

## Teaching Digital Signal Processing Using MATLAB

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***Abstract:** This article presents an overview of a Digital Signal Processing (DSP) course, based on usage of MATLAB software to support teaching. DSP can be taught at different levels and students need sound mathematical skills. MATLAB is used as a suitable, interactive, programming language and development environment. It is utilised as a tool to support teaching mathematical theory, physical principles and also to develop real life applications. Principles of safety, sound relationships, sequence and reinforcement, praxis, respect for learners, ideas, feelings, actions, immediacy, clear roles, teamwork, engagement and accountability, important for adult teaching, have been used in the course design. By doing projects, students learn how to perform MATLAB programming, learn something about hearing and produce a product that can be used by non-professionals. This paper also presents and discusses the finding of the course evaluation. Student feedback results are presented graphically.*

***Keywords:** DSP, Screening Test, Hearing Aid*

### Introduction

There are many books about Digital Signal Processing, available to be used as textbooks, or reference books: such as one from Richard (1990), together with a very useful, Analogue Devices application book (1990). There are also relatively new ones, like ‘Solving engineering problems with MATLAB’, Dolores (1997), or one from James, Ronald and Mark (1999), with a multimedia approach. Thomas (2000), covers theory in great details, while Childers (2000), treats a specific area of speech processing. There are also many online DSP courses available on the Internet, as well. To do the actual DSP chip programming one has to use an assembler, C, MATLAB, or some other language.

DSP is a complex subject as this variety of resource material shows. A decision was made in this course, to support theory using a new, modern, interactive and powerful tool called MATLAB, as much as possible. In addition to doing revisions on complex numbers, trigonometry, Euler’s formulas, powers and roots, students are encouraged to use MATLAB as a scientific calculator, interactively.

- By following a schedule, and a sequential approach, described in this paper, the educational principle of Sequence and Reinforcement is applied.
- By asking students to start designing useful applications from the very beginning, the principle of praxis is applied.

- By interaction during the course, and asking students for their evaluation of the course, respect for learners, as subjects of their own learning, is shown.

Most of the projects are designed and given to students as team exercises. Andragogy suggests that the teaching role should be more responsive and less directive, Knowles, Burns (1995). According to that, students have their say about methodology, assessment techniques, content, as well as program design.

### **MATLAB as a DSP tool**

MATLAB (matrix laboratory) is a language for technical computing, that integrates computation, visualisation, and programming. Problems and solutions are expressed in mathematical notation. MATLAB is typically used for math and computation, algorithm development, modelling, simulation and prototyping, data analysis, exploration, scientific and engineering graphics and application development including “Graphical User Interface” building.

MATLAB is an interactive system whose basic data element is an array. This allows us to solve many technical computing problems, especially those with matrix and vector formulations. It is easier than using languages like Fortran, or C. MATLAB is a comprehensive, programming, interactive environment. Using an API (Application Programming Interface), it is possible to write C and Fortran programs that interact with MATLAB, which means that we can use our own programs, written before, for existing applications. MATLAB also features a family of application-specific solutions called *toolboxes*. Toolboxes are collections of MATLAB functions, or M-files, that extend the MATLAB environment to solve particular classes of problems, like: signal processing, control systems, neural networks, fuzzy logic, simulation and others.

An additional component to MATLAB is *Simulink* that can be used to simulate non-linear dynamic systems. It is a graphical, mouse-driven application that allows us to model a system by drawing block diagrams on the screen, and manipulate them dynamically. It can work with linear, non-linear, continuous-time, discrete-time, multivariable, and multirate systems. *Blocksets* are add-ons to Simulink that provide additional libraries of blocks for specialised applications. *Real-time Workshop* is an application that allows us to generate C code from block diagrams and run it on a variety of real-time systems.

In university environments, MATLAB is the standard instructional tool for introductory and advanced courses in mathematics, engineering, and science.

### **Sequence and Reinforcement - from Analogue to Digital World**

The principle of Sequence and Reinforcement is used to educate students how transition from analogue to digital signals takes place. Once the signals are in digital form, Digital Signal Processing Techniques can be applied.

**Harmonic signals revision** Students in the class express their needs by requesting an appropriate level of revision to be done, to support each learning unit.

The next step, following basic math, is to study signals, and to explain to students, transition from analogue to discrete domain using the principle of Sequence and Reinforcement.

Let us consider the following signal:

$$x(t) = A \cos(\omega_0 t + \Phi) \quad (1)$$

The basic equation (1), where

$A$  represents a amplitude, in [unit] according to the physical quantity,  
 $\omega_0$  is radian frequency, in [rad/sec]  
 $t$  is the time, in [sec] and  
 $\Phi$  is phase shift, in [rad],

could be rewritten as following:

$$x(t) = A \cos(2\pi f_0 t + \Phi) \quad (2)$$

where  $f_0$  is cyclic frequency, in [Hz = 1/sec] and as known

$$\omega_0 = 2\pi f_0 \quad (3)$$

The next step would be to introduce sampling period  $T_s$ , and index  $n$ , as

$$x(t) = A \cos(2\pi f_0 n T_s + \Phi) \quad (4)$$

where  $T_s$  can also be represented as  $1/f_s$ , which finally gives

$$x(t) = A \cos((2\pi f_0 / f_s) n + \Phi) \quad (5)$$

Students put this all into MATLAB programs that they design so that they can apply what they are learning to a real situation.

In their programs, apart from implementing basic expression (5), students are encouraged to try different sampling rates, frequencies, durations, amplitudes and listen to the sound, as well as, look at generated plots. By doing so, they can research what happens with signals when over-sampling or under-sampling is implemented. They are also encouraged to generate the sum, or product of signals and analyse outcomes.

Expanding what Vella (1994) has stated regarding praxis as a process of doing – reflecting – deciding – changing – new doing, students are asked

- 1) What do you see and hear happening, and how does that relate to your practical skills and knowledge?
- 2) Why do you think it is happening, what do you think will happen, before you do some changes, and do the changes as much as possible. Analyse and do something else, many times.
- 3) When it happens, in our case under-sampling, for example, what problems does it cause (folding, harmonics...)?
- 4) What can we do about it to prevent, or to implement it?

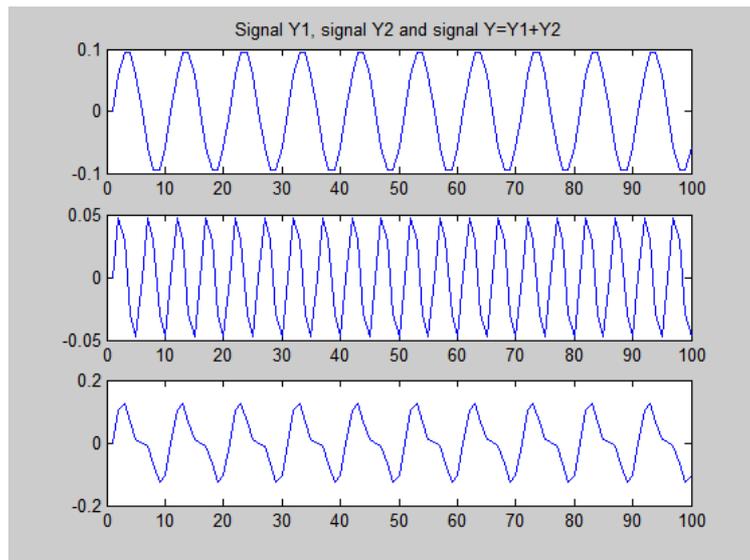
Students produce different programs and use them as tools to analyse the physics behind events. They have full freedom in their design, while following only syntax rules for MATLAB. At the same time they are encouraged to improve design by enforcing known good practices.

An example of the program is shown in Figure 1.

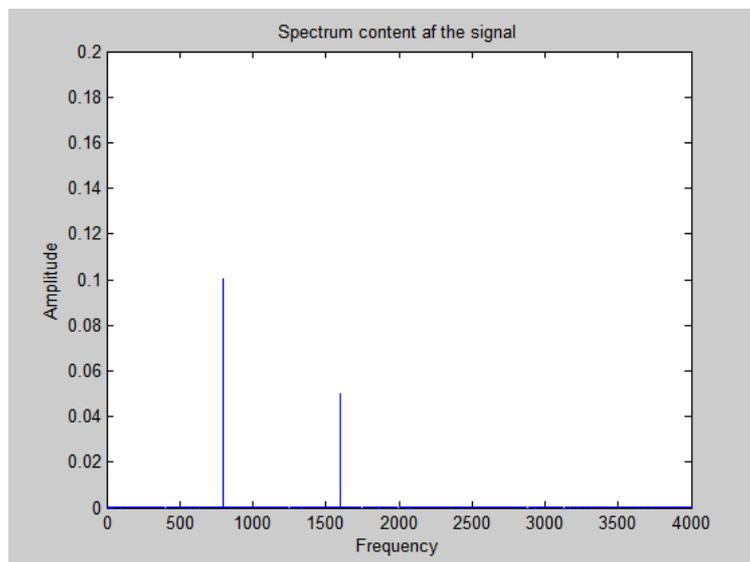
```
% Sampling rate study, spectrum study
Fs=8000;           % Sampling Rate
T=2;              % The length of the signal in seconds
F0=800;          % Frequency of the signal in Hertz
Ph=-pi/2;        % Phase of the signal in radians
A=0.1;           % Amplitude of the signal
n=0:(T*Fs)-1;    % Index vector, dimensionless
Y1=A*cos(2*pi*F0/Fs*n+Ph); % Generating a signal
Y2=A/2*cos(2*pi*2*F0/Fs*n+Ph); % Generating another signal
Y=Y1+Y2;
%
sound(Y, Fs)      % Playing the sound
figure(1)         % Preparing to plot signals
subplot(3,1,1)
plot(Y1(1:100))
title('Signal Y1, signal Y2 and signal Y=Y1+Y2')
subplot(3,1,2)
plot(Y2(1:100))
subplot(3,1,3)
plot(Y(1:100))
%
Yf=fft(Y);        % FFT of the signal
f=linspace(0, Fs*(1-1/(Fs*T)), length(n)); % Frequency axis values
figure(2)         % Preparing to plot spectrum
plot(f, abs(Yf)/length(n)*2) % Plots the magnitude
title('Spectrum content of the signal')
xlabel('Frequency')
ylabel('Amplitude')
axis([0 Fs/2 0 2*A])
```

**Figure 1: Example program that can be used to analyse sampling and signal spectrum**

In this simple example, the program generates two sinusoidal signals, adds them, plays the sound and plots all signals and the spectrum. Figure 2 presents two signals and their sum in time domain. Figure 3 shows the spectrum of the combined signal.



**Figure 2: Signal Y1, Signal Y2 and Signal Y as a sum of two sinusoids Y1 and Y2**



**Figure 3: Spectrum of the composite signal  $Y = Y1 + Y2$**

Students can easily test theory concepts using MATLAB instead of just doing mathematical exercises.

### **Practise and application - test for hearing screening**

#### **Screening for Hearing Loss**

According to Lynn S.A and Washington Sounds (1968), Humphrey, Herbst and Faurgi (1981), and Gallup (1980), there are two populations with hearing problems:

- 30 to 50 percent of the population over 65,
- Young children are usually not detected before the age of 3 to 7.

For children, early detection of hearing problems through screening is very important. Equipment like FONIX 6500-CX Hearing Aid Test System, are complete hearing analysers. FONIX provides test sequences for ANSI IEC and JIS standards.

Acoustic drive signal uses frequencies of 100Hz through 8000 Hz in 100 intervals.

Screening is a simple procedure and can be done by non-health professionals.

Having that in mind an assessment task was created. The idea is not to replace a Hearing Aid Test System, but to give students a chance to design an application. Their product can be used to discover possible problems and if so, issue a recommendation to a person to see a medical practitioner.

### **MATLAB test**

After learning how to generate a signal, by doing all of the above, students are ready to do a practical assessment task. The task is part of a bigger project that will follow later.

### **Digital signal processing project task on generating signals: Screening Test**

- 1) Design a MATLAB program as a tool to test hearing.
- 2) Produce a report to explain hearing map, and how to use your program.  
Put proper comments in the program, as well as your name and copyright statement. Your program should create and print a hearing map that should show: frequencies, in the range of 100Hz to 20KHz, in variable increments and, signal amplitudes needed, for a tested person, to hear the sound.

**Assessment Method:** Printed report, program and demonstration in the class.

**Due date:** A date

**Assessment Criteria:**

Program working, only.	PA
Program working, with appropriate comments.	CR
Program working, with appropriate comments and report (1000 words).	DI
Well designed program working, with appropriate comments, and report addressing issues of program design, as well as, issues of hearing (1000 – 1500 words).	HD

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### **Possible solution**

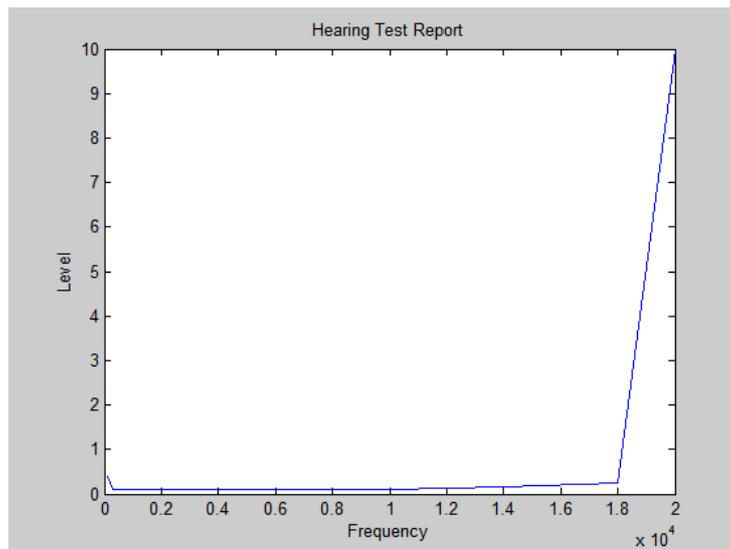
```
%Matlab screening for hearing loss test
%Tested frequencies are given in the vector fr
fr = [100 300 500 700 1000 1250 1500 1750 2000 2500 3000 4000 5000
7000 10000 14000 18000 20000];
Level(1:18)=0; %vector to hold needed sound level
fs = 40000; %sampling frequency
dur = 2; %duration of the signal is 2sec
tt = 0:(1/fs):dur;

%
Choice=1;index=1;
msgbox(' Welcome ');
pause(5)
while Choice == 1,
    i = 1;
    while i<19, %loop through frequencies given in the fr vector
        M=2;Amp=0.001; %Amp is tested intensity level
        while ((M == 2)&(Amp<=10))
            Amp = Amp*10; % increase sound intensity
            xx = Amp*sin(2*pi*fr(i)*tt);
            sound(xx,fs,16);
            M = menu ('Have you heard that?', 'Yes','No','Next');
            if M==3
                Amp=10
                break
            end
        end
    end
end
```

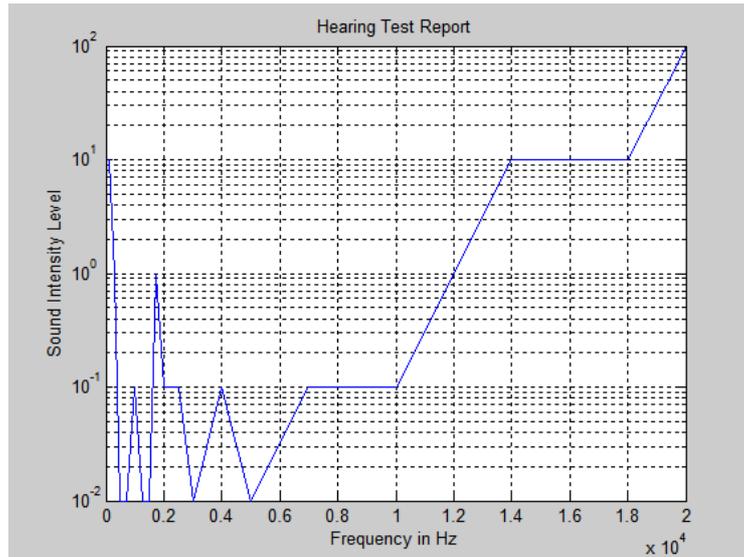
```
end
    Level(i) = Amp           % save the current level
    i = i + 1;
end
%
    figure(index); semilogy(fr,Level) %log scale(base 10) for Y
    grid on
    ylabel('Sound Intensity Level');
    xlabel('Frequency in Hz');
    title('Hearing Test Report');
    Choice=menu('Would you like to do the test again?','YES ', 'NO');
    index=index+1;
end
% Copyright © 2002 Dr Milan Simic
```

**Figure 4: A simple MATLAB Screening test program**

Figures 5, 6 and 7 present hearing maps produced by different programs, for different persons. Sound pressure level should be expressed in dB, but since different PC multimedia devices will produce different outputs, calibration should be done for more accurate results. It was not the intention in the first place. Instead of that, Y-axis represents just amplitude of the signal in a log scale, Figures 6 and 7. Students produce similar figures and show interest in hearing and seeing signals.

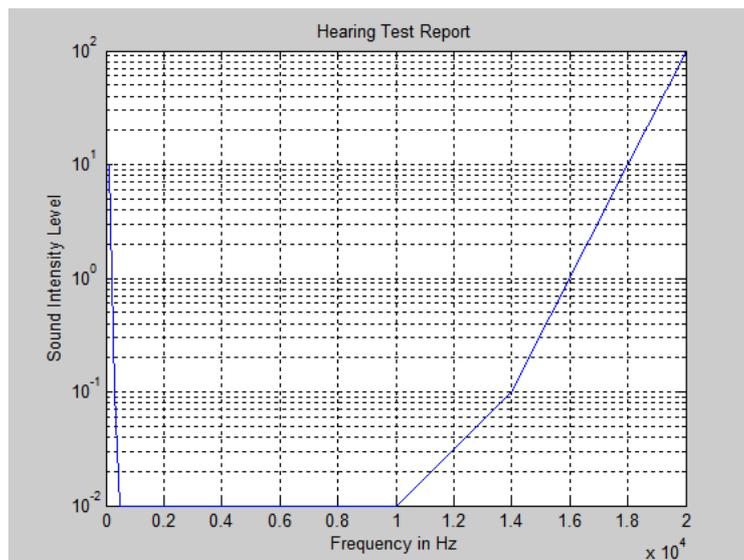


**Figure 5: Hearing test report 1, linear axis, increment of amplitude is 0.05**



**Figure 6: Hearing test report 2 showing some hearing problems**

With a slight modification of the given program, Figure 4, we can produce a printed report like one shown in the Figure 8. Intensity of the sound is expressed in dB and frequency is shown in log scale, as expected.



**Figure 7: Hearing test report 3, logarithmic scale for amplitude**

By asking students to work in teams and demonstrate their product, their ideas, feelings and actions are enforced. As team members, students have their responsibilities; they share ideas, teach and help each other. There is also competition among teams. The project is practical, so that immediacy is in place as well. The roles of the teacher and the student are clear and the role of each team member is usually clear, too. All students are fully engaged in designing and testing their applications. If students are exposed to too much theory and new concepts, they can easily lose interest in the learning process.

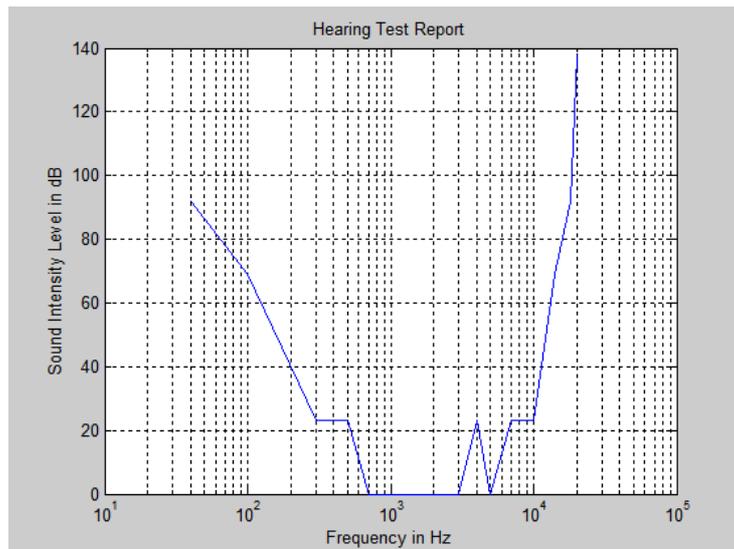


Figure 8: Hearing test report 4, logarithmic scale for amplitude and frequency

### MATLAB solution for Fletcher-Munson Equal Loudness Curves

In 1933, two researchers at Bell labs, Fletcher and Munson gathered information about how we perceive different frequencies at different amplitudes. They created “Equal Loudness Contours” or “Fletcher and Munson Curves”.

Those curves show as that there is a different threshold of hearing at different frequencies and the apparent levels of equal loudness at different frequencies.

The softest sound that we can hear is equivalent to a pressure variation of 20  $\mu\text{Pa}$ , at the frequency of 1kHz, Martin (2002). If we change the frequency we need higher amplitude of the signal.

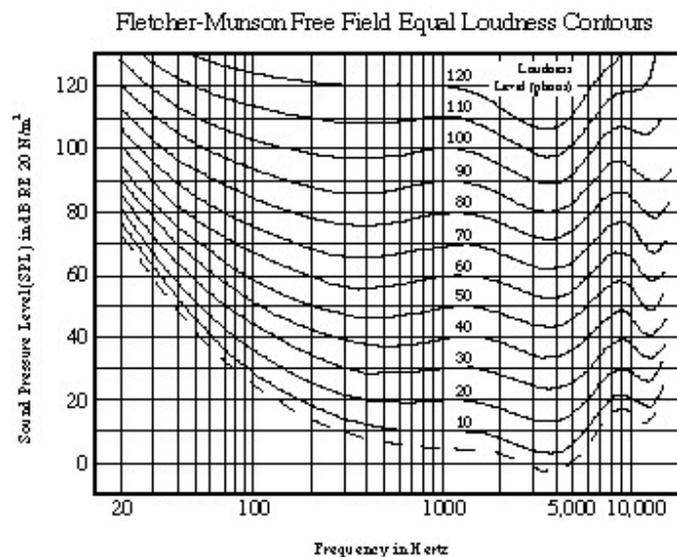


Figure 9: Fletcher-Munson equal loudness curve, from ACS

This forms the basis of another assessment task. Students have to write a program that can be used to draw Fletcher-Munson Equal Loudness Curves.

## Processing signals and Deconvolution

After being introduced to FIR Filters, z-Transform, IIR filters, and after learning more about spectrum analysis, students are ready for the final project. The project objective is to design a program that can be used as a hearing aid. Using the aid, a person with hearing problems, such as that given in Figure 6, should be able to hear normally for his/her age, as given in the Figure 7, and generally according to the Fletcher and Munson Curves shown in Figure 9.

The system that includes human ear and a hearing aid, is the LTI (Linear time-invariant) system, as given in equation (6). Input and output are sound signals. The Z-transform of the output,  $Y(z)$ , is equal to the Z-transform of the input,  $X(z)$ , multiplied by the system function of the LTI system,  $H(z)$ .

$$Y(z) = H(z) X(z) \quad (6)$$

The human ear is a filter, with the system function  $H_e(z)$ .

Hearing aid is another filter, with the system function  $H_a(z)$ .

We have to cascade two systems so that the new system, with the system function  $H(z)$ , as given in equation (7), will have system function equal to 1.

$$H(z) = H_e(z) H_a(z) = 1 \quad (7)$$

In other words, the second filter, which is our hearing aid, has to undo convolution produced by the first filter, the human ear. That process is called deconvolution, or inverse filtering.

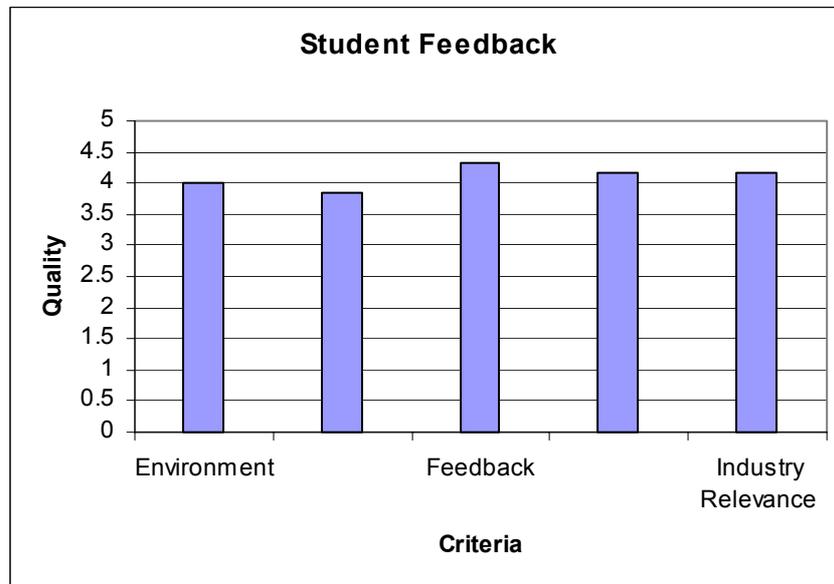
## Addressing principles and theories of adult learning

By following the approach of Sequence and Reinforcement, described in this paper, I hope to have made it easier for students to learn and apply DSP principles in engineering. By asking students to start designing useful applications, from the very beginning I am applying the principle of praxis. Students can easier understand complex principles by describing, discussing and analysing them. Through designing application, they learn how to implement skills and further enforce the knowledge. By interaction during the course and asking for their evaluation of the course, after, I practise respect for learners as subjects of their own learning. Most of the projects are designed and given to students, so that they have to do the job as team members. As team members, students share responsibility for their own learning. They help each other. Through communication they share skills and achievements. All of these support learning and speed up the learning process. Finally, industry projects are team projects. By being exposed to the teamwork, students are better prepared for the future workplace projects.

## Student Feedback

Every year, during and after the course surveys on student satisfaction are conducted. Although the course is hard and heavily loaded with maths, students enjoy it and achieve a lot. Students are asked to comment on aspects of the course such as: are they happy with learning environment, does teacher provide clear instructions and feedback, how diverse are assessment methodologies and do they reflect learning outcomes, is material being presented effectively, and is the material being thought relevant for industry. Grading for each question goes from 1 to 5, 5 being the best. Students are encouraged to add comments.

The survey results given in the Figure 10, show that students are happy with the environment, as the average satisfaction number was 4. They are happy with clear instructions and the feedback, which is very important having in mind that, we are speaking maths language in this course. The survey is just a part of the feedback that is happening in the class, and after the class: consultation time, email... According to the survey, the course is on industry track, with mark 4.16. From the surveys one can always get some ideas how to improve teaching for the future.



**Figure 10: Student Survey Results**

Some of the typical comments from the survey are given below.

Student comments about project-based teaching were:

“I think that hands-on approach supports and reinforced theory. It is a good way to learn”.

“Very good as you get to practice theory and have it in an operating program at the end.”

Comments about real understanding the physical concepts underlying DSP by interacting with MATLAB were:

“Yes as you could visually see and hear the end results.”

“MATLAB is a good tool for manipulating signals. And demonstrate concepts well.”

Comments about Sequence and Reinforcement:

“Yes the more reinforcement given the easier the theory becomes”

A comment about how the given teaching approach is performed:

“Transferring theory into practice worked well.”

Suggestions to improve teaching:

“More step by step, from basic to difficult.”

The subject material of DSP changes and that influences teaching accordingly. Traditionally DSP was taught mainly mathematically and this made it very difficult for students to grasp the physics behind the concepts. This later evolved into teaching DSP via a combination of, maths and FORTRAN, ASM language, C++, C. Currently MATLAB is used to support teaching and industry use it to develop real life applications. This paper has presented a methodological approach for introducing MATLAB into DSP teaching.

## References

### Book:

- Analog Devices (1990). *Digital Signal Processing Applications*. Englewood Cliffs, New Jersey: Prentice Hall.
- Burns, R. (1995). *The Adult Learner at Work*. Business & Professional Publishing
- Childers, D.G. (2000). *Speech Processing and Synthesis Toolboxes*. New York: John Wiley & Sons.
- Dolores M.E. (1997). *Engineering Problem Solving with MATLAB*. Upper Saddle River, New Jersey: Prentice Hall.
- James, H.M., (1999) Ronald, W.S., & Mark, A.Y. *DSP First, A Multimedia Approach*. NJ: Prentice Hall.
- Richard, J.H. (1990). *Digital Signal Processing in VLSI*. Englewood Cliffs, New Jersey: Prentice Hall.
- Vella, J.(1994). *Learning to listen, learning to teach: The power of dialogue in educating adults*. San Francisco: Jossey Bass
- Thomas, J.C. (2000). *Digital Signal Processing*. New York: John Wiley & Sons.

### Journal article:

- Washington Sounds: Senate Aging Committee launches investigation of hearing aids with two day of hearing before Consumer Interests Subcommittee. (1968, July 22). Washington, D.C.: House Publication
- Humphrey, C., Herbst, K., & Faurgi, S. (1981). Some characteristics of the hearing-impaired elderly who do not present themselves for rehabilitation. *British Journal of Audiology*, 15, 25-30.
- A Survey Concerning Problems and Hearing aids in the United States (1980). Princeton, NJ: The Gallup Organization, Inc.

### Online source:

- <http://www.nl.edu/ace/Resources/Knowles.html>
- <http://www.infed.org/thinkers/et-knowl.htm>
- Martin, G. (2002). *Introduction to Psychoacoustics*. From <http://www.tonmeister.ca/main/textbook/psychoacoustics/02.html>
- <http://www.frye.com/products/analyzers/6500ts.html>
- <http://www.allchurchsound.com/ACS/edart/fmelc.html>
- Fletcher-Munson Equal Loudness Curves, from <http://www.allchurchsound.com/ACS/edart/fmelc.html>