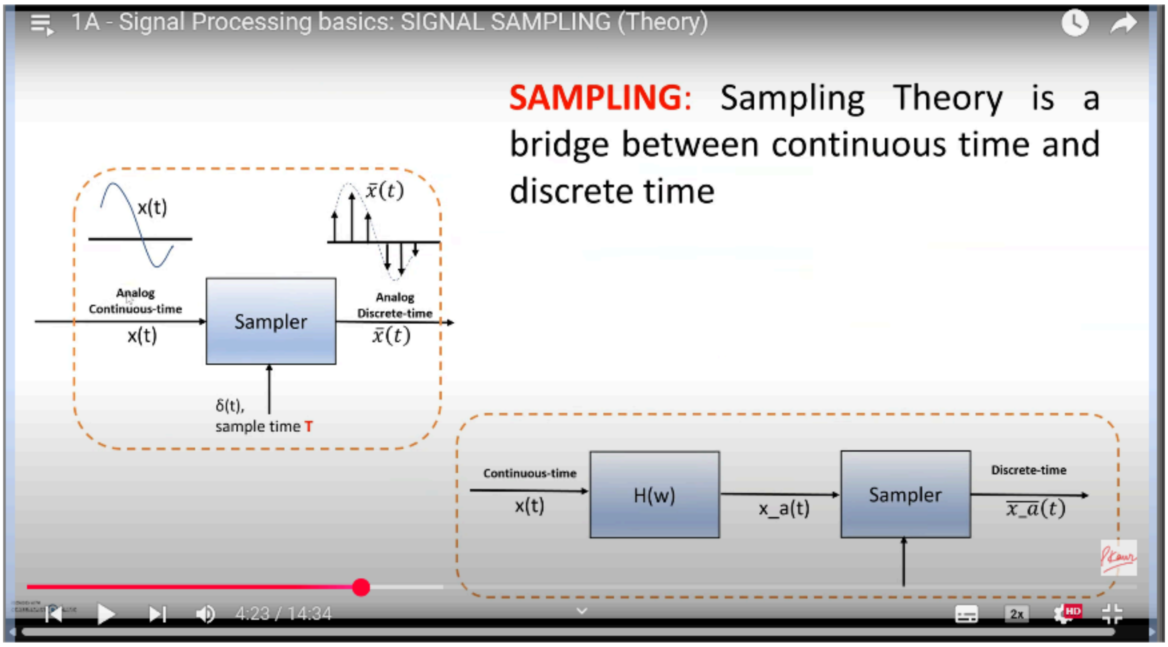
Fundamentals of Signal Processing : Types of Signals -

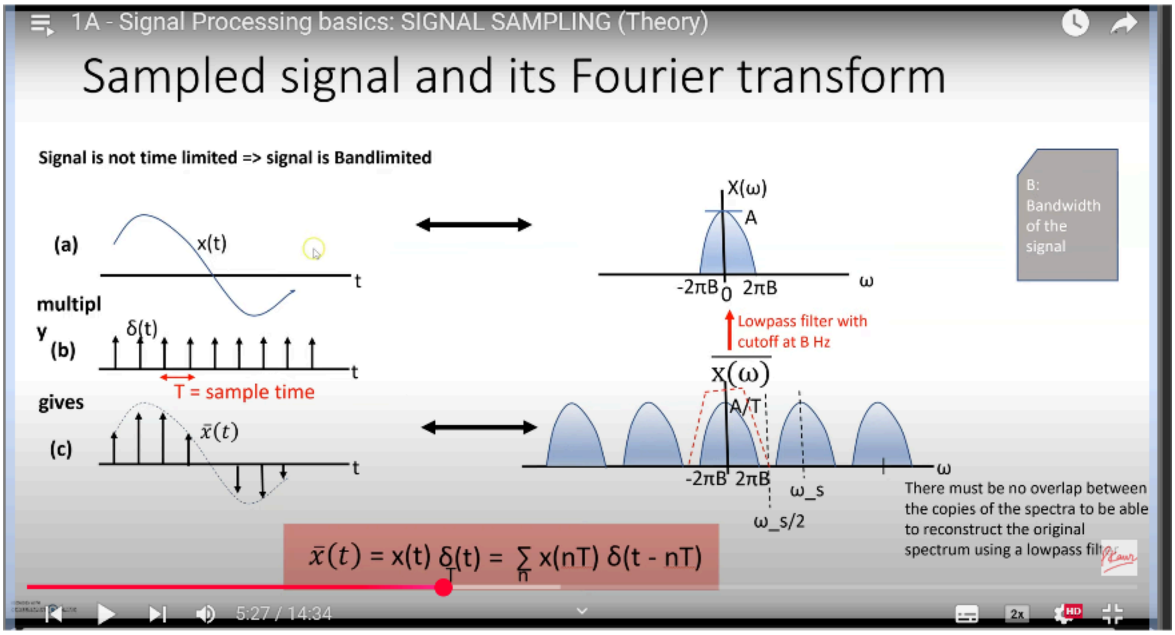


Definations Analog Signal : The amplitude or the value of signal can be any value as opposed to digital signal where the amplitutde get can only take some difined values

Continous time - 1) analog continous time 2) Digital continous time Discrete time - 1) Analog discrete time 2)digital discrete time







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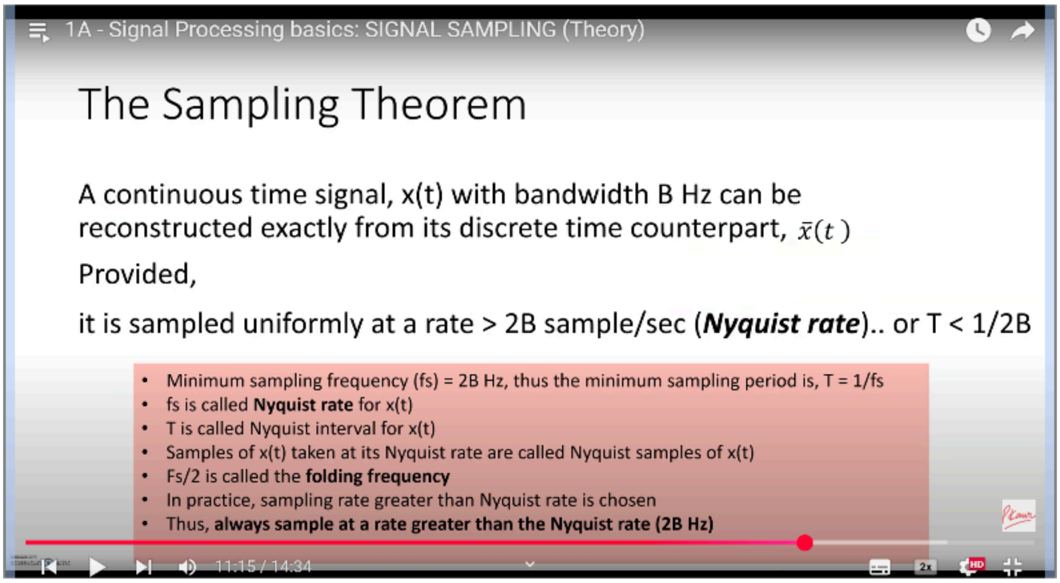
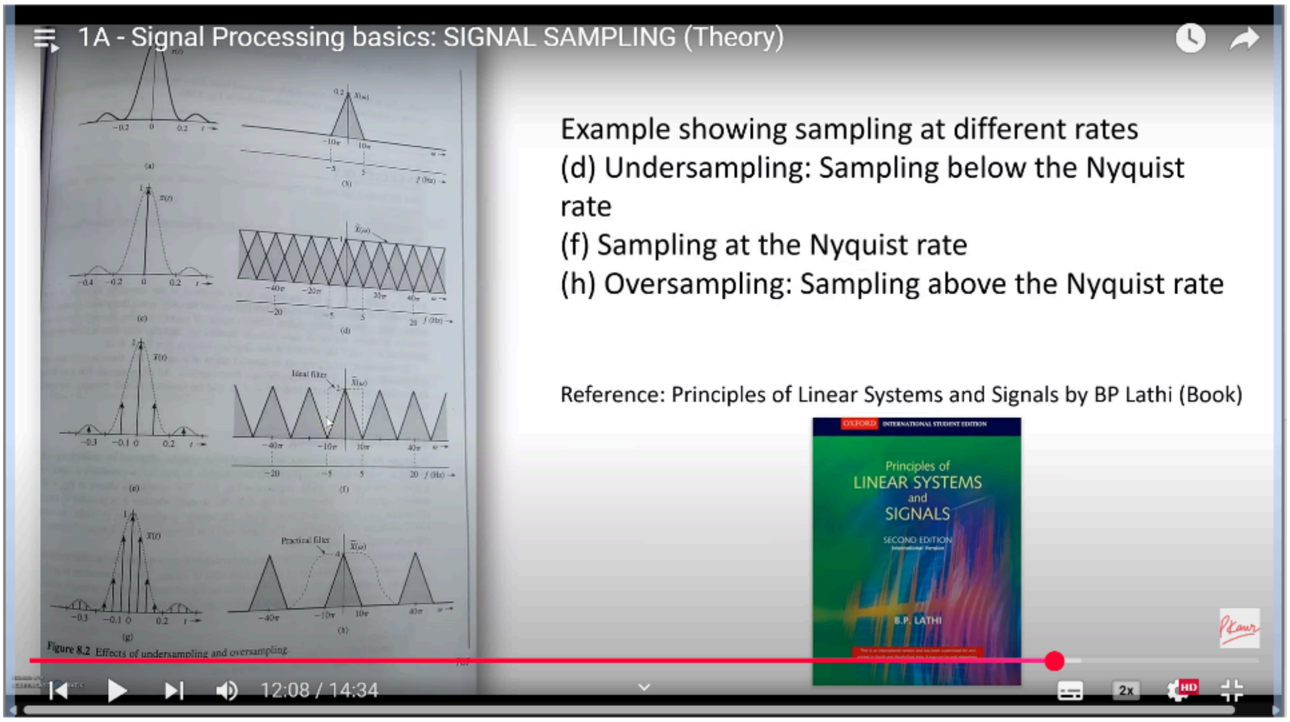
frequency of the signal and the lowest frequency of the signal so that's the bandwidth of a signal so this is our band limited spectrum and if we analyze the sampled signal in the frequency domain in other words if we take the fourier transform of the sampled signal then we get a spectrum or or the frequency representation of the signal that looks like this so uh it's basically the original spectrum and then these are just the copies of this original spectrum so this is what the sampled signal links looks

like in the in the frequency domain now in order to in order to decide what a good sample time is we want to make sure that the sampled signal can be easily used to reconstruct the original signal now why that is important i mean for it's important for many reasons but just the basic uh reason is that when we take this original signal then we sample this we have already lost some information right so we we don't want to lose all the information so in order to preserve the information we want to make sure that we sample it 09:47 not not at very wide sampling periods so the key is that we must be able to easily reconstruct the original signal from the sampled signals so the way we do that is we take this frequency transform of the sampled signal and we pass that through our low pass filter and when we pass that through the low pass filter it essentially it essentially filters out everything that is above a certain frequency in this case we have the cutoff at b hertz so we it will block out all the frequencies and only allow

this particular frequency to pass through so this way we are able to reconstruct the original spectrum from the as a spectrum of the sampled signal and then it's easy i mean this is just the in inwards for your transform going from here to the time domain so the key here is that we want to make sure when we sample that the sample signal can be easily used to reconstruct the original signal so that's the key in order to answer that now what should be the sample time we have this theorem called sampling



theorem so the sampling theorem basically states that a continuous time signal x of t with a bandwidth b can be reconstructed exactly from its discrete time counterpart provided it is sampled uniformly at a rate greater than 2 b samples per second and we call this a nyquist rate in other words the time period the sampling period should be less than 1 over 2b where again b is the bandwidth of the original signal so now this is our answer that the sample time should be less than or equal to 1 over 2b where 2b is the nyquist rate so that's the key here in this slide it's just showing the it's just showing what happens when we sample a different time frame so here we will start with f so this is the this is the frequency this is the signal that is sampled at nyquist frequency so it's sampled at t equals 1 over 2b and here we see these spectrums are next to each other of the sample signal and the first one you are sampling below the nyquist frequency so here you have an overlap of spectrum and here you are sampling above the



nyquist frequency so here the spectra are spaced well apart so if we need to reconstruct the original signal then we just pass this into the low pass filter and we get the original signal as opposed to this one where we are sampling below the nyquist frequency it's just not possible to reconstruct the original signal because we wouldn't know where to put the low pass filter so just just an example it is from the book called it's really a good book it's called linear signal linear systems and signals by bp latti so that is how we sample as i said you might have seen before this in one of the other slides so prep in practice you have to pass the signal through anti-aliasing filter before we sample so that's that this completes the pipeline to go from continuous time to discrete time through sampling and the reason why we have to do that is because as we saw in the previous slide here i said that signal is not time limited but in reality in real life most of the signals are time limited so they are not band limited so what 14:07 anti-aliasing filter does is it first band limits the signal and then it passes through the sampler and then we are able to get what we want so that is it for this video in the next view i'm going to do an example where we're going to convert the analog signal to our digital signal by hand so tune in for the next video thank you bye