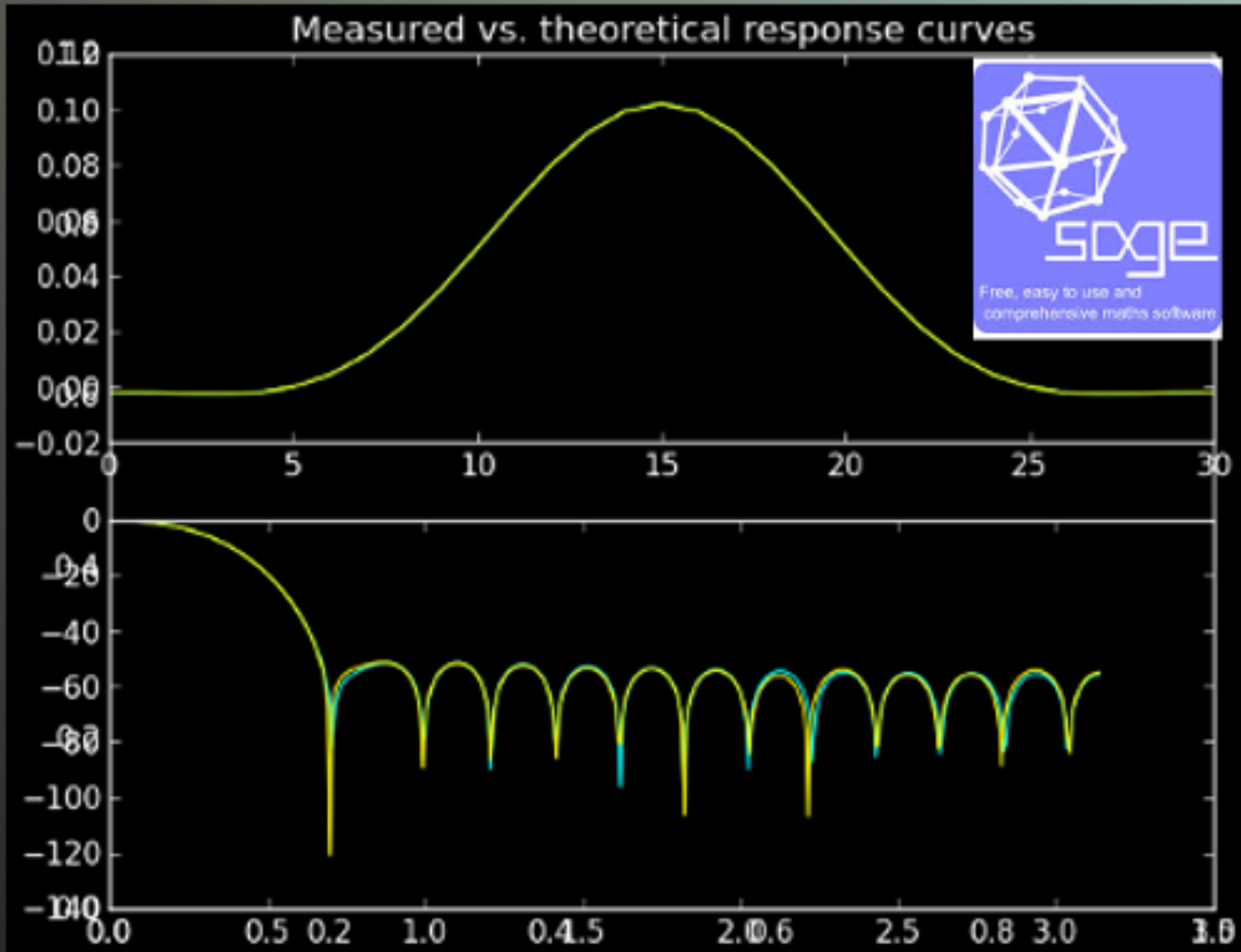
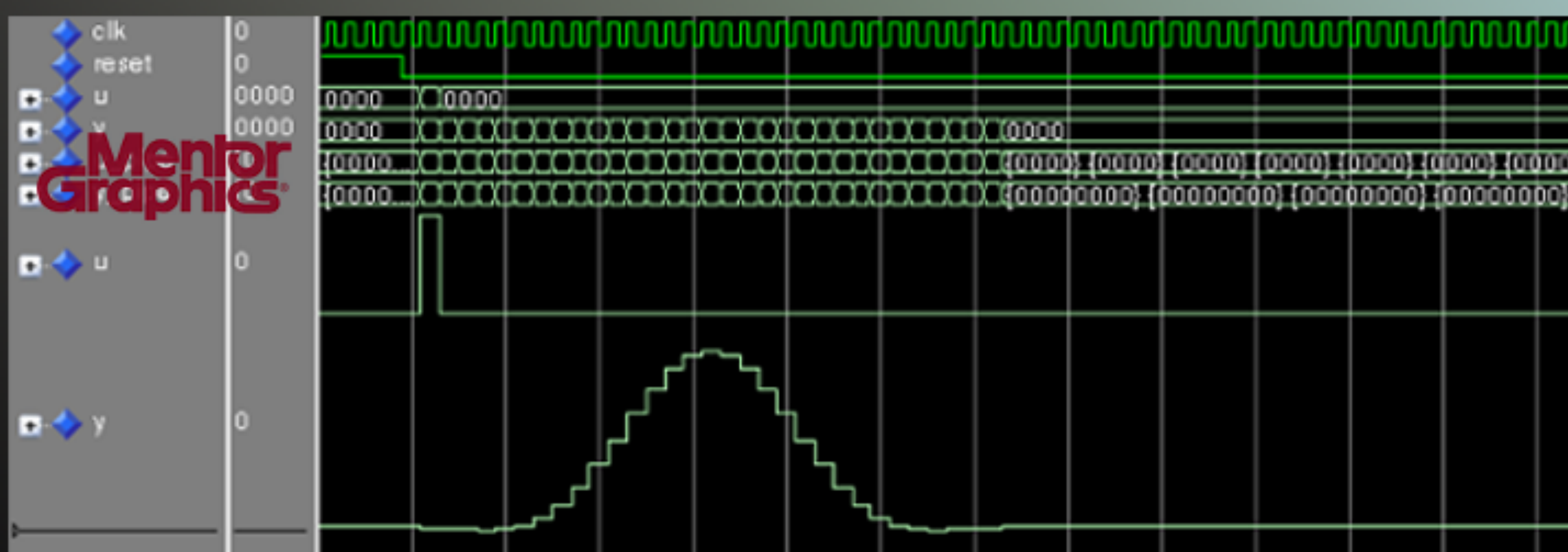


DIGITAL SIGNAL PROCESSING WITH VHDL

- GET HANDS-ON
- FROM THEORY TO PRACTICE IN 6 DAYS
- MODEL WITH SCILAB, BUILD WITH VHDL
- NUMEROUS MODELLING & SIMULATIONS
- DIRECTLY DESIGN DSP HARDWARE

Brought to you by:



DAY 1

INTRODUCTION

Overview of digital signal processing. Applications of DSP.

DSP FUNDAMENTALS

- Cosine signals and Complex Exponentials as the building blocks of signals
  Categories of signals:
    - Continuous and Periodic Signals in Time Domain
    - Continuous and Non-Periodic Signals in the Time Domain
    - Discrete and Non-Periodic Signals in the Time Domain
    - Discrete and Periodic Signals in the Time Domain

- Time to Frequency conversion of Discrete and Periodic Signals :- DFT & FFT

- Computational Efforts of DFT and FFT

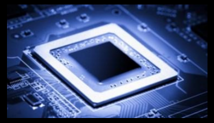
Lab Exercises

DSP FUNDAMENTALS(CONT.)

DAY 1

- RC System as a Basic Analog System
- Characteristics of an RC System
- RC System Equivalent First Order Discrete Time System
  - Difference Equation
  - Characteristics
- Impulse Response and Frequency Response
- System Function of RC System and its Discrete Time Equivalent
- Laplace Transform and Z Transform
- Mapping from S-domain to Z-domain
  - Impulse Invariant Transformation
  - Bilinear Transformation and
  - Matched Z-Transformation
- Difference Equation and System Function of DSP Systems
- Implementation of Difference Equation
- Filtering in the frequency domain.

Lab Exercises



INTRODUCTION

Overview of DSP and its applications in digital logic systems. Number storage and representation formats.

DSP FUNDAMENTALS

- Discrete-time (DT) vs. continuous-time (CT) systems and notations. Difference equations.
- Convolution sum: A widely-used concept for DSP designs.
  - The unit impulse excitation and response for determining system characteristics
  - Concept of convolution
  - Similarities between the DT convolution sum and the CT convolution integral
  - Responses of systems to other standard impulse excitations.
- Time-to-frequency transformations: the Laplace, Fourier, and z transforms.
- The Fourier transform.
  - The discrete Fourier transform (DFT).
  - The fast Fourier transform (FFT): a popular DFT algorithm.
  - Designing the FFT algorithm for an FPGA/ASIC.
- The z-transform.
- Similarities between the DT z-transform, and the CT transforms such as Laplace (s-domain) and Fourier transforms.

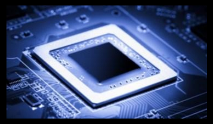
INFINITE IMPULSE RESPONSE SYSTEMS

DAY 3

- Applications of IIR System
  - Sine wave generator
  - Goertzel Algorithm
  - Notch Filter
- Design Steps involved in IIR System Design
- Design methods for All Pole Analog Systems
  - Butterworth
  - Chebyshev
  - Bessel
- Transformtion of All Pole Analog Systems into Digital Systems
- Prototype Low Pass Filter for High Pass, Band Pass and Band Stop Filters
- Frequency Domain Transformation
  - Low Pass to High Pass, Band Pass and Band Stop
- Examples for Design and Implementation of Low Pass, High Pass, Band Pass and Band Stop

Lab Exercises

DAY 4



- IIR filter design and simulation
  - Discrete Butterworth and Chebyshev filters
  - Designing an IIR filter from a given Sage/SciLab/Matlab model
  - Functional simulation flow for pre-silicon verification
  - Logic simulation of digital Chebyshev IIR filter model
- IIR filter synthesis
  - Representing filters with block diagrams
  - Deriving hardware from block diagrams
  - Shift registers, multipliers and adders
  - Developing the control unit
  - Synthesis of a Chebyshev IIR filter
- FPGA / ASIC implementation of IIR filter
  - Automatic placement-and-routing (APR)
  - ASIC/FPGA floorplanning, chip layout
  - Design constraints
  - Design assembly
  - Chip / FPGA implementation of a Chebyshev IIR filter.
- IIR filter hardware verification
  - Hardware verification flow for post-silicon verification
  - Overview of the development board, clock and reset assignments, JTAG set-up.
  - Verifying the IIR filter.

FINITE IMPULSE RESPONSE SYSTEMS

- Disadvantages of IIR Systems
- Advantages of Non-Recursive Systems
- Finite Impulse Response Systems
- Linear Phase FIR Systems
- Design methods of FIR Systems
    - Windowing Method
    - Frequency Sampling Method
    - Remez Algorithm
- Implementation Methods of FIR Systems
    - Linear Buffer method
    - Circular Buffer Method

Lab Exercises

ADAPTIVE FIR SYSTEM FOR DSP APPLICATIONS

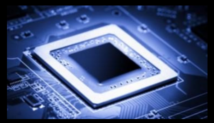
- Noise cancellation
- Echo Cancellation
- Spectral Subtraction based Noise Cancellation
- Wiener Filtering

2-DSP

- 2-D Convolution
- 2-D FFT
- 2-D FIR Filtering
- Median Filter Implementation

Lab Exercises

DAY 6



- FIR filter design and simulation
  - Designing an FIR filter from a given Sage/SciLab/Matlab model
  - Logic simulation of digital FIR filter model

- FIR filter synthesis and FPGA/ASIC implementation
  - Synthesis of an FIR Filter
  - Chip / FPGA implementation of an FIR filter

- FIR filter hardware verification
  - Verifying the FIR filter

- Optimising for speed
  - Overview of pipelining
  - Pipelining the datapath
  - Pipelined FIR Filter.

DAY 6

Adaptive equalisation/filtering

- Overview of estimation theory
  - Maximum-likelihood (ML) estimators
  - Designing and implementing an adaptive channel estimator using ML sequence detection
- Adaptive linear equalisers
  - Zero-forcing equaliser. Least-mean-square (LMS) equaliser: Mean-square-error minimisation
- Adaptive decision-feedback equaliser (DFE)
  - Designing and implementing an adaptive DFE
- Recursive least-squares (RLS) algorithms for adaptive equalisation
  - Wiener equaliser / filter (Wiener algorithm)
  - Lattice filter (linear prediction algorithm)
  - Designing and implementing a Wiener filter
- Blind equalisers

Contact:

info@tauhop.com | +60-16-333-0498