# Sinyal Sistem Week 2b – 3a: Periodic Sampling

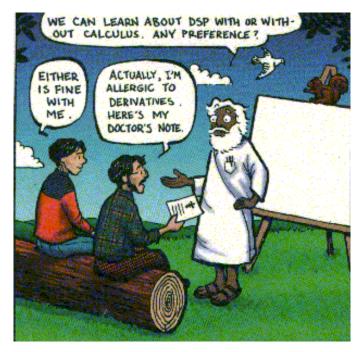
@btatmaja
Compiled fror

Compiled from :-David Marshall: Scientific Computing...

-Mark D. Shattuck, Image Analysis, HOS 2015

# Periodic Sampling

- From time scaling to resampling
- Sampling Theorem and Aliasing
- Konvolusi
- Korelasi



### Sample Rates and Bit Size

#### Bit Size — Quantisation

How do we store each sample value (Quantisation)?

```
8 Bit Value (0-255)
```

16 Bit Value (Integer) (0-65535)

#### Sample Rate

How many Samples to take?

11.025 KHz — Speech (Telephone 8 KHz)

22.05 KHz — Low Grade Audio (WWW Audio, AM Radio)

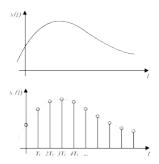
44.1 KHz — CD Quality



### Digital Sampling (1)

#### Sampling process basically involves:

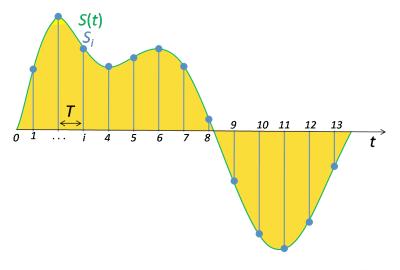
- Measuring the analog signal at regular discrete intervals
- Recording the value at these points







### Digital Sampling (2)







#### Nyquist's Sampling Theorem



The Sampling Frequency is critical to the accurate reproduction of a digital version of an analog waveform

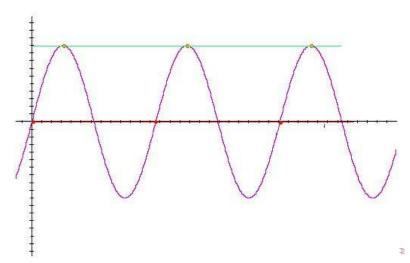
#### Nyquist's Sampling Theorem

The Sampling frequency for a signal must be at least twice the highest frequency component in the signal.



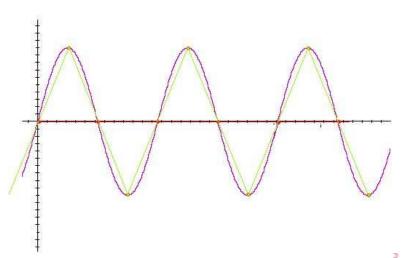


### Sampling at Signal Frequency



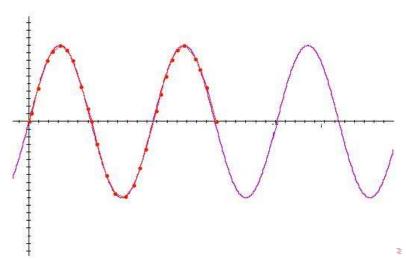


### Sampling at Twice Nyquist Frequency



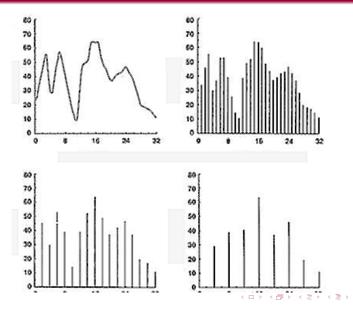


### Sampling at above Nyquist Frequency



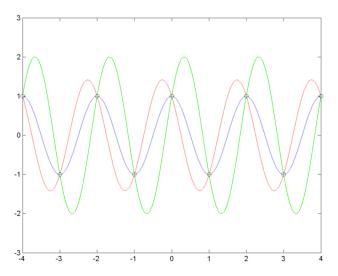


### If you get Nyquist Sampling Wrong? (1)





### If you get Nyquist Sampling Wrong? (2)







### Implications of Sample Rate and Bit Size (1)

#### Affects Quality of Audio

- Ears do not respond to sound in a linear fashion
- Decibel (dB) a logarithmic measurement of sound
- 16-Bit has a signal-to-noise ratio of 98 dB virtually inaudible
- 8-bit has a signal-to-noise ratio of 50 dB
- Therefore, 8-bit is roughly 8 times as noisy
  - 6 dB increment is twice as loud

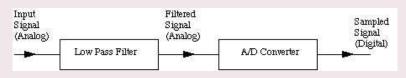




### Practical Implications of Nyquist Sampling Theory

### Filtering of Signal

• Must (low pass) filter signal before sampling:



 Otherwise strange artifacts from high frequency (above Nyquist Limit)signals would appear in the sampled signal.





### Why are CD Sample Rates 44.1 KHz?

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Upper range of human hearing is around 20-22 KHz — Apply Nyquist Theorem





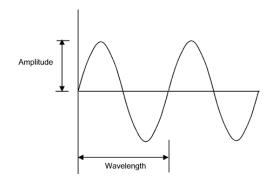
### Basic Digital Audio Signal Processing

In this section we look at some basic aspects of **Digital Audio Signal Processing**:

- Some basic definitions and principles
- Filtering
- Basic Digital Audio Effects



### Simple Waveforms



- Frequency is the number of cycles per second and is measured in Hertz (Hz)
- Wavelength is inversely proportional to frequency
   i.e. Wavelength varies as 
   <sup>1</sup>/<sub>frequency</sub>





Convolution and Cross Correlation

$$C(f,g)_n = \sum_{k=0}^{K-1} f_k g_{(n-k)}$$

$$X(f,g)_n = \sum_{k=0}^{K-1} f_k^* g_{(n+k)}$$

1 3 2 1 3 1 0 2

1 3 2 1 3 2 0 1

1 3 2 1 3

102

1 3 2 1 3

201

1 3 2 1 3

102

1 3 2 1 3 1 3 2 1 3 2 0 1 1 3 4 7 7 2 6 full

same

valid

Convolution and Cross Correlation

$$C(f,g)_n = \sum_{k=0}^{K-1} f_k g_{(n-k)}$$

$$X(f,g)_n = \sum_{k=0}^{K-1} f_k^* g_{(n+k)}$$

1 3 2 1 3

1 3 2 1 3

1 3 2 1 3

1 3 2 1 3

valid

102

13213 13213 1 0 2 2655813 full same

Convolution and Cross Correlation

$$C(f,g)_{nm} = \sum_{k=0}^{K-1} \sum_{l=0}^{L-1} f_{kl} g_{(n-k)(m-l)}$$

$$X(f,g)_{nm} = \sum_{k=0}^{K-1} \sum_{l=0}^{L-1} f_{kl}^* g_{(n+k)(m+l)}$$

