INTRODUCTION TO DIGITAL AUDIO SIGNALS

EE6641 Analysis and Synthesis of Audio Signals

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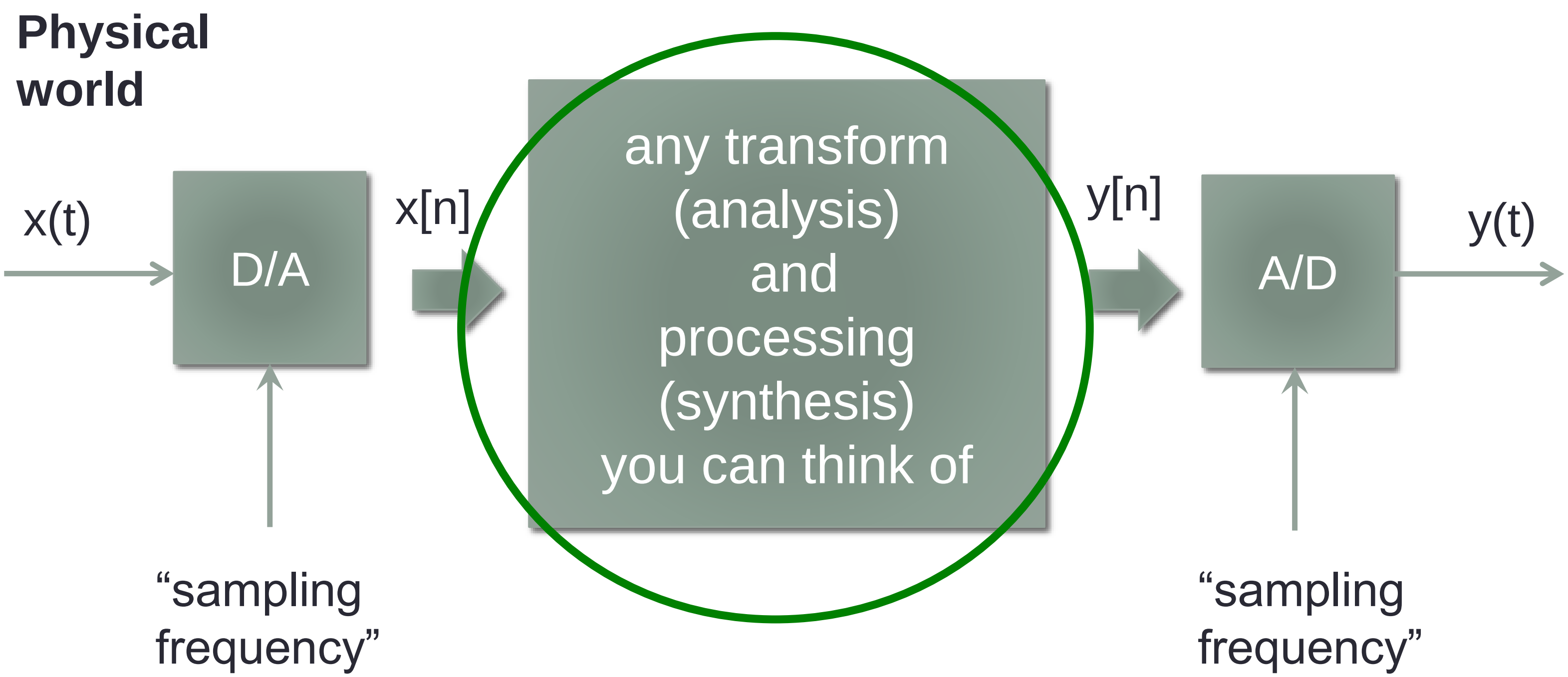
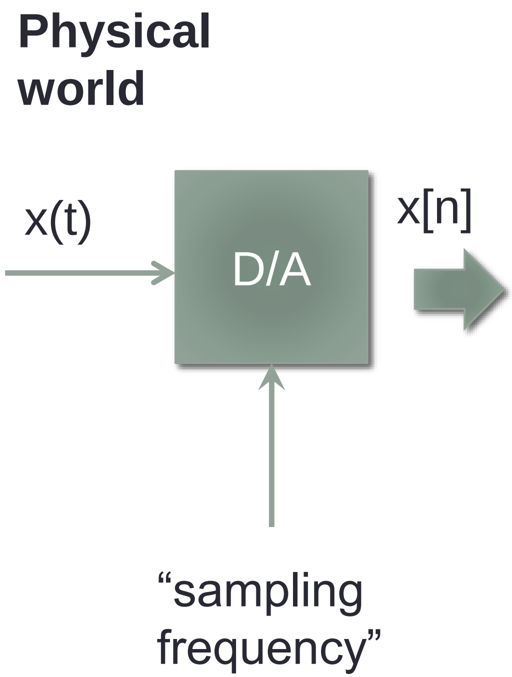
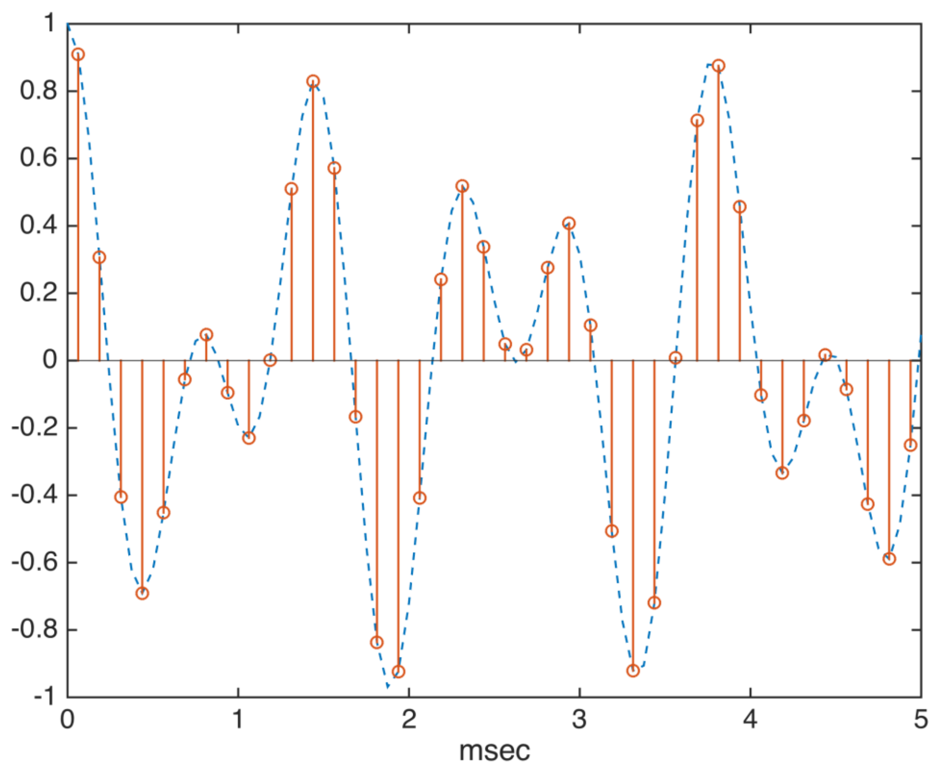
What is digital audio about?

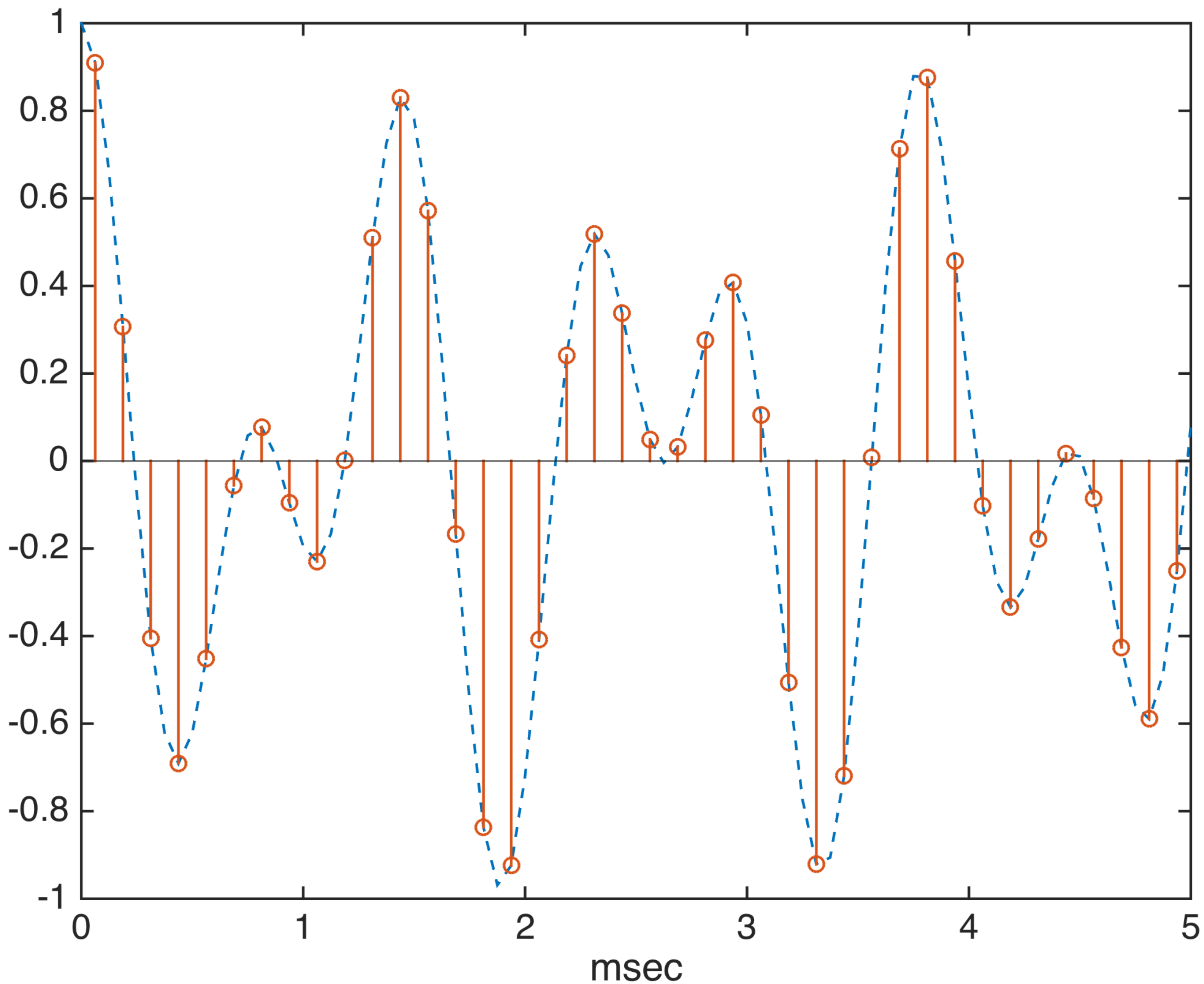


Sampling:
To look at the waveform at discrete-time

- Taken care by your soundcard with the "input"
- Need to specify how often you want to sample
- The unit of sampling rate = Hz.

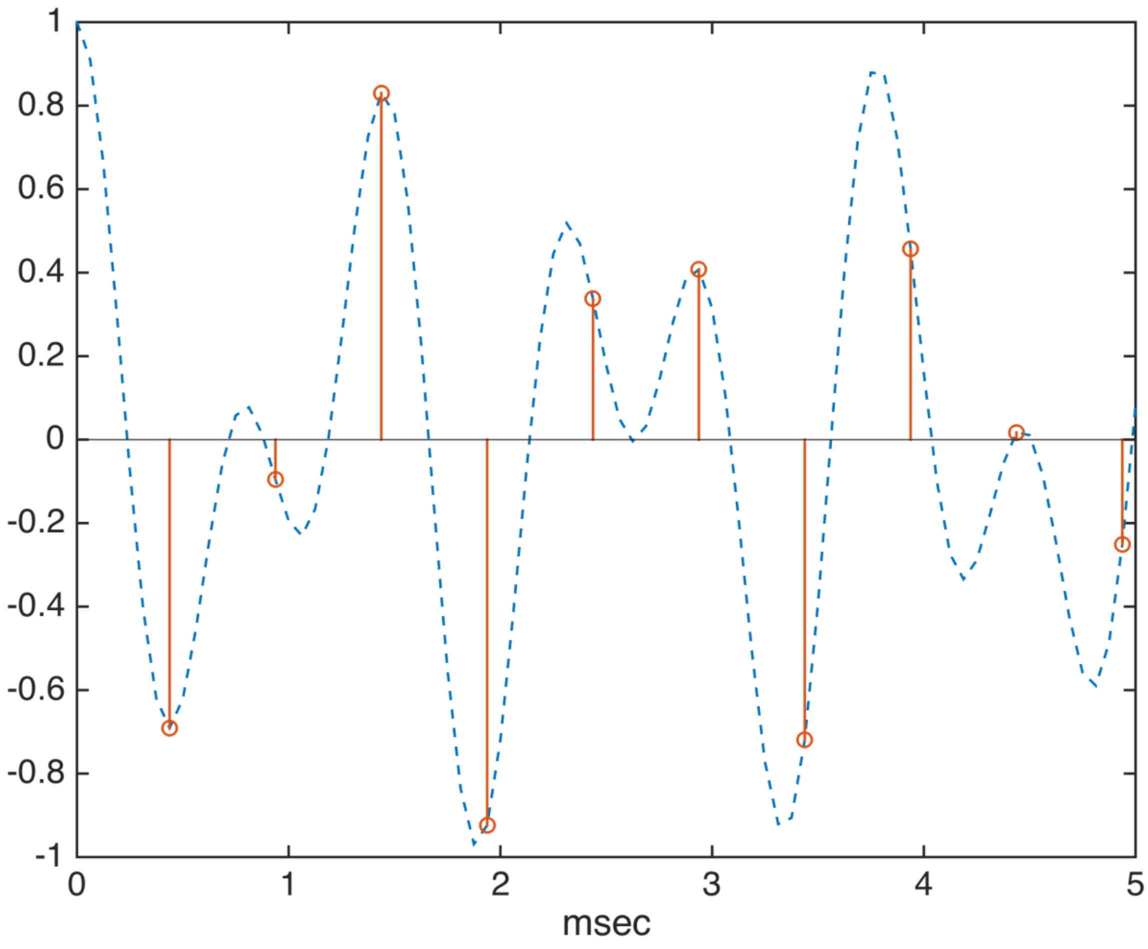






1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
0
1
2
3
4
5
msec

Undersampling

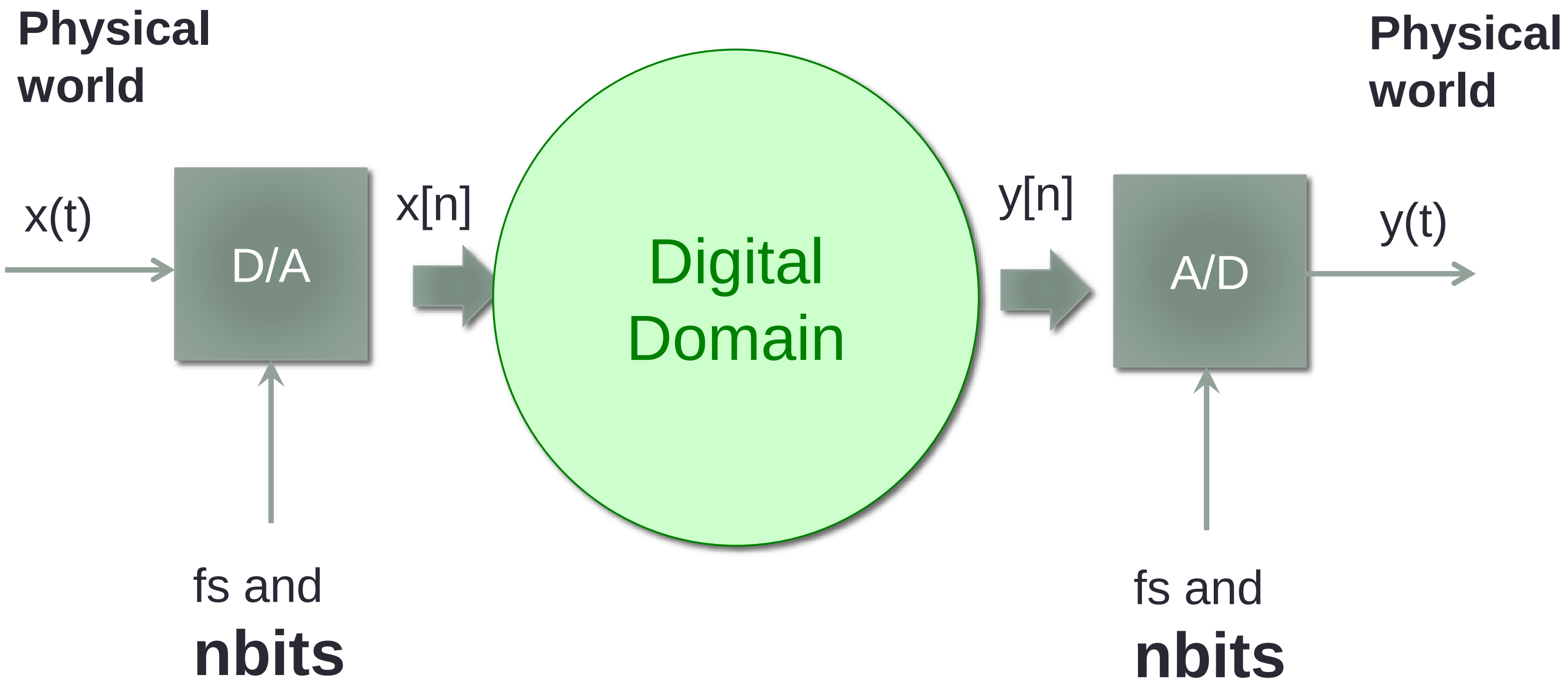


Nyquist theorem

Need to sample at least at 2x the rate of the highest frequency of interest.

E.g., f_s = 44.1 kHz for *CD quality*

Quantization: fixed-point representation (usually)



Nbits = 8, 16, 24, 32

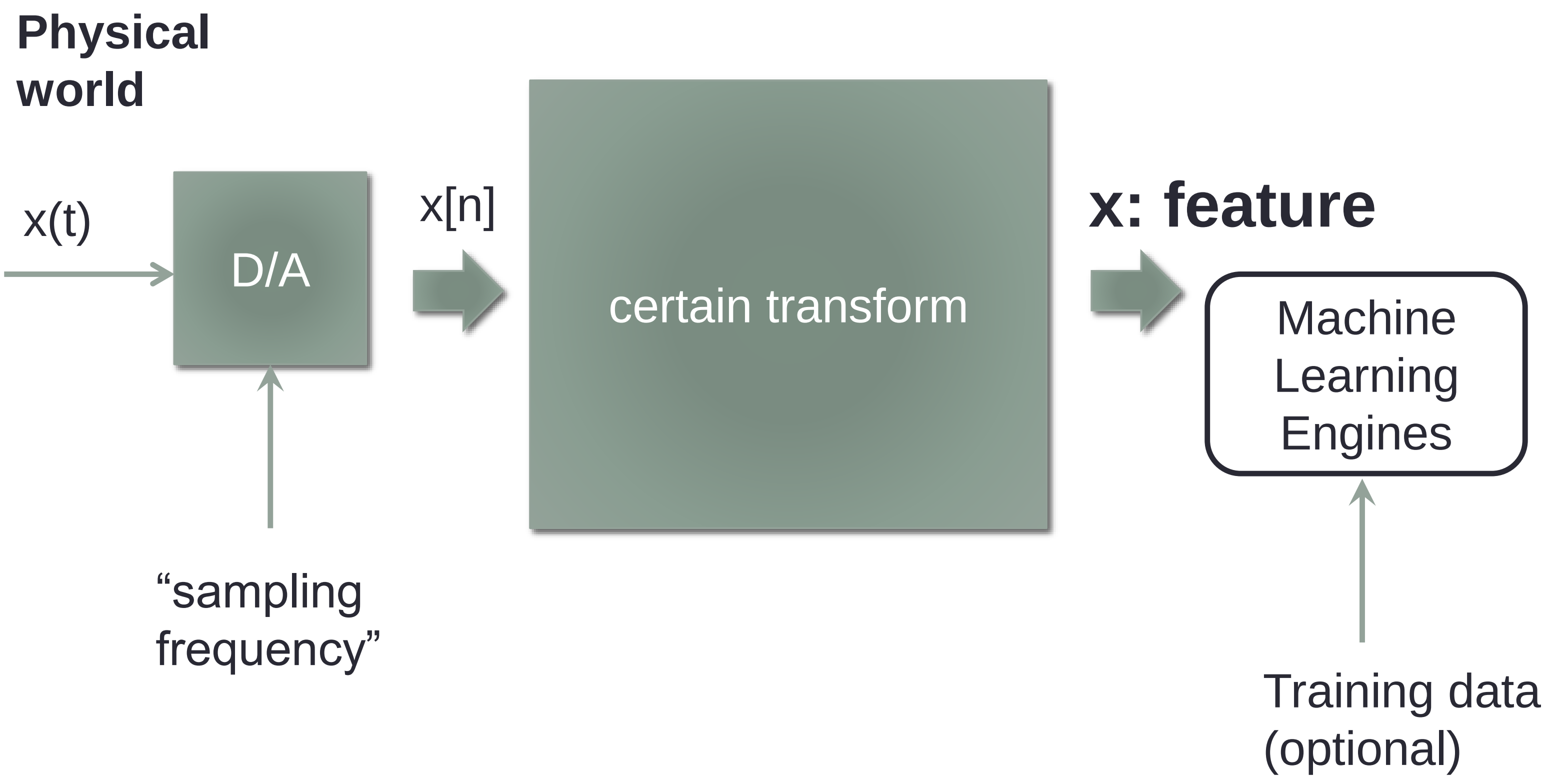
Quantization step Δ

Below are directly printed from MATLAB's document for wavwrite()

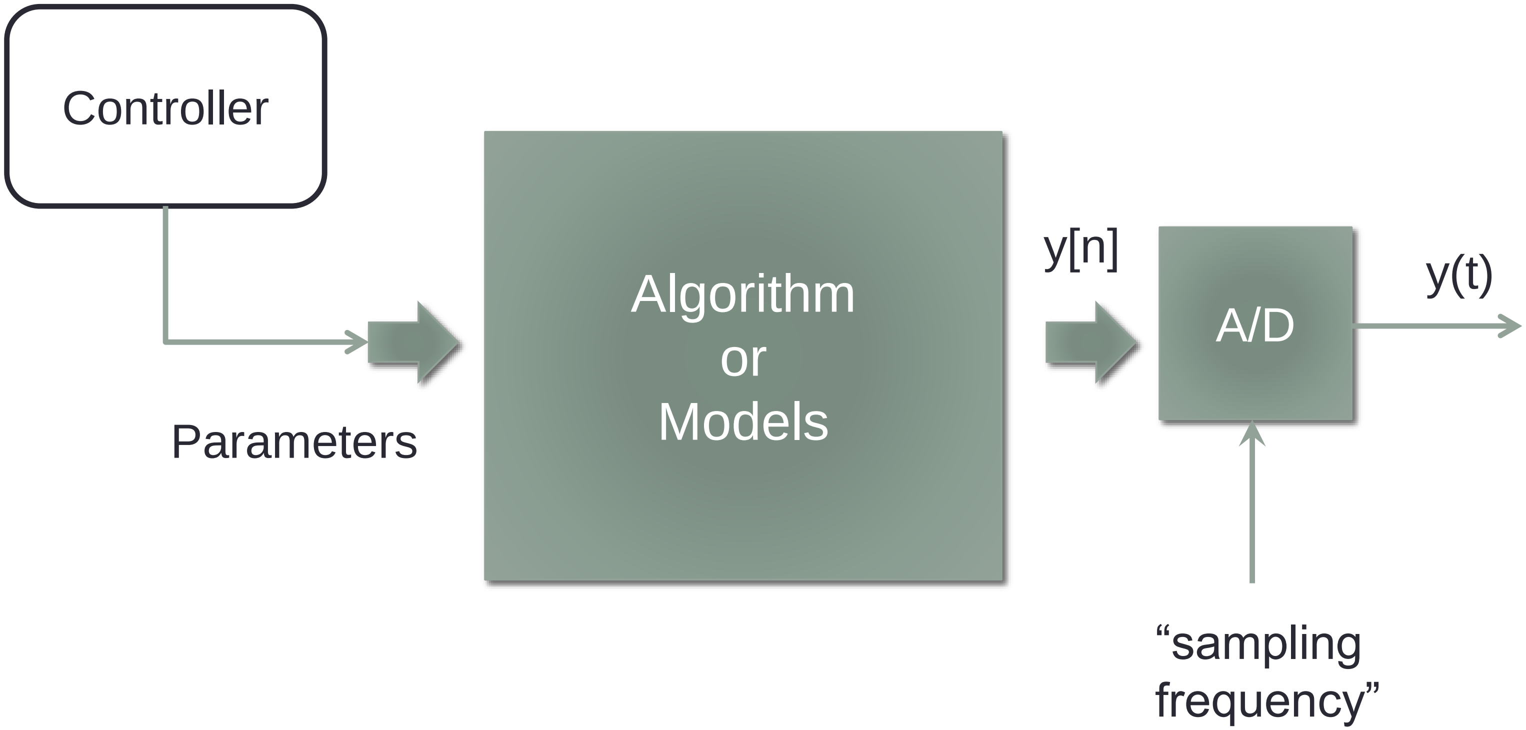
NBits	y Data Type	Data Range	Output Format
8	single or double	–1.0 <= y < +1.0	uint8
16	single or double	–1.0 <= y < +1.0	int16
24	single or double	–1.0 <= y < +1.0	int32
32	single or double	–1.0 <= y <= +1.0	single

重點: $\Delta = 2 / 2^{Nbits}$

Analysis without synthesis: *Machine listening*



Digital Sound Synthesis



Demo of sound synthesis with a very specialized controller

Related courses

- 李祈均教授：語音訊號處理
- 白明憲教授：聲學、陣列訊號處理
- 黃志方教授（交）：Max/MSP
- 冀泰石教授（交）：聽覺資訊處理