The bachelor's degree in ICT Systems Engineering Teaching results in DSP and S&S

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Signals & Systems: Contents (Q4, 3h+1h)

- Introduction time domain: transformations of the independent variable, definitions, basic signals, properties of systems, impulse response and convolution
- Introduction frequency domain: Fourier transform, windowing, FT & FS relations
- Random signals & noise: noise equivalent bandwidth, Friis formula, noise factor (F), noise figure (NF), SNR
- Signal Processing: AM (DSB, SSB), FM (NBFM, WBFM) & PM modulations, sampling theory, TDM, some implementations
- Filter design: active (templates, Butterworth & Chevichew Low-pass, second order synthesis from the poles), passive (some formulas)

In the lab

- Octave introduction
- QAM modulation (simulation)
- Sampling theory (implementation)
- Secretitzador (implementation)
- A narrow-band (1 kHz) voice ultrasound link at 40 kHz using a SSB modulation & non coherent demodulation (illustrative)

Teaching results

Achieved goals

A lot of exercises to handle functions (scale change,delay, convolution)

Tried but not achieved goals

Understanding the possibilities of Octave (matrix formulation, Python inertia...)

Future goals

- In future subjects: PDS can finish the SSB demodulation lab
- Other subjects can: ?
- Changes: Add PWM modulation to be used in PDS

Digital Signal Processing: Contents (Q5, 2h+2h)

- Different approaches to DSP: simulation of analog systems, digital implementation
- Sampling & quantization: sampling, aliasing, undersampling, noise quantization, ADC, DAC
- The z-transform: equivalent to the Laplace transform in Circuit Theory, properties, filtering (transient & steady state response)
- Frequency response: sinusoidal steady state response, basic filter design, zeros&poles
- DFT: FT & FS of analog & discrete signals, DFT, interpretation, filtering
- Calaix de sastre: filter design, FIR, IIR, direct form I, direct form II, finite precision arithmetic, echo cancelation

Simulation in the lab: using Octave

- Audio pseudo-random generator: rand.m, linear feed-back shift registers $\boxed{x^{16} + x^{14} + x^{13} + x^{11} + x + 1}$
- Sound card: sampling from Octave/Matlab (configuration problems in Linux-Octave), frequency limitations in generation (alias mirrored at 22050 Hz), antialiasing filter, artificial undersampling of modulated signals
- Filtering in the time domain: computing the frequency response of a moving average filter
  $\boxed{y_n = x_n + x_{n-1} + ... + x_{n-m}}$
- Filtering in the frequency domain: filtering a PWM voice modulated signal, PWM noise quantization effects
  $\boxed{X_k = DFT(x_n), Y_k = H_k X_k, y_n = DFT^{-1}(Y_k)}$

Implementation in the lab: using Arduino & FPGA

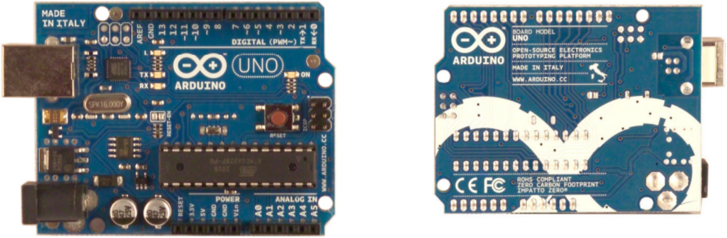

Figure: Arduino UNO

Implementation in the lab: using Arduino & FPGA

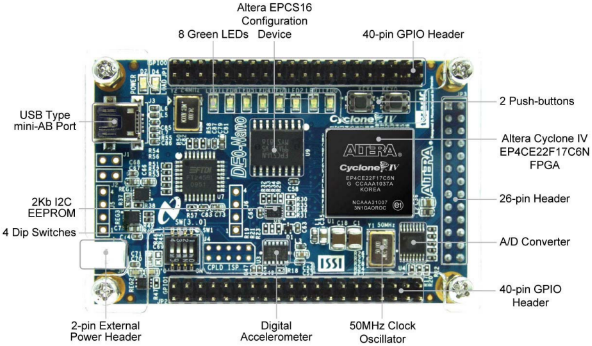

Figure: DE0-Nano

Implementation in the lab: using Arduino & FPGA

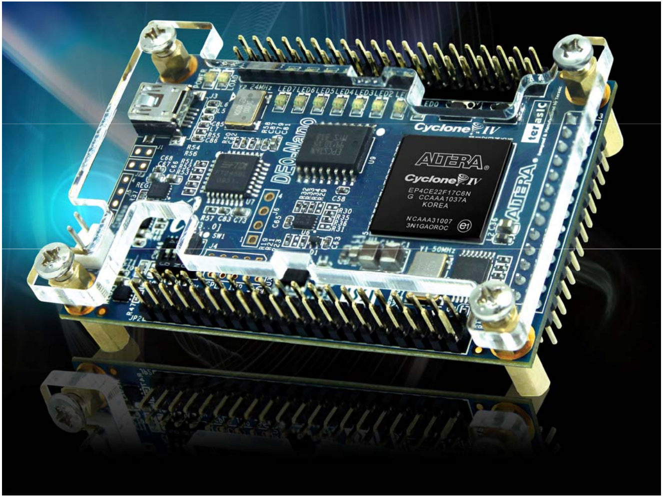

Figure: DE0-Nano

Implementation in the lab: using Arduino & FPGA

- Linking with a previous lab in S&S ▸ Refer to this page: ultrasound link at 40 kHz, SSB modulation, non coherent demodulation
- Sampling with Arduino
  - ADC- Arduino - DAC (PWM+RC), $\boxed{y_n = x_n}$
  - working with TMR, ADC & PWM module
  - ADC & PWM noise quantization effects, programing in C
- Undersampling with Arduino
  - non coherent SSB demodulator,
  - limitations on choosing the sampling period, step=62.5ns
- Implementing a digital oscillator with Arduino
  - computation of $\boxed{y_n = 2\cos(\omega_0)y_{n-1} - y_{n-2}}$ in one sampling period, speed and finite precision arithmetic limitations, floating point refused, fixed point used
- Implementing 4 filters in an FPGA (DE0-Nano board)
  - just seeing the effects of finite precision arithmetic,
  - avoiding VHDL code generation of blocks (PLL, ADC, PWM, filters: MA, generic FIR, Osc, Notch IIR)

Teaching results

Achieved goals

They know how to: use Octave and simulate with a lot of problems, basic digital filters, implement digital filters

Tried but not achieved goals

Consolidate concepts, finite precision arithmetic

Future goals

- In future subjects: Use z-transform in Control, get a deeper insigth on audio and image processing
- Other subjects can: previously work some of the Arduino modules, finite precision arithmetic problem, PWM
- Changes: Ordering and making coherent the labs, small changes in Contents, data aquisition and/or DSP card?