Digital Signal Processing Education: Technology and Tradition

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ABSTRACT

In this paper we discuss a DSP course presented to both university students and to participants on industrial short courses. The "traditional" DSP course will typically run over one or two semesters and usually covers the fundamental mathematics of z-, Laplace and Fourier, followed by the algorithm and application detail. In the courses we will discuss, the use of advanced DSP software and integrated support software allow the presentation time to be greatly shortened and more focussed algorithm and application learning to be introduced. By combining the traditional lecture with the use of advanced DSP software, all harnessed by the web, we report on the objectives, syllabus and mode of teaching.

1. INTRODUCTION

Over the last 10 years DSP has clearly moved from being an optional and probably graduate EE course to being core curriculum for all undergraduate students. Furthermore in the last 5 years, DSP classes have been slowly integrating into computer science degrees, and courses based around multimedia and information technology. So what has changed in the last 10 years? Well, clearly the technology of DSP has fully matured and the prevalance of modems, sounds cards, CDs, MP3 formats, digital communications and of course the internet has precipitated the enormous growth in the commercial sector. This growth is supported by the expansion and broadening of DSP courses at our educational institutions.

Mathematical Prerequisites: Nothing has fundamentally changed about DSP theory and algorithms; it is still the same digital filters, mathematical transforms (z- and Laplace), least squares analysis (LMS, RLS) and (some) non-linear processing that was clearly presented by our lucid DSP grandfathers in the 1960s and 1970s.

Technology Change: The key change in the last 10 years for DSP education are the same as those for the practising engineer: technology. The increase in availability of powerful desktop computers and advanced DSP communication analysis software has allowed both the practising engineer and the educator to perform extremely advanced and involved simulations with just a few clicks of the mouse.

Esoteric DSP: Despite the availability of powerful supporting software DSP is usually seen by potential students to be an esoteric and very mathematical subject. Our view is that this is because the traditional means of teaching DSP is both esoteric and (overly) mathematical. For example, while we may rely on the foundation mathematics of the z-transform, for the practising DSP engineer the z-transform is essentially a scripting notation for software analysis. The DSP design and analysis software will do all of the necessary math required to investigate stability and transient behaviour.

Therefore in this paper we will present the aim, aniticpated learning outcomes, syllabus and most importantly the mode of teaching for our DSP courses to very varied audiences.

2. COURSE PRESENTATION

The course discussed below is presented to three different groups. The precise mode of presentation, the content, the timing and emphasis changes depending on the audience and their background. The presentations are to:

- University of Strathclyde, Information Technology Students:
  One semester DSP course, 24 lecture, 20 hours laboratory.
- Industry: 5 day Short Course for Industry at UCLA Extension.
  50% lecture/demonstration and 50% DSP software.
- DSP Comms Industry: 3 to 4 day on-site short course.
  50% lecture/demonstration and 50% DSP software.

In all cases we present from the same set of presentation materials, and allow the student to progress to their required level.

2.1 Objectives

The learing objectives of the courses are to educate students to:

- Analyse discrete time systems using time domain mathematics and frequency domain/ z-domain mathematics;
- Understand the fundamental theory relating to sampling rate, quantisation noise and the architecture of a generic DSP system.
- Design and implement FIR, IIR, and adaptive digital filters for real world applications in digital audio and acoustics, and telecommunications;
- Understand the theory and real world application of adaptive signal processing systems;
- Understand the key DSP theory of signal source coding and compression;
- Understand the theory and advantages of over-/under-sampling, multirate, and noise shaping;
- Undertake DSP system design using advanced analysis and design software;
- Acquire the know-how to implement real time fixed and adaptive

digital filters using DSP simulation software and real time DSP processor hardware;
- Apply DSP theory and algorithms in the application domains of modern computing, multimedia systems, and communication systems.

The level to which individual objectives are pursued of course depends on the student or engineer's requirements. However given the comprehensive nature of the materials available, and the professional grade software used, it is straightforward to modify the scope of the course.

2.2 Pragmatic Education

The syllabus is based around the well known traditions of DSP education. Probably the key differential is the de-emphasis of the mathematical detail. Hence, for example, we are not aiming to teach the theory of z-transforms; we are aiming to teach students what the z-transform means (a simple change of a signal flow graph) and how to use design software to assist in this type of analysis. As another example, we are not aiming to teach the dependence of the LMS stability on the eigenvalues of the covariance matrix of the input signal; we are aiming to intuitively reason why a filter would go unstable, and how to control this in practical system via the step size and then design and implement systems in SystemView. If the student wants the detailed math, then this is made available in the presentations, but is not a requirement of the course.

2.3 Syllabus

The general syllabus we use is presented below. General timings are given which of course vary slightly depending on the audience.

Signal Processing Review (1 hour)
- Signals, Systems and Applications
- Amplification, distortion, and noise
- The 1990s DSP Revolution to Software Radio in 2000

The Generic DSP System (2 hours)
- ADCs and DACs / Signal Conditioning
- Anti-alias and Reconstructions Filters
- Distortion, Quantisation Error and Noise
- The Nyquist Sampling Rate
- z-domain representation and transforms

Frequency Domain Analysis (4 hours)
- Periodic, aperiodic and random signals
- The DFT, FFT and Power Spectra
- Spectral Leakage and Data Windowing
- Modern spectral analysis
- Time/Frequency Representation

Digital Filtering (4 hours)
- FIR and IIR Digital Filters
- Digital Filter Design Parameters and methods
- All-pass, CIC, MA, ARMA, comb filters etc..
- Poles and zeroes and the z-notation
- Bit true simulations

DSP Software/Hardware (2 hour)
- The Generic DSP Processor Architecture
- Application Specific Integrated Circuits
- DSP Design and Analysis Software

Signal (Audio) Source Coding (3 hours)
- Waveform speech coding (ADPCM etc)
- Speech model coding
- Linear prediction techniques (LPC, CELP etc)
- Transform coding
- Perceptual/Psychoacoustic Coding

DSP Audio/Baseband Processing (3 hours)
- Over-/under-sampling; Sigma delta ADC/ DACs
- Sample rate; decimation & interpolation
- Quantisation noise shaping

Adaptive DSP Algorithms (3 hours)
- Least squares (LS) minimization
- Least mean squares (LMS)
- Channel equalisation / Inverse system identification
- Echo Control for feedback suppression
- Acoustic echo control /noise control

Computationally Efficient DSP Linear Systems (2 hours)
- Uninform and Octave Subband filter banks
- Quadrature mirror filters
- Polyphase implementation

DSP Baseband Communications (4 hours)
- Information theory
- AM/FM/PM modulation; ASK/PSK/FSK Signalling
- Pulse shaping / Matched Filtering
- Data equalisation
- QPSK, QAM digital communications
- Bandpass sigma delta
- Error control and coding
- GSM/CDMA Perspective

Now, the experienced DSP lecturer may ask: *How can you possibly teach these topics in such a short time?*

First, we should make clear that our industry based audiences are capable engineers from other disciplines (analog, ASIC, digital and so on) with math skills and usually clear objectives on what they require to learn and by when! Second, we generally know that lecturing is usually a directive session rather than one of students absorbing and understanding. In the class lectures we make extensive use of SystemView, audio examples and animating slides. All of these materials are then made available in the lab sessions, and perhaps most importantly given to the students to take home on CDROM. Therefore when the self-study commences (and the real learning begins) we believe the materials will allow the student to quickly learn the rudiments of DSP, and then understand and simulate to an advanced level if of interest.

From experience at UCLA Extension, where have presented to more than 400 professional engineers since 1997, we believe that the philosophy and style work very well. Our overall course evaluation by students is consistently in the range 4.6 to 4.8 out of a possible 5.

3. Course Materials

The same superset 7 volume of materials are used for all three courses [2]:

- A DSP A-Z Reference (500 pages);
- Introductory DSP Class Notes (350 slides);
- Advanced DSP Class Notes (350 slides);
- DSP Communications (400 slides);
- 3rd Generation Mobile Communications (250 slides)
- DSP SystemView DSP Software Workbook (280 pages);
- DSP Hardware Workbook (Motorola DSP56307).

Each course attendee receives a copy of the **DSP***edia* multimedia CDROM which contains:

- Hypertext of all printed notes above;
- Integrated multimedia presentations;
- More than 100 DSP audio demonstrations;
- More than 200 SystemView DSP demos;
- SystemView DSP Evaluation License;
- Links to DSP WWW sites;
- Mnemonic video DSP ideas/tips/animations;
- The DSPClass Navigator programme.

All of the notes and presentation slides on the CDROM are in Adobe PDF format [5] and certain sections can be printed by the student if required. For the lecturer all slides, notes and software are available on the CDROM for OHP projection in the class. Taking this CDROM route thus allows the lecturer to keep all notes up to date by simply adding and modifying sections when required and then remastering the CDROM.

Copyright protection is always an issue of concern and while authors may often desire a widespread use of their materials this must always take place in a controlled way and the illicit duplication of CDROMs is to be discouraged. Hence the information on the CDROM is protected by commercial software for PDFs called FileOpen [4]. To date this has worked very successfully.

4. DSP Software

SystemView by Elanix® DSP design software is used for the DSP laboratory session [1]. This advanced software provides a comprehensive and state of the art DSP toolbox for modern signal processing. This software runs on Windows95/98/NT and is professional grade. SystemView is particularly useful in allowing communication and DSP systems and analysis tools to be presented in a very practical way. After a very short learning curve SystemView can be used to rapidly develop simulations and real time blueprints of very complex communication and DSP systems. We have found it to be the most intuitive and easy to use of the various DSP software packages on the commercial and academic market.

4.1 The DSPClass Navigator

In order to integrate the printed materials and simulations we have developed a navigation program titled *DSPClass*. The aim of this program is to provide a Windows facility to allow and integration of SystemView simulation examples, notes, video clips and mini-lectures. DSPClass will therefore fetch the appropriate PDF documents, simulations, and any supporting video and audio. The key reason for producing the navigator was to ensure that the student working in the lab or following a lecture presentation can navigate through the various teaching media and materials with relative ease and minimum effort.

5. Internet/Intranet Presentation

Currently the course has *not* moved completely onto the internet. This is due to bandwidth requirements (the CD has 580 Mbytes) and reliability concerns. For students to work at home, download over a 56k modem of even 1Mbyte is frustrating and slow. Networks are also invariably busy usually at the most crucial times. Therefore although *DSPClass* will already function over the internet, as far as possible the SystemView examples, PDF information and so on is held locally on the CDROM - only the controlling information comes (optionally) via the internet.

6. Final Comments

For the paper presentation at the conference we are presenting a complete demonstration of the DSPedia notes and materials, of SystemView, and of the integrating function of *DSPClass*. Educators interested in inspecting or adopting our materials, please contact us via email or on the web.

7. References

[1]SystemView DSP Communication Design Software.
http://www.elanix.com
[2]The DSPedia: A multimedia resource for DSP.
http://www.DSPedia.com
[3]UCLA DSP Theory, Algorithms & Application
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