Signal Processing
(EE2S31)

Richard Heusdens, Richard Hendriks

About the Course - People

Instructors:

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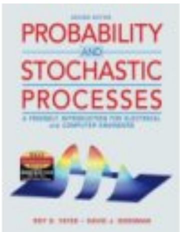
About the Course - Literature

The digital signal processing part is based on

J.G. Proakis and D.G. Manolakis. Digital Signal Processing, Principles, Algorithms and Applications. Fourth edition. Pearson Prentice Hall, 2007.

The stochastic processes part is based on



R.D. Yates and D.J. Goodman. Probability and Stochastic Processes, A Friendly Introduction for Electrical and Computer Engineers. Second edition. John Wiley & Sons, Inc., 2005.

About the Course - Structure

The course consists of plenary lectures only; no instructions (werkcolleges) are given.

The plenary lectures will be given (mostly) on Monday afternoon, 15:45-17.30, Tuesday afternoon, 13:45-15:30, and Thursday afternoon, 15:45-17:30, in EWI-lecture hall Boole/Ampere (check the schedule!)

- Goal: give easier access to reading material and highlight important points
- Presence is highly recommended!

About the Course - Topics

For the digital signal processing part, the following topics will be discussed:

- Introduction, recap LTI systems, Fourier transforms, Z-transform, poles/zeros/ frequency response, filter functions
- Non-ideal sampling and reconstruction, spectral analysis, STFT
- Frequency domain sampling, DFT, FFT, circular convolution
- Quantization and round-off effects, sigma-delta modulation
- Multi-rate signal processing
- All-pass filter structures

About the Course - Topics

Date			Topic
18/4/2016	EE2s31		RH: SP intro DSP
21/4/2016	EE2s31		RH: SP recap 1
25/4/2016	EE2s31		RH: SP recap 2
26/4/2016	EE2s31		RCH: Stoch lec 1
28/4/2016	EE2s31		RH: SP Sampling
2/5/2016	EE2s31		RH: SP FFT
3/5/2016	EE2s31		RH: SP Quantization
9/5/2016	EE2s31		RCH: Stoch lec 2
10/5/2016	EE2s31		RCH: Stoch lec 3
12/5/2016	EE2s31		RCH: Stoch lec 4
18/5/2016	EE2s31	mid-term exam	
26/5/2016	EE2s31		RCH: Stoch Lec 5
30/5/2016	EE2s31		RCH: Stoch lec 6
31/5/2016	EE2s31		RH: SP Multi rate
2/6/2016	EE2s31		RCH: Stoch lec 7
6/6/2016	EE2s31		AvdV: SP Filterontwerp
7/6/2016	EE2s31		RCH: Stoch lec 8
9/6/2016	EE2s31		RCH: Stoch lec 9 reserve
13/6/2016	EE2s31		Oefen tentamens
24/6/2015	EE2s31	exam	

About the Course - Exam

- Written exam in two parts:
- First part: 13:30-15:30, May 18, 2016
- Second part: 13:30-16:30, June 24, 2016
- Final mark is the average of the two exams
- First part not used in re-examination!
- You are allowed to bring two A4 formula sheets
- Old exams available on Blackboard

The Concept of Signals and Systems

Signal:

- A signal can be broadly defined as any quantity that varies as a function of time and/or space and has the ability to convey information
- Signals are ubiquitous in science and engineering:
  - Electrical signals: currents and voltages in electronics, audio, speech and video signals.
  - Mechanical signals: sound or pressure waves.
  - Biomedical signals: electro-cardiogram, X-ray images, MRI images.

The Concept of Signals and Systems

- Any series of measurements of a physical quantity can be considered a signal (temperature measurements, water levels, ...)

Signal characterization:

- The most convenient mathematical representation of a signal is via the concept of a function, say $f(x)$.
- Dependent on the nature of f and x, different types of signals can be defined:
  - **analog signal:** $x \in \mathbb{R}^k, \; f : x \mapsto f(x) \in \mathbb{R}^m$ (or $\mathbb{C}^m$)

    When x denotes time, we also refer to such a signal as a *continuous-time signal* (usually denoted by $f(t)$)

The Concept of Signals and Systems

- **discrete signal:** $x \in \mathbb{Z}^k, \ f : x \mapsto f(x) \in \mathbb{R}^m$ (or $\mathbb{C}^m$)

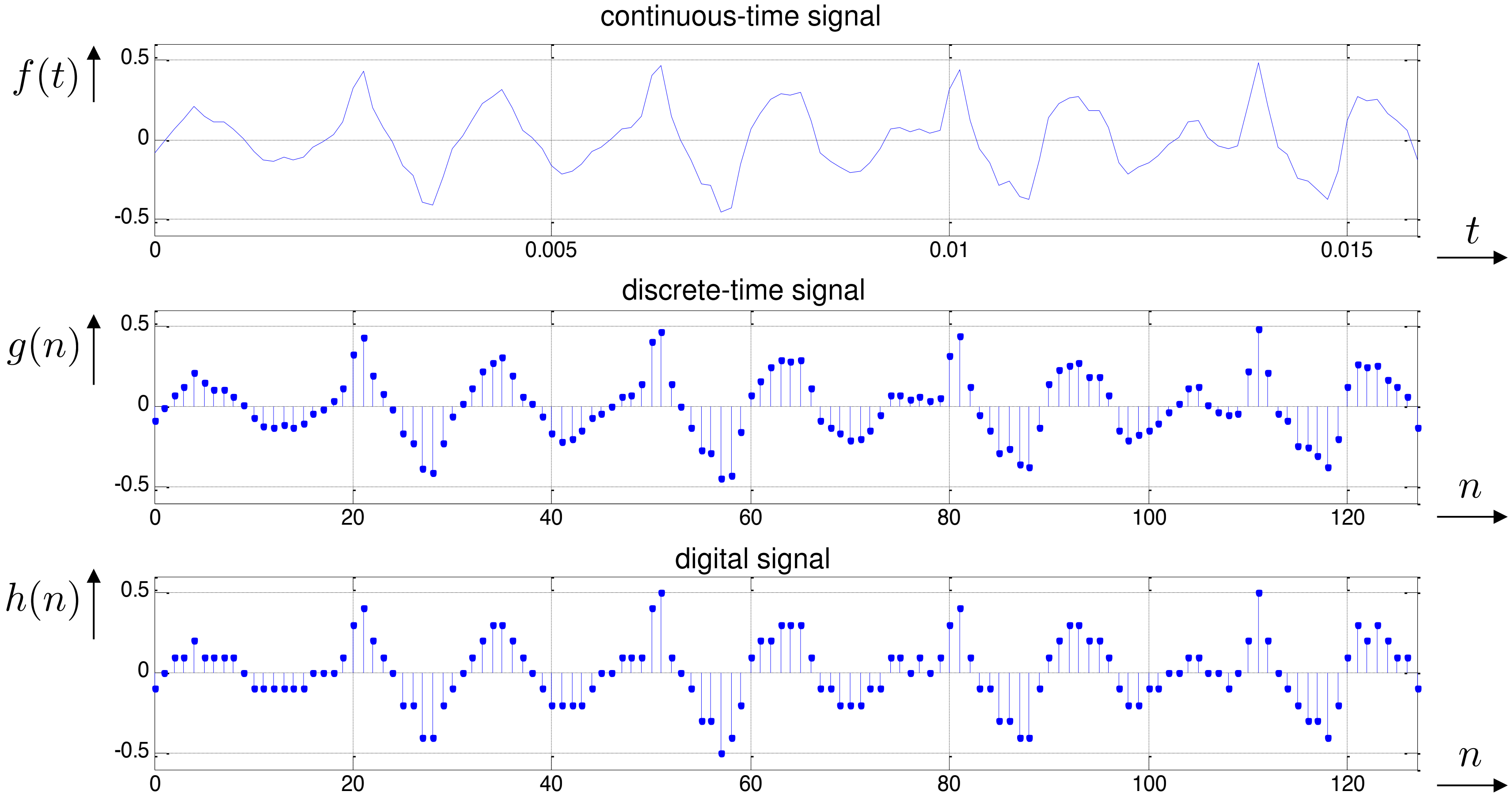
  When x represents sequential values of time, we also refer to such a signal as a *discrete-time signal* (usually denoted by $f(n)$)

- **digital signal:** $x \in \mathbb{Z}^k, \ f : x \mapsto f(x) \in A \subset \mathbb{R}^m$ (or $\mathbb{C}^m$), where $A = \{A_1, A_2, \ldots, A_N\}$ represents a finite set of signal levels

- Distinctions can also be made at the model level, for example, whether f is considered to be deterministic or random in nature

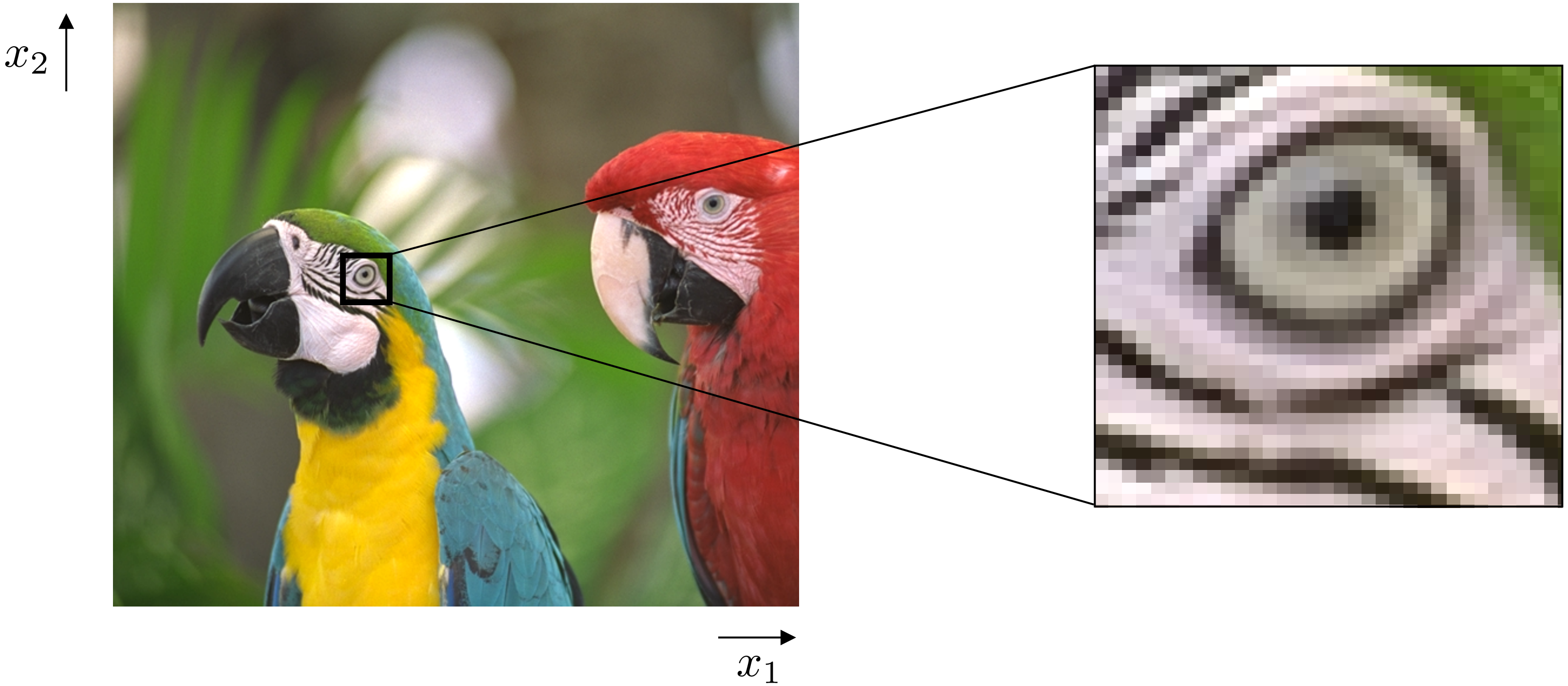
The Concept of Signals and Systems

Example: Speech signal



The Concept of Signals and Systems

Example: Digital image: $x \in \mathbb{Z}^2,\ f : x \mapsto f(x) \in \mathbb{Z}^3_{[0,255]}$ (R,G,B)

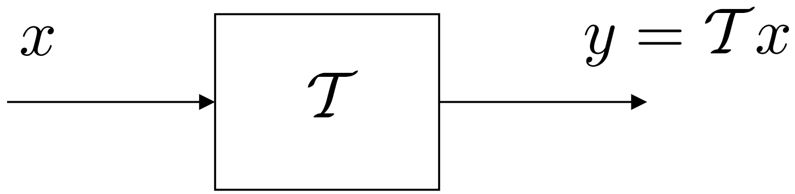


The Concept of Signals and Systems

System characterization:

- A system can be respresented mathematically as a mapping $\mathcal{T}: S_1 \longrightarrow S_2$:

$$x \in S_1 \mapsto y = \mathcal{T}x \in S_2$$



The Concept of Signals and Systems

- Depending on the nature of the signals on which the system operates, different basic typs of systems may be indentified:
  - analog system: the input and output signals are both analog in nature
  - discrete system: the input and output signals are both discrete
  - digital system: the inputs and outputs are digital
  - mixed system: a system in which different types of signals coexist

Digital Signal Processing

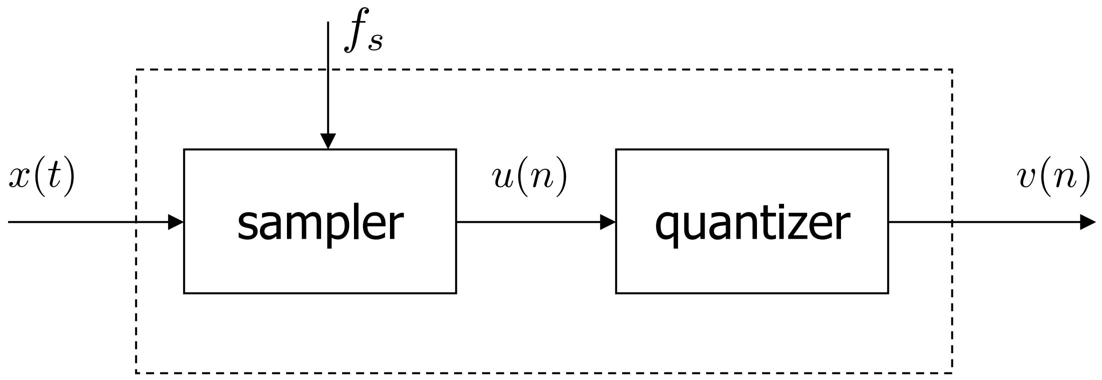
In its most general form, *digital signal processing* (DSP) refers to processing of analog signals by means of discrete-time (discrete-space) operations implemented on digital hardware.

From a system point of view, DSP is concerned with mixed systems:

- the input and output signals are analog
- the processing is done on the equivalent digital signals

Analog-to-Digital Converter

Two-step approach:



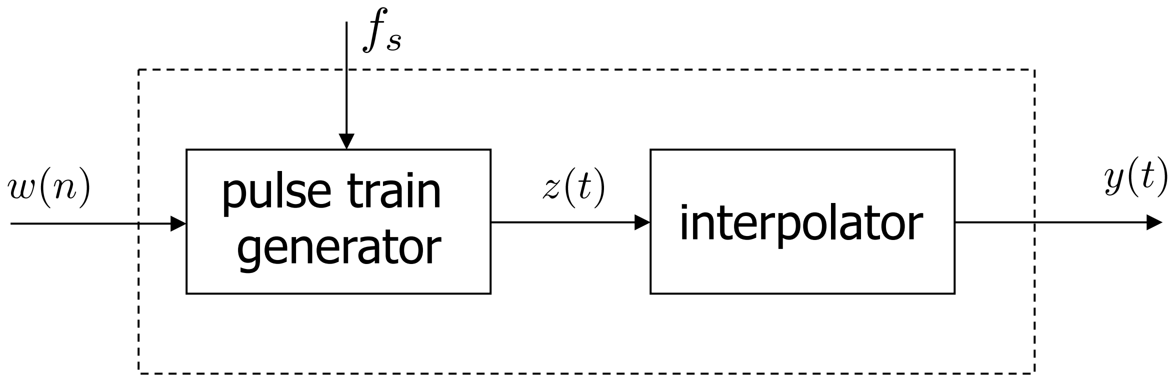
- Sampler: $u(n) = x(nT_s)$ where $T_s = 1/f_s$, the sampling period
- Quantizer: $v(n) = (Qu)(n)$, where Q is a (nonlinear) mapping from intervals of the real line (quantization cells) to reproduction levels

Analog-to-Digital Converter

- The number of representation levels is hardware defined, typically 2^B, where B is the number of bits in a word
- The sampling frequency depends on the frequency content of the analog signal (sampling theorem)
- In many systems the discrete levels are uniformly spaced. This is the case for audio CDs, where levels are represented by 16 bits words. The sampling frequency in this example is 44.1 kHz.
- In digital telephony, for example, the quantization is nonuniform and telephone calls are digitized to 8 bit resolution at a sampling rate of 8 kHz.

Digital-to-Analog Converter

Two-step approach:



- The pulse-train generator transforms the sequence of numbers $w(n)$ into a sequence of scaled, analog pulses (spaced $T_s = 1/f_s$ seconds apart)
- The interpolator removes high-frequency components in z (via low-pass filtering) to produce a smooth analog output signal

Digital System

- The digital system is functionally similar to a microprocessor: it has the ability to perform mathematical operations on a discrete-time basis and can store intermediate results in internal memory
- The operations performed by digital systems can usually be described by means of an algorithm on which the implementation is based
- In real-time systems the computing associated to each sampling interval can be accomplished in a time interval $< T_s$. Off-line (non real-time) systems operate on stored digital signals.

Pros and Cons of DSP

Advantages:

- Robustness
  - signal levels can be regenerated
  - results are reproducible
- Flexibility
  - easy control of system accuracy (sampling rate, word length)
  - software programmable
- Storage capability
  - DSP can be interfaced to low-cost storage devices
  - allows for off-line computations

Pros and Cons of DSP

Disdvantages:

- Cost/complexity added by the A/D and D/A conversion
- Input signal bandwidth is technology limited
- Quantization effects: discretization of the levels introduces quantization noise to the signal
- Simple conversion of a continuous-time signal to a binary stream of data involves an increase in bandwidth required for data transmission (can be mitigated by data compression techniques, like mp3-coding for audio signals)

Digital Signal Processing

- It is not at all clear that an arbirtary analog system can be realized using a DSP system
- Fortunately, for the important class of linear time-invariant systems, this equivalence can be proved under the following conditions:
  - the number of representation levels provided by the digital system is sufficiently large that quantization errors may be neglected
  - the sampling rate is higher than twice the highest frequency contained in the analog input signal (sampling theorem)

Discrete-Time Signal Processing

Based on these considerations, it is convenient to break down the study of DSP into two distinct sets of issues

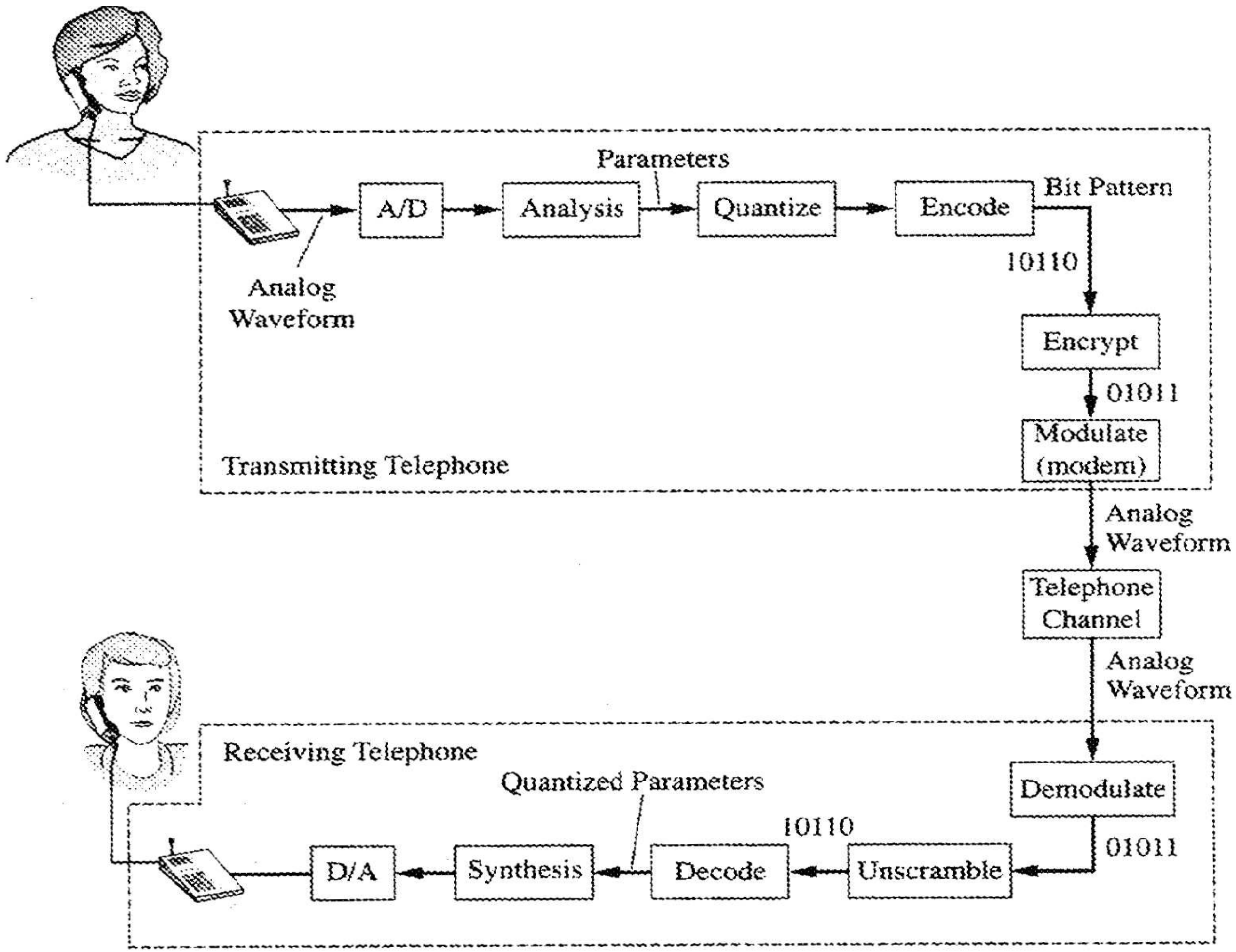
- discrete-time signal processing
- study of quantization effects

- The main object of discrete-time signal processing is the study of DSP under the assumption that finite-precision effects may be neglected
- Quantization is concerned with (important!) practical issues resulting from the use of finite-precision digital hardware

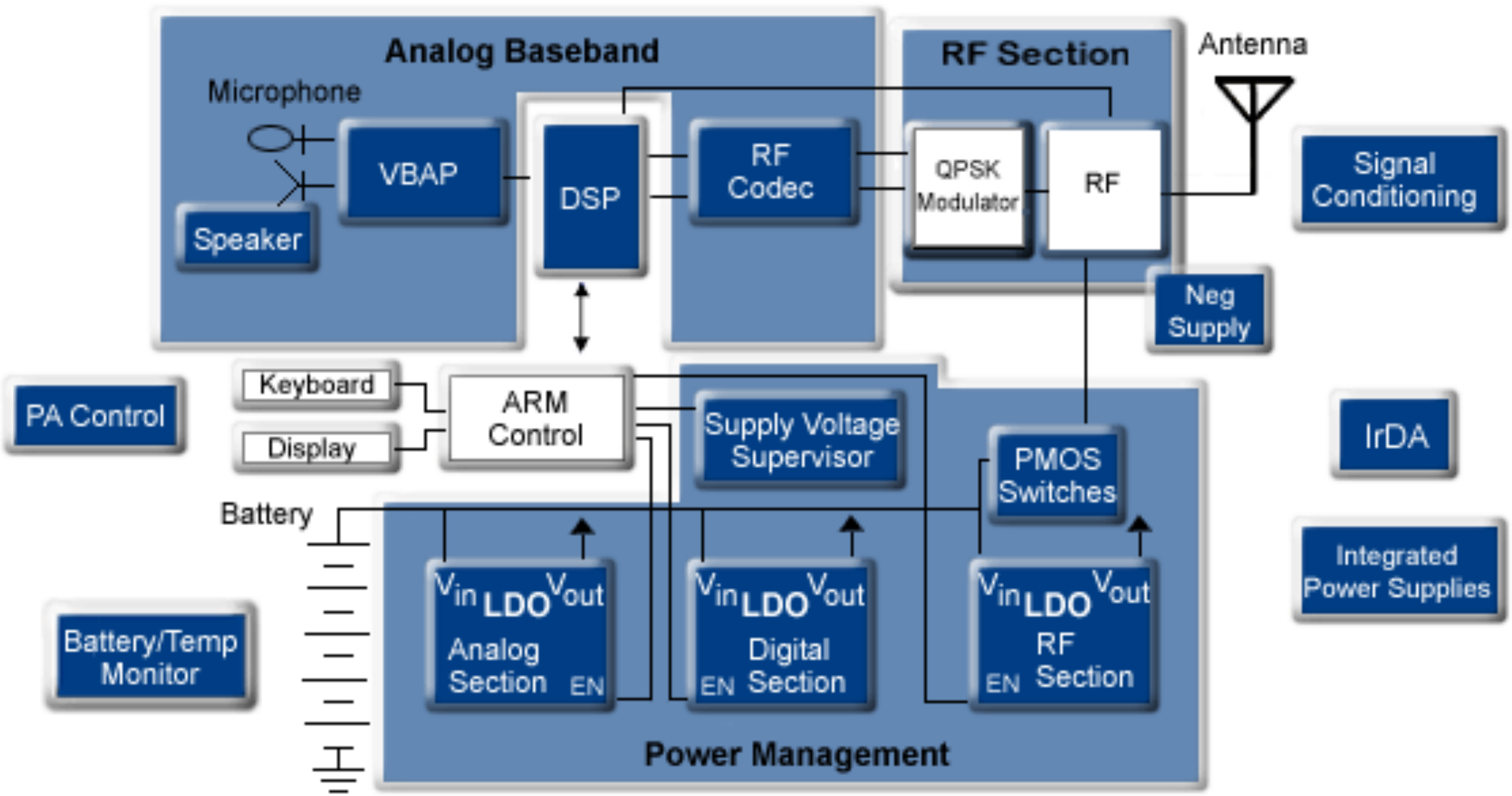
Applications of DSP

- Cell phones (speech coding, echo cancelation, ...)
- DVD players and other home audio equipment (filtering, decoding, ...)
- Cars (the anti-lock braking system, engine control chips, ...)
- Computer disk drives (management of hard disk partitions, ...)
- Satellites (remote sensing, GPS, ...)
- Digital radios (digital modulation (OFDM), wireless communication, ...)
- Hearing aids (speech enhancement, directional microphones, ...)
- Booking services (speech recognition, ...)
- ...

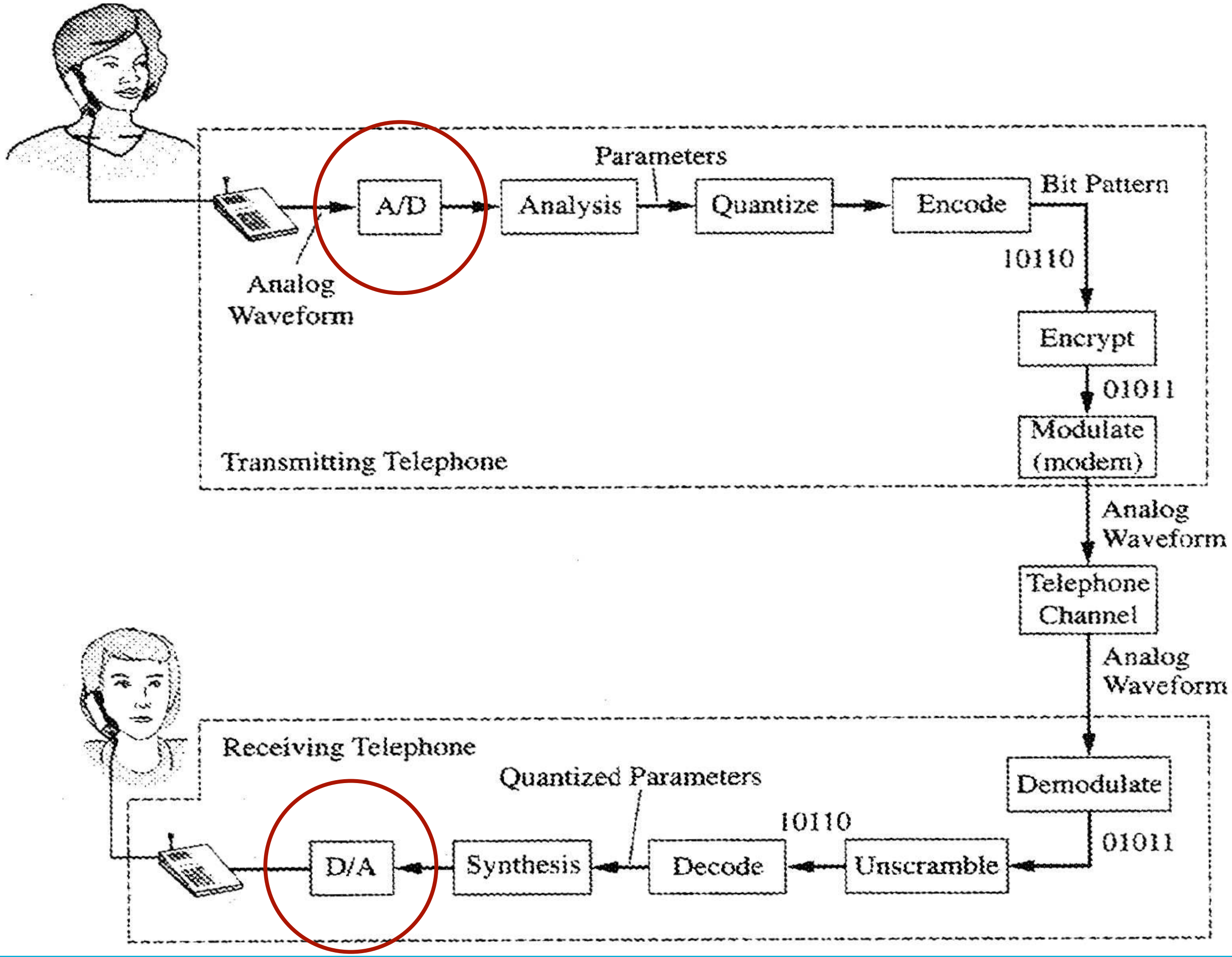
Example: Speech Coding



Block Diagram of GSM Cell Phone

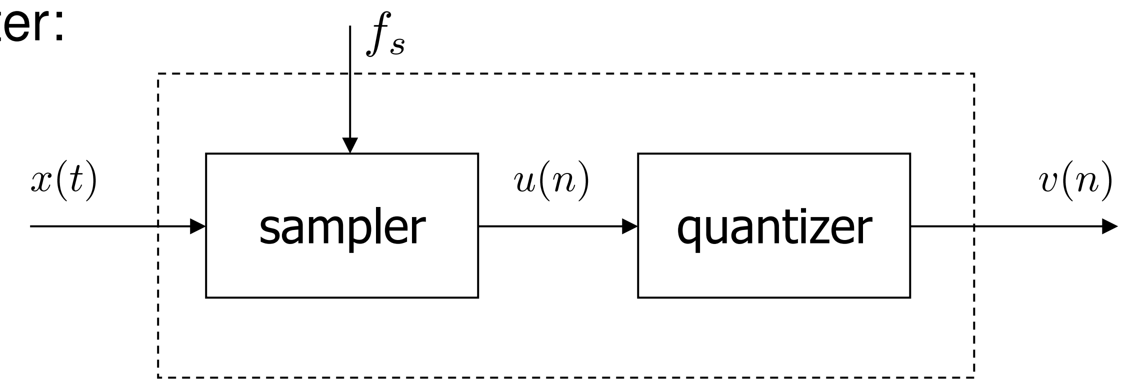


Example: Speech Coding

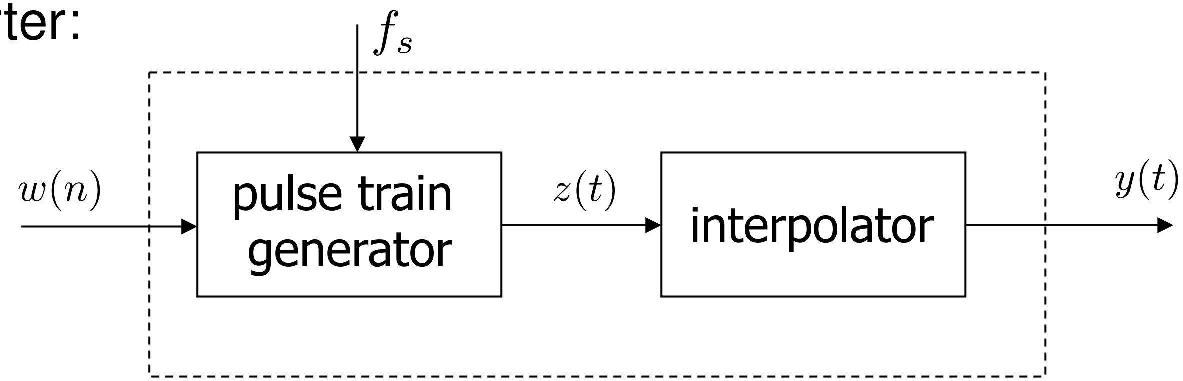


A/D and D/A Conversion

A/D converter:



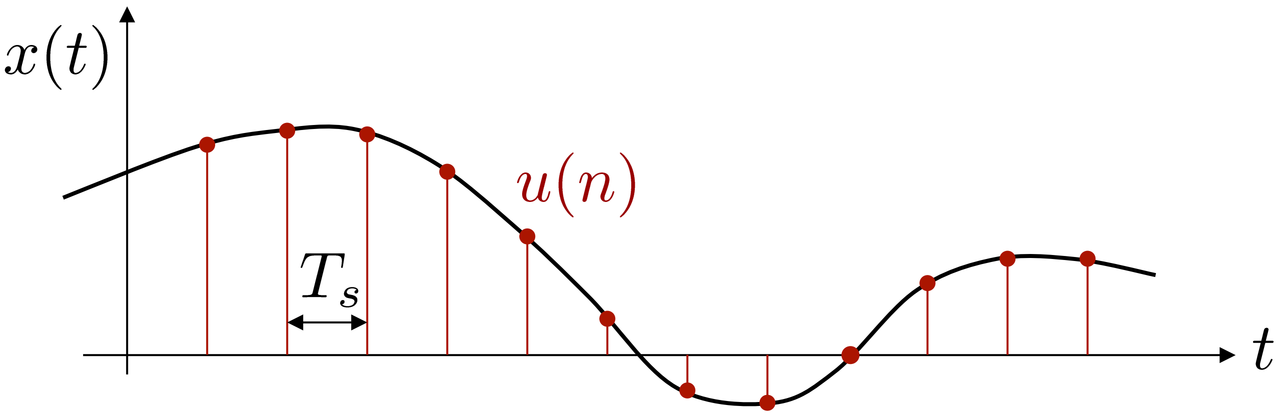
D/A converter:



Sampling

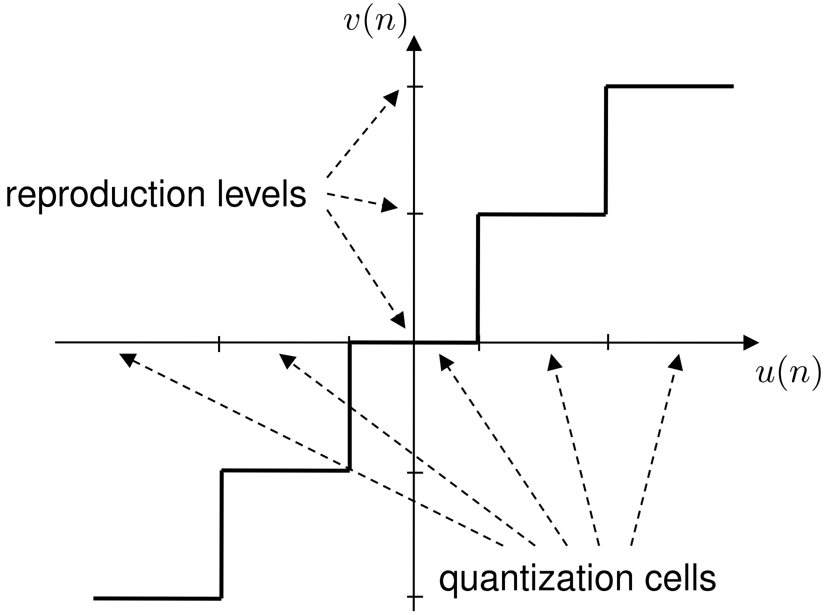
To process a continuous-time signal using digital signal processing techniques, it is necessary to convert the signal into a sequence of numbers. This is usually done by *sampling* the analog signal, say $x(t)$, periodically every T_s seconds to produce the discrete-time signal $u(n)$ given by

$$u(n) = x(nT_s), \qquad -\infty < n < \infty$$

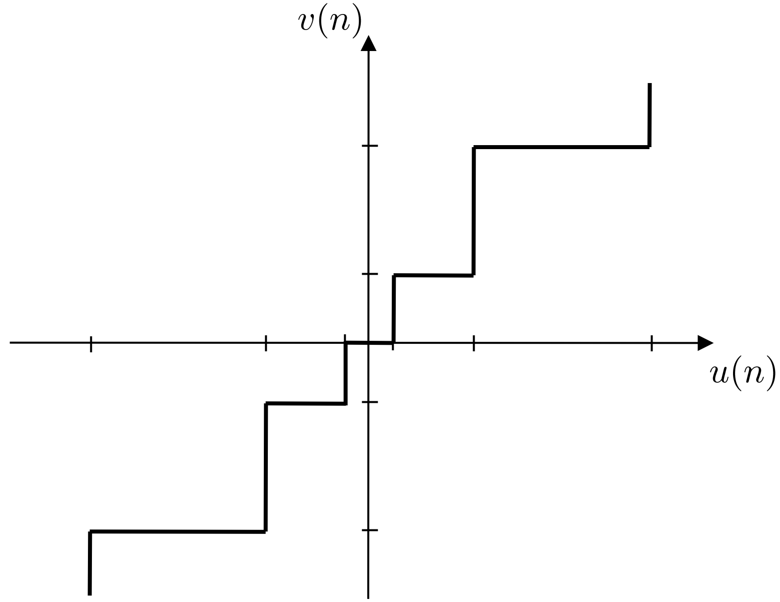


Quantization

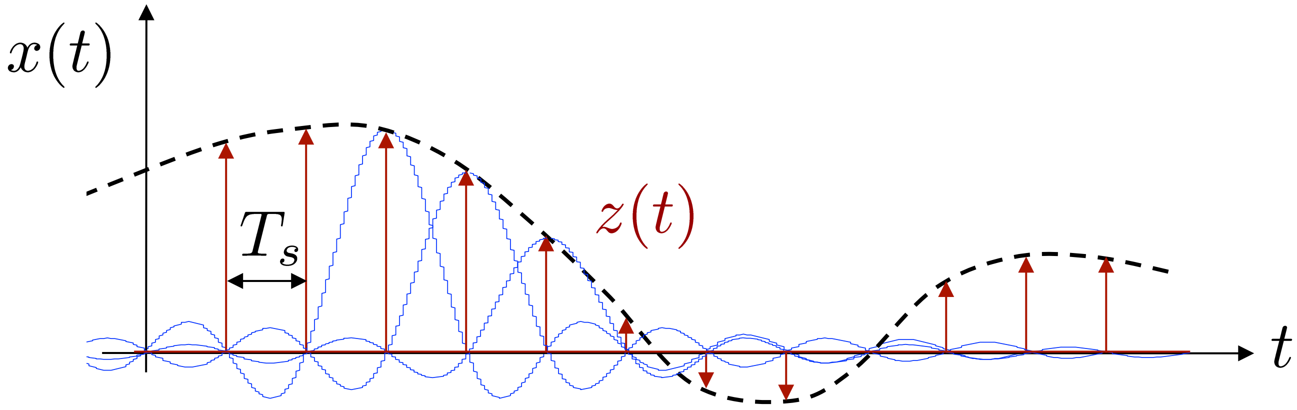
uniform quantization



nonuniform quantization



Reconstruction

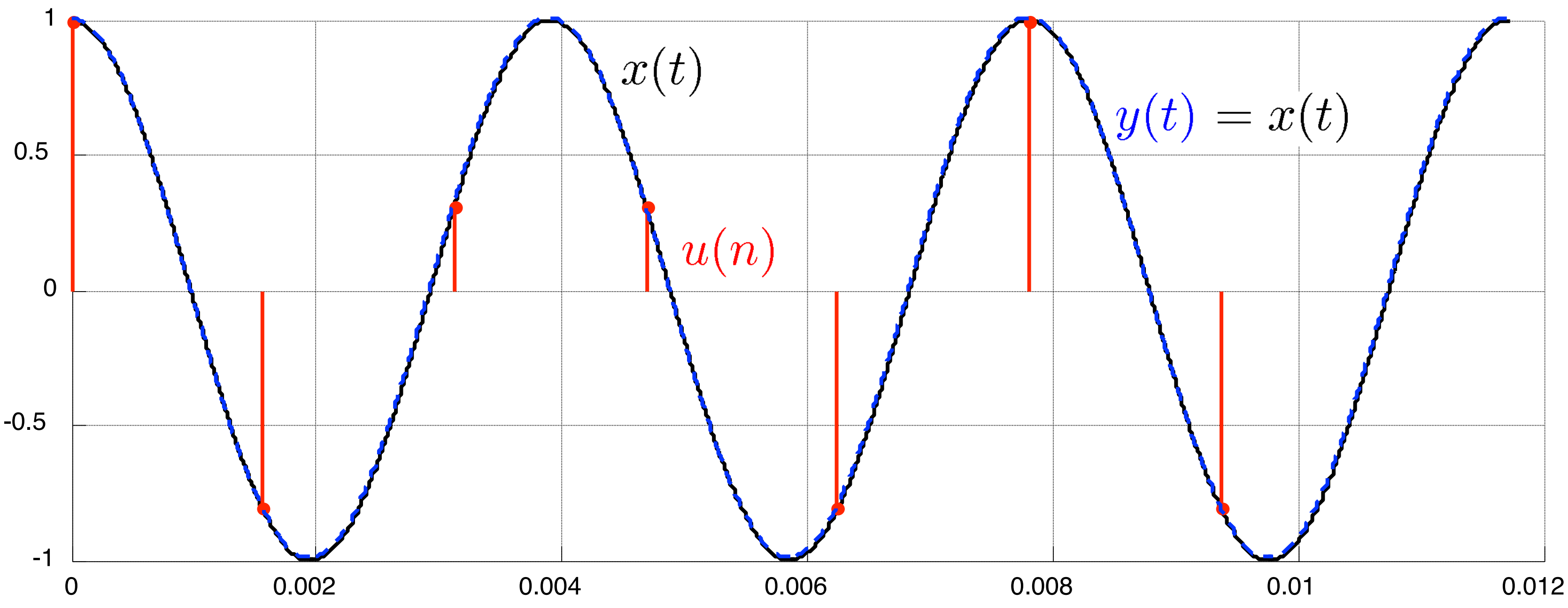


Assuming that $f_s > 2B$ (sampling theorem), we can perfectly reconstruct the analog signal $x(t)$ out of its samples using interpolation techniques, where

$$z(t) = \begin{cases} u(t/T_s) = x(t), & \text{for } t = nT_s \\ 0, & \text{otherwise} \end{cases}$$

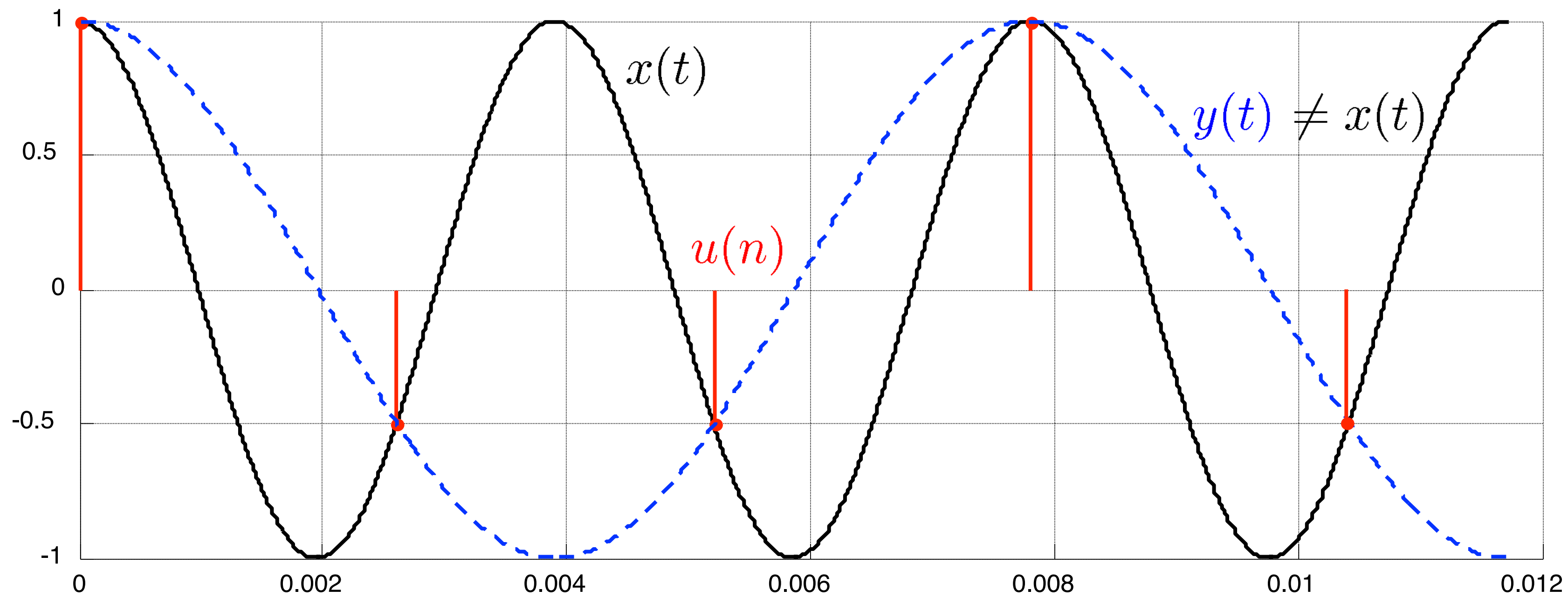
Sampling and Reconstruction

$f_s > 2f_0$:

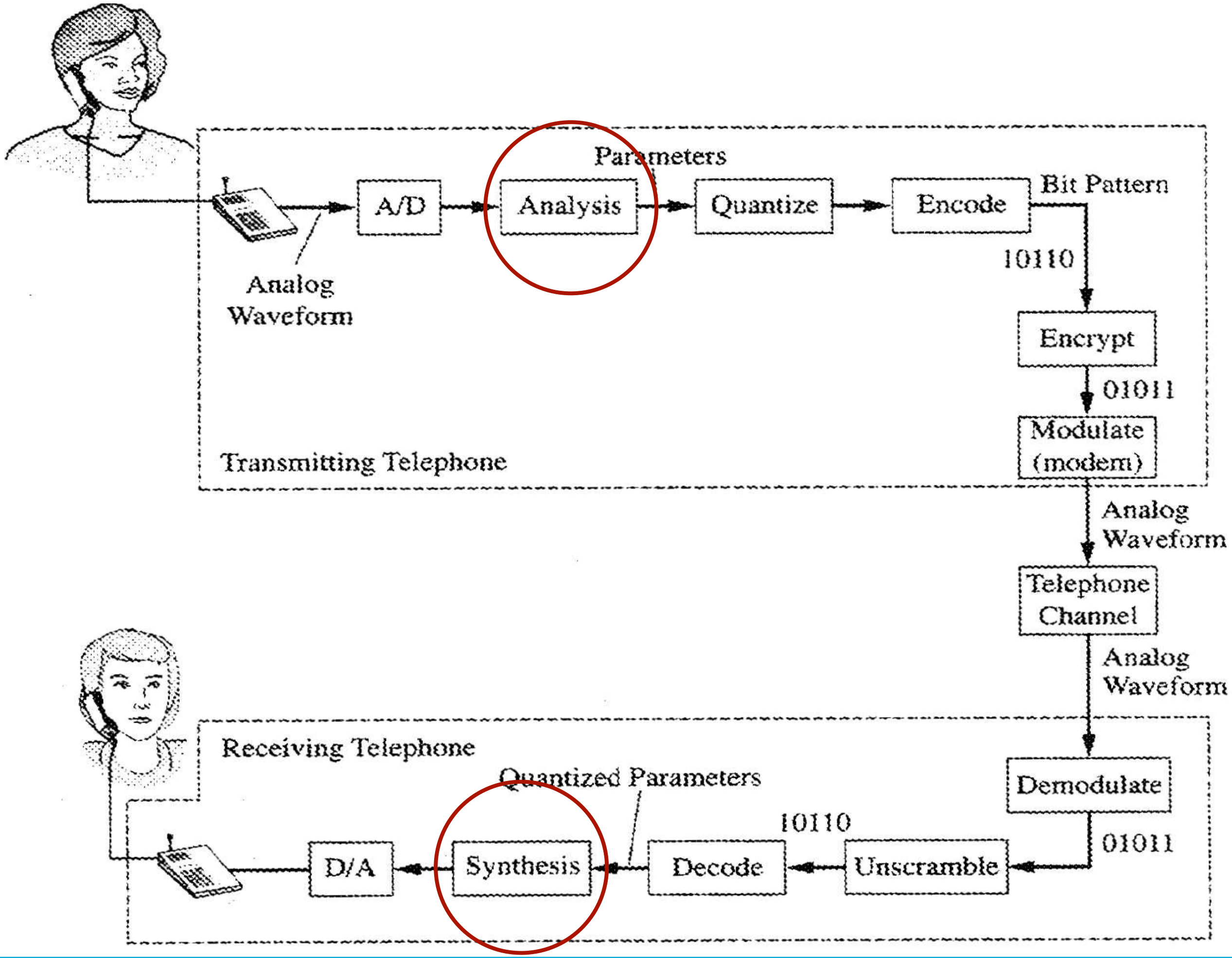


Sampling and Reconstruction

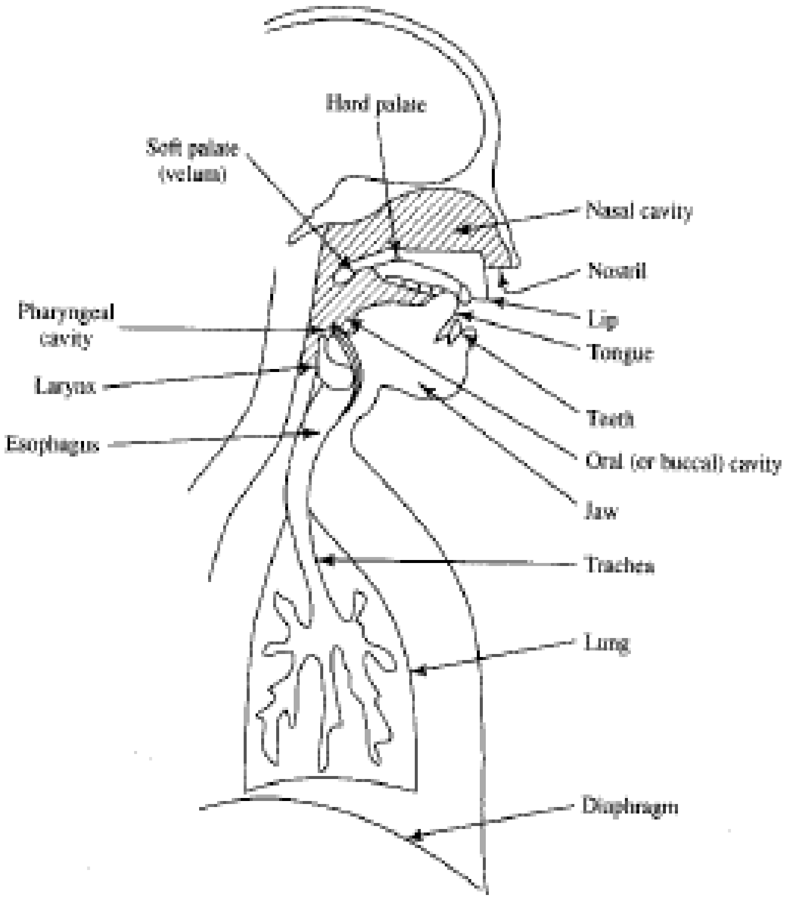
$f_s < 2f_0$:

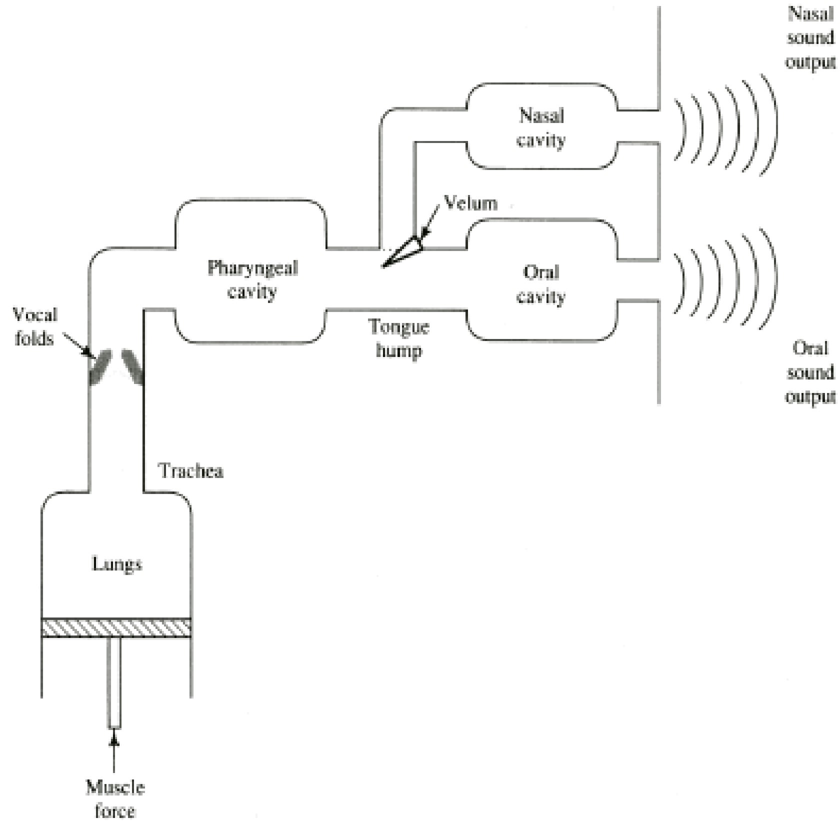


Example: Speech Coding



Speech Production Model

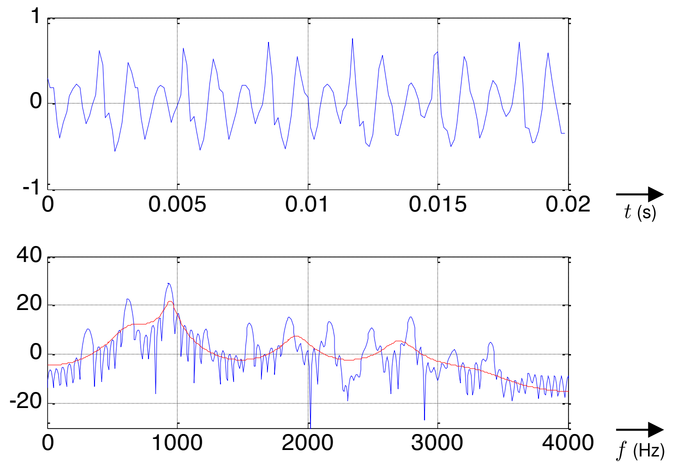




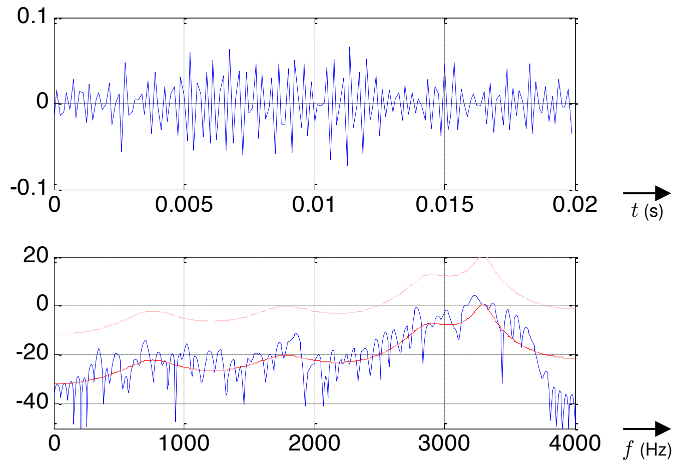
Voiced vs. Unvoiced Speech

- Voiced: Air pushed through the glottis which oscillates, generating quasi-periodic puffs of air (e.g. vowels /a/, /i/, etc.)
- Unvoiced: Air forced through constriction somewhere along the vocal tract (e.g. /s/, /f/).

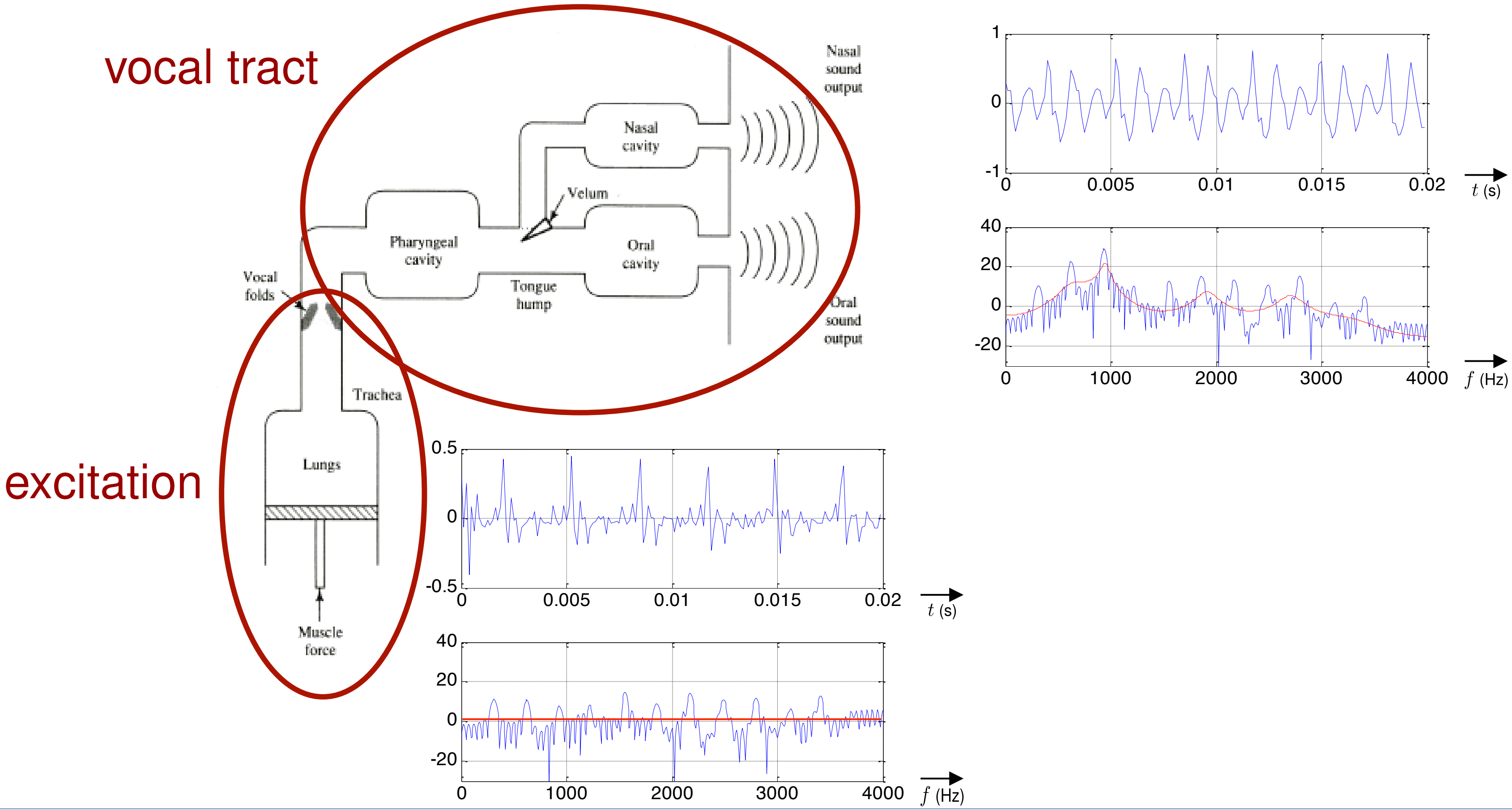
voiced:



unvoiced:

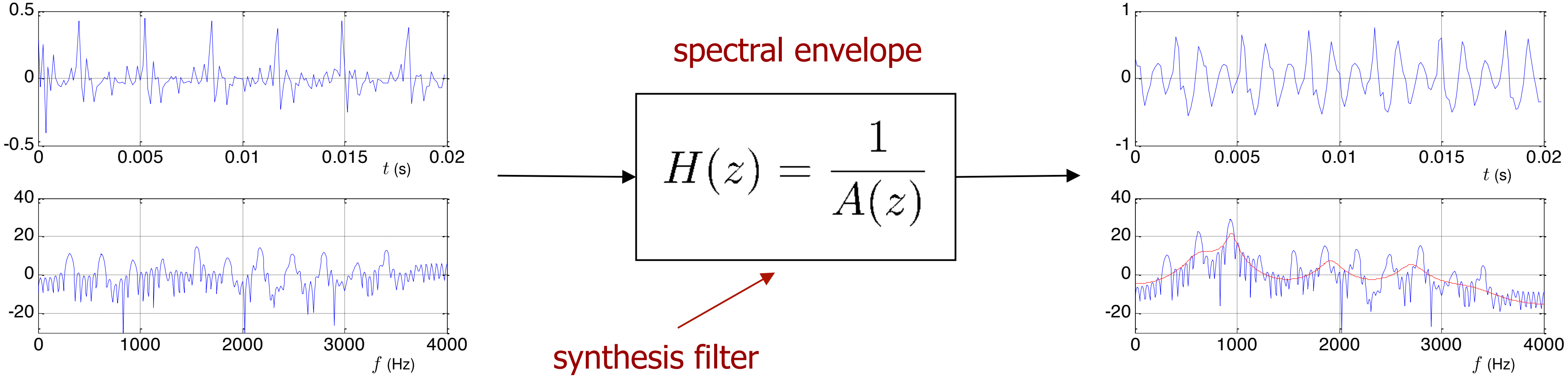


Speech Production Model



Speech Production Model

- Discrete-time linear source-filter model of speech production



Speech can be synthesized if we know:

1. spectral envelope
2. excitation signal

Linear Prediction

How to find the filter coefficiens of the synthesis filter?

- linear prediction:

$$\min_{\alpha_1,\ldots,\alpha_p} \|x(n) - \underbrace{\sum_{k=1}^{p} \alpha_k x(n-k)}_{p\text{th-order prediction } \hat{x}(n)} \|^2$$

- prediction error:

$$e(n) = x(n) - \hat{x}(n) = x(n) - \sum_{k=1}^{p} \alpha_k x(n-k)$$

Linear Prediction

Equivalently, we have

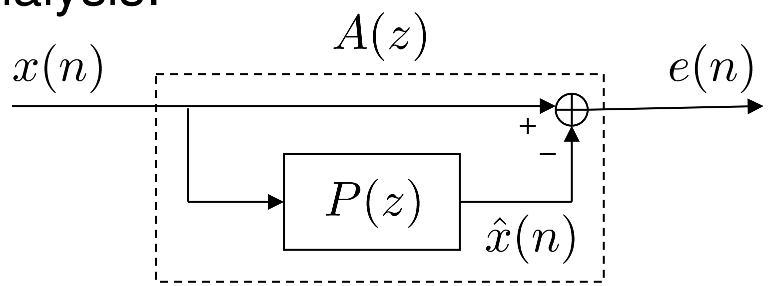
$$E(z) = X(z) - P(z)X(z) = (1 - P(z))X(z) = A(z)X(z)$$

and thus

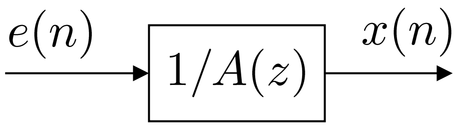
$$X(z) = E(z)\frac{1}{A(z)} \qquad \text{(assuming } A(z) \text{ is invertible)}$$

speech = excitation $*$ synthesis filter

Analysis:



Synthesis:



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Example: Speech Coding

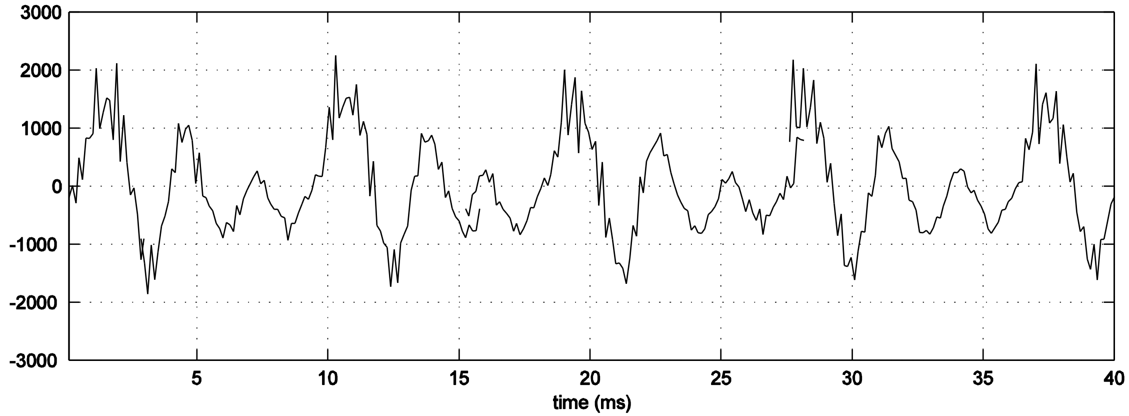
- For each time frame, the prediction coefficients (typically 10 per frame) are quantized and coded
- How about the excitation signal? Do we need to transmit the complete (quantized) waveform?
  - in mobile phones, the excitation is chosen out of a pre-trained codebook (typical 1024 entries) $\Rightarrow$ waveform coding of speech
  - in the vocoder, the excitation signal is modelled by either a pulse train (voiced sounds) or random noise (unvoiced sounds) $\Rightarrow$ parametric coding of speech

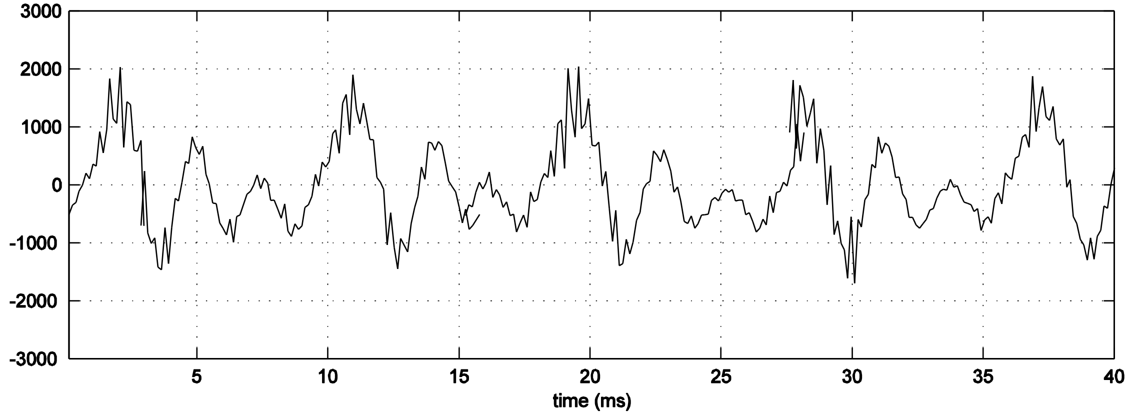
Stochastic vs. Deterministic Signals

- In many applications we are not interested in the exact waveform, but in properties of the signal, like mean value, total energy, spectral envelope, etc.
- For those applications, we consider the waveform as a particular realization of a process that is random in nature, but satisfies desired properties, like the ones mentioned above
  - every time we generate such a signal, the signal values itself will differ from generation to generation

Voiced Speech

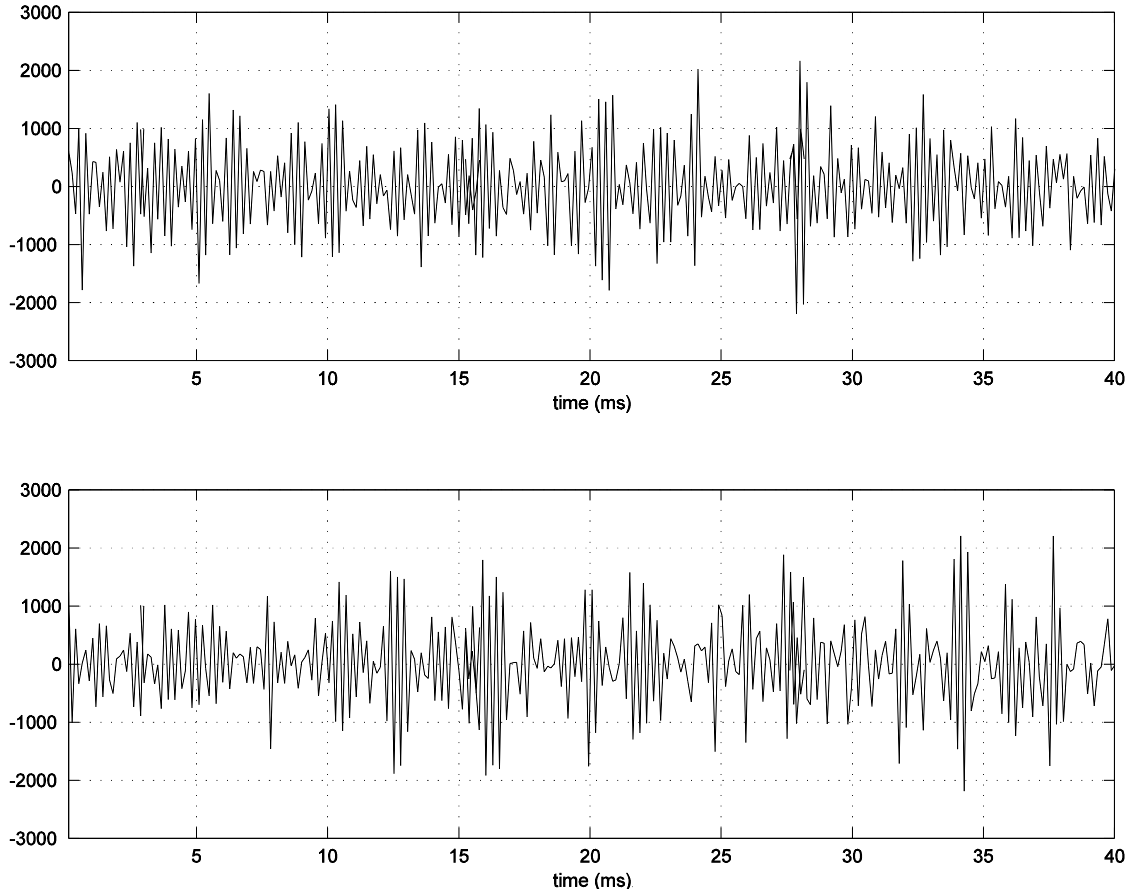
Example:





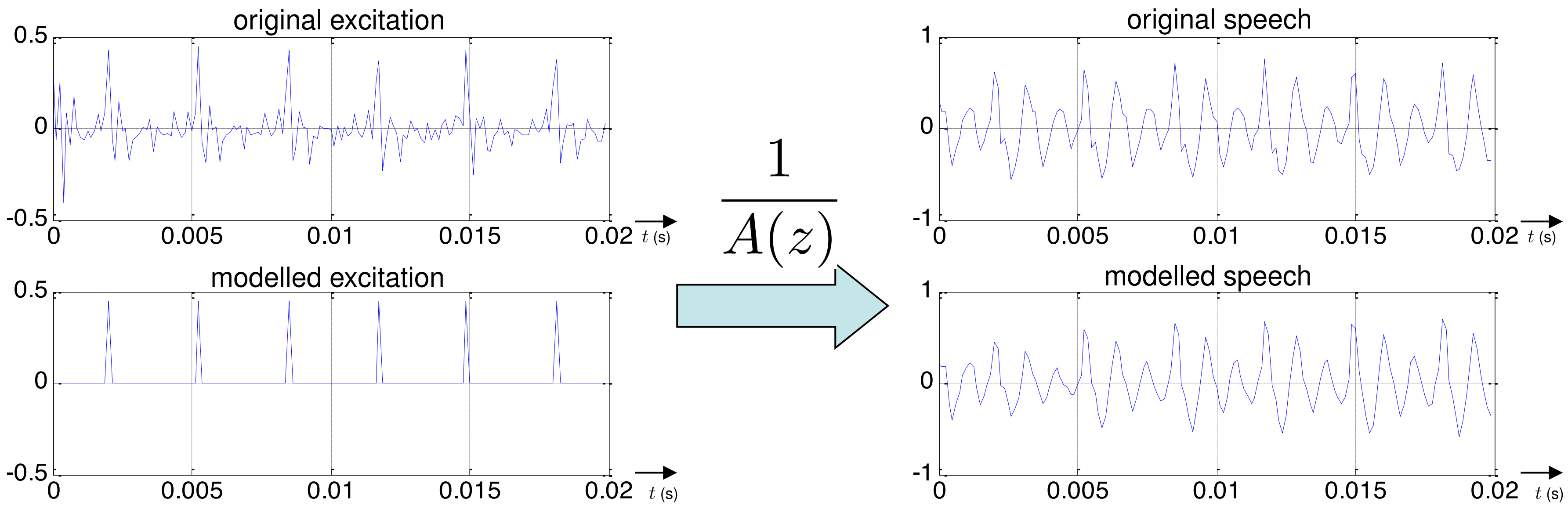
Unvoiced Speech

Example:



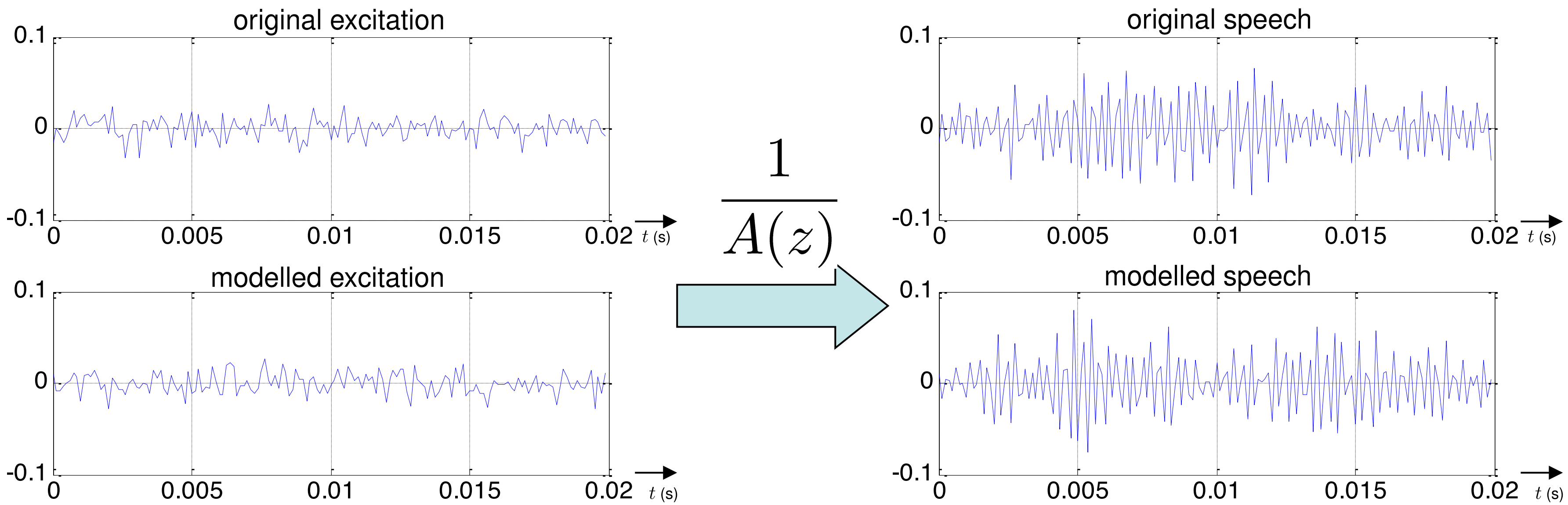
Example: Speech Coding

voiced speech:



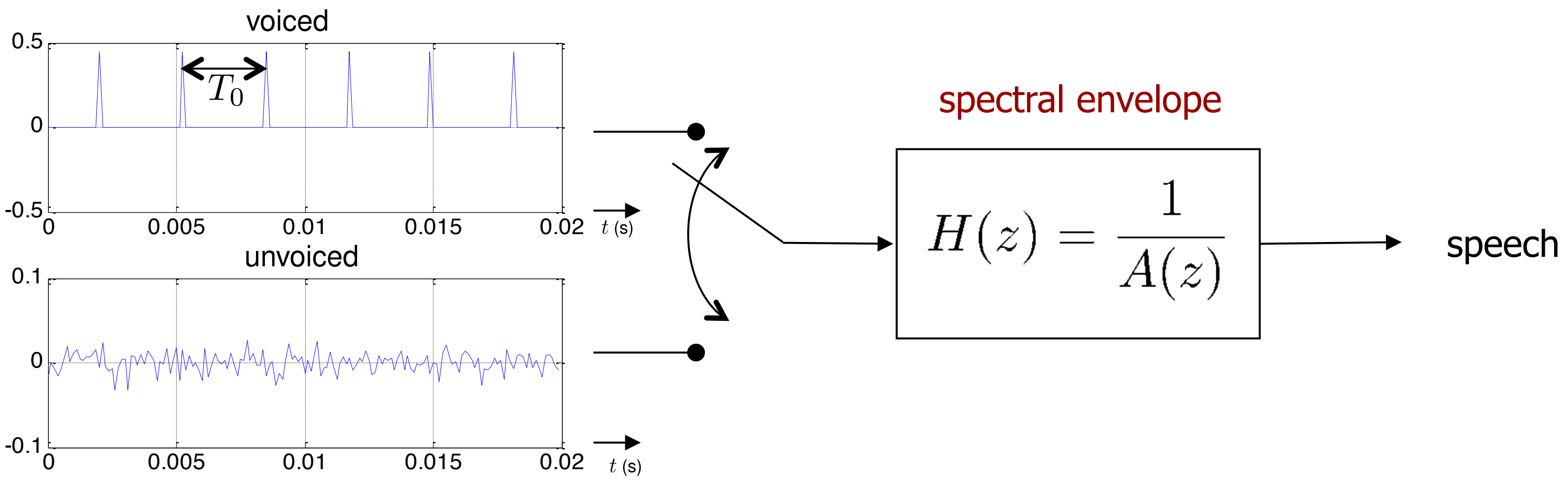
Example: Speech Coding

unvoiced speech:



Vocoder

- Discrete-time linear source-filter model of speech production



Speech can be synthesized if we know:

1. spectral envelope
2. pitch period T_0

(can be computed using the auto-correlation function)

Demo Vocoder

English female speech, $f_s = 8$ kHz, mono, 8 bit/sample

- original
- modelled
- modelled as unvoiced speech

German male speech, $f_s = 8$ kHz, mono, 8 bit/sample

- original
- modelled
- modelled as unvoiced speech