Journal of Engineering Education Transformations,
Volume 30, No. 4, January 2017, ISSN 2349-2473, eISSN 2394-1707

Improving Teaching-Learning Process for Digital Signal Processing

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Abstract: Digital Signal Processing (DSP) is one of the core courses in Electronics and Telecommunication Engineering which involves lot of mathematics. This has led to poor performance of students in the subject. We made an attempt to improve the performance by adopting two main techniques in the teaching-learning process for the subject. First, to overcome the computational mistakes, we gave individual assignments with the same task but different data to every student. We used Matlab to get the answers for all these different data. Second, to make slow learners interact with their better knowledge peers, we made groups of 8 students each and every group was asked to solve 7 questions using Matlab. Along with this graphical user interface was used to explain the applications of DSP. Analysis of results for two consecutive academic years, indicates improvements in the performance of students after adopting these techniques.

Keywords: Individual Assignment, Computing Assignment, DFT, FIR Filters, Graphical User Interface

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1. Introduction

Digital signal processing (DSP) has been tagged as one of the difficult subjects by many students because of its mathematical nature and the difficulty in understanding the concepts. Any course for that matter will be appreciated by the students if concepts and applications are explained clearly. However, signal processing courses such as signals and systems and digital signal processing are being approached as mathematical course by many faculty members. This has also lead to poor results in such courses. Though some students manage to get average grades in these courses, they find it difficult to answer questions which are based on concepts and applications. The problem exists even in answering the oral questions in practical examinations. Rao and Prema [1] presented a few basic concepts in signal processing with simulations. Their focus was on effective way of introducing time and frequency domain differences in the digital signal processing course.

At Siddaganga Institute of Technology, an Autonomous Institute under Vishveswaraiah Technological University, DSP is a core course for Electronics and Communication Engineering and Telecommunication Engineering in V Semester. Successful completion of this course demands knowledge of Mathematics along with clear understanding of concepts. Concept and application oriented questions are very rarely answered correctly by students. Also, students fail in answering the questions which involve calculations. But, these are the two main types of questions asked for an analytical subject like DSP. To improve the performance of students in this subject, we tried modifications in teaching-learning process for the subject. Different types of assignments were given to students and different ways of teaching was adopted based on the topic to improve the students' performance.

The syllabus comprises of representation of signals and systems in time and frequency domain by using Discrete Fourier Transform (DFT), efficient computation of DFT, design and realization of finite and infinite impulse response filters and application of DSP. First and important aim was to improve the % pass in the subject compared to the previous years. In filter design, finite impulse response (FIR) filters and infinite impulse response (IIR) filters have their own advantages based on the application [2]. In fact, FIR filters are the only choice when the application demands linear phase systems. So, second aim was to introduce some changes in teaching FIR filters. In the process of teaching the subject in mathematical way, teachers find hardly any time to teach applications which is actually very important. Third aim was to improve the effectiveness of teaching applications of DSP.

In the next section, we discuss the method adopted to improve the % of pass. Third and fourth sections explain the steps taken to improve teaching FIR filter design and applications of DSP respectively. Further, to clear the concepts of DSP and to get a practical feel, computing assignments were given whose details are presented in Section 5. Effect of these modifications on the students' performance in the end semester examination is discussed in Section 6 and Section 7 presents the conclusion.

2. Individual Assignment With Same Task But Different Data

Beginning of the semester students will be having relatively more free time compared to other days of the semester. So, our focus was to make students perform well in the first two modules. Modules 1 and 2 comprise of sampling theorem, DFT and its efficient computation using algorithms. Also, the intention was to avoid copying of assignments. To achieve this, assignment questions were prepared so that the questions were common, but every student has different data as part of the numerical. A copy of the assignment is shown in Figure 1. As the same questions were given with different data, even if a student is not comfortable in solving the question, he/she could discuss the same with friends, but solve the problem with his/her data. As seen at the bottom of the figure, every student was asked to assume two sequences. However, it is a burden for a teacher to evaluate all the assignments carefully. So, a Matlab code was written for the same which accepted the two sequences from the user. So, different data was given during every run of the code and answers were verified quickly. This improved the level of students' confidence to answer questions related to first two modules.

Siddaganga Institute of Technology, Tumkur

Department of Electronics and Communication Engineering

Academic Year: 2015-2016

Course: Digital Signal Processing

Assignment-1

Last date for submission-7th September 2015

Please read the note at the end of this page before solving the assignments

1. Consider two real sequences $x_1(n)$ and $x_2(n)$ of length 3 and 2 respectively.
2. Compute 4-point DFTs of $x_1(n)$ and $x_2(n)$.
3. Compute 8-point DFTs of $x_1(n)$ and $x_2(n)$. Also, plot the magnitude and phase for both.
4. Write the DFT of $x_1((-n))_4$ and $x_2((n-3))_4$ and IDFT $X_2((k+1))_4$
5. Prove Parseval's theorem for $x_1(n)$.
6. Compute linear convolution of $x_1(n)$ and $x_2(n)$ in time and frequency domain.
7. Compute 3-point circular convolution of $x_1(n)$ and $x_2(n)$ in time and frequency domain.
8. Compute 4-point DFTs of $x_1(n)$ and $x_2(n)$ using a single 4-point DFT.
9. Given the impulse response of the filter is $h(n)=x_1(n)$, and the input $x(n)=\{1\ 4\ -1\ 8\ 0\ 0\ 2\ 7\ -2\ -1\ -3\ 8\ 2\ 5\ 2\ 4\ 0\ 0\ 1\}$, find using the output using overlap and save method. Use 5-point circular convolution.
10. Given the impulse response of the filter is $h(n)=x_1(n)$, and the input $x(n)=\{1\ 4\ -1\ 8\ 0\ 0\ 2\ 7\ -2\ -1\ -3\ 8\ 2\ 5\ 2\ 4\ 0\ 0\ 1\}$, find using the output using overlap and add method. Use 6-point circular convolution.

Note: Assume your own data for $x_1(n)$ and $x_2(n)$ and proceed. Don't copy the data from others. All your assignments will be checked and you will be asked to correct in case if there are any mistakes.

Figure 1: Copy of one such individual assignment

3. Interactive Teaching of Fir Filters

Design and realization of filters is another important topic in the DSP course. We made an attempt to improve the teaching-learning process of filters. Windowing method of designing FIR filters, Hilbert transformer and differentiator especially effect of varying size and type of the window was demonstrated using graphical user interface (GUI).

A. Effect of varying the type of the window

Different windows used in digital signal processing are: rectangular, triangular, Hanning, Hamming and Blackmann windows. First, we discuss the variation in type of window by taking examples of differentiator and Hilbert transformer. Equations (1) and (2) give frequency responses of ideal digital differentiator and Hilbert transformer respectively.

$$H\left(e^{j\omega}\right)=\begin{cases} j\omega, & 0<\omega<\pi \\ -j\omega, & -\pi<\omega<0 \end{cases} \tag{1}$$

$$H\left(e^{j\omega}\right)=\begin{cases} -je^{(-j\omega\alpha)}, & 0<\omega<\pi \\ -je^{(-j\omega\alpha)}, & -\pi<\omega<0 \end{cases} \tag{2}$$

Where α = (N-1)/2 and N is the length (number of samples) of the window [3]. Magnitude responses of ideal digital differentiator and Hilbert transformer are shown in Figures 2(a) and 2(b) respectively.

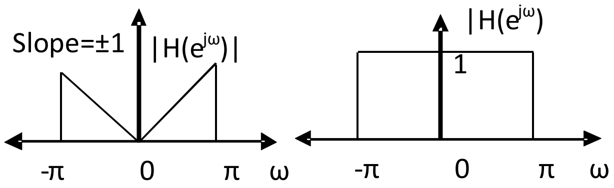


Figure 2: Magnitude responses of digital ideal (a) differentiator and (b) Hilbert transformer

Impulse responses of these systems are of infinite duration. Truncation of the infinite length impulse response using window function results in convolution of frequency response of the ideal system with the frequency response of the window function. Frequency domain representation of the rectangular window of length N is the sinc function with the main lobe width 4π/N rad. and side lobe attenuation (difference between the main lobe peak and first side lobe peak) 13 dB. This results in introduction of ripples and transition band in the frequency response of an ideal system. Side lobe attenuation of the window decides the ripple size while width of the main lobe decides the transition width. Therefore, main lobe width should be as small as possible and side lobe attenuation should be as large as possible. Main lobe width and side lobe attenuation of different windows is given in Table 1.

Table 1: Main lobe width and side lobe attenuation Of different windows

Window	Main lobe width	Side lobe attenuation
Rectangular	4π/N	13
Triangular	8π/N	27
Hanning	8π/N	32
Hamming	8π/N	43
Blackman	12π/N	58

As we move from rectangular to Blackman window, side lobe attenuation increases but main lobe width decreases. Therefore there is a trade-off between main lobe width and side lobe attenuation. Top pane of Figure 3 shows the GUI developed to understand the effect of changing the type of the window. The bottom pane of Figure 3 shows the effect of changing the window on the magnitude response of differentiator and Hilbert transformer. By selecting different windows the user will be able to observe the decrease in the ripple size and increase in the transition width of differentiator and Hilbert transformer as one move from rectangular window to other windows.

A. Effect of changing the length of the window

As time and frequency are inversely related, increase in the length of the window results in decrease in the temporal resolution and increase in the spectral resolution. Temporal resolution means the ability to resolve two nearby time instants and frequency resolution is the ability to resolve two closely spaced frequency components. The vowel 'ee' as in the word 'been' is recorded and shown in the GUI in Figure 4. Hamming window is used to truncate the vowel. Increase in the length of the vowel results in separation of closely spaced frequency components.

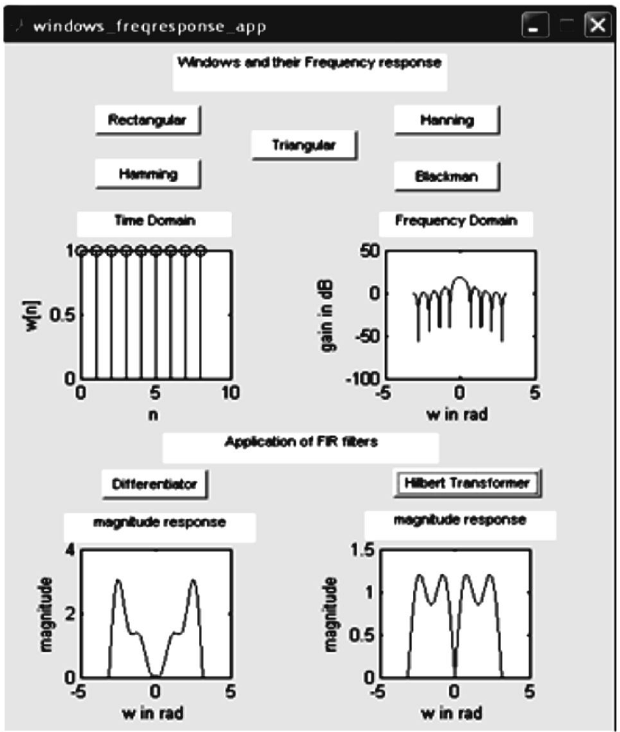


Figure 3: GUI developed for observing the effect of changing the length of the window

Top pane of Figure 4 shows the spectrum of the vowel when it is truncated using 5 ms window where many frequency components are merged. Bottom pane of

Figure 4 shows the spectrum of the vowel when it is truncated using 30 ms window where frequency components are separated.

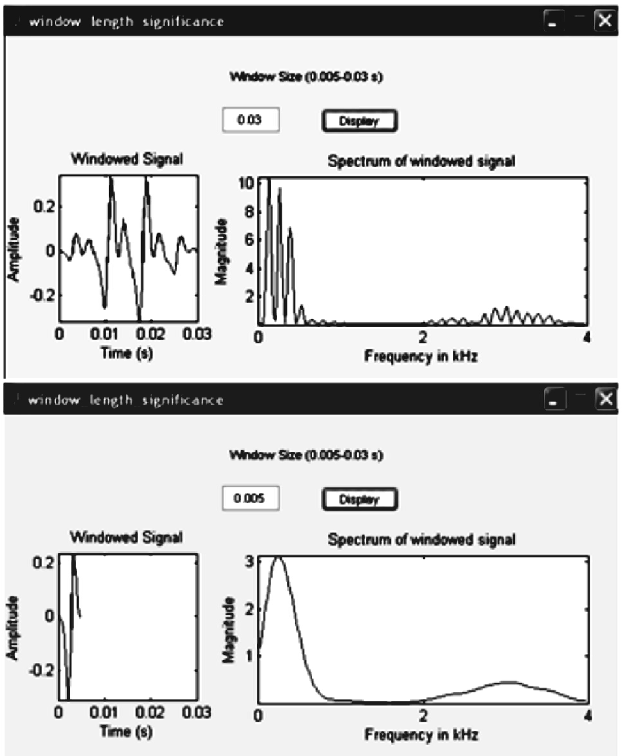


Figure 4: Gui Developed For Observing The Effect Of Changing The Type Of Window

4. Teaching Applications of DSP

The DSP techniques are increasingly replacing conventional analog signal processing methods all most all the fields of Engineering. Module 5 of our syllabus comprises of applications such as spectral analysis using DFT, DTMF generation and detection and Musical sound processing. GUIs were developed and used during teaching-learning process of these applications for the better understanding of the concepts.

A. DTMF generation and detection

Push button telephones have the 4x4 structure keypad. Each row and column corresponds to a particular frequency. When a key is pressed in push button telephones, combination of the two tones corresponding to the row and column of the key pressed is generated which is named as dual tone multi frequency (DTMF) signal. Frequencies corresponding to rows are 697 Hz, 770 Hz, 862 Hz and 941 Hz. Frequencies corresponding to columns are 1209 Hz, 1336 Hz, 1477 Hz and 1633 Hz. The sampling rate used in telephone signals is 8 kHz. This DTMF signal is decoded at the exchange to identify the key pressed. Goertzel algorithm is used to decode the DTMF signal [4].

Goertzel's algorithm is used to compute the strength of a particular frequency component in a given signal. Using Goertzel's algorithm, the strengths of 8 different frequency

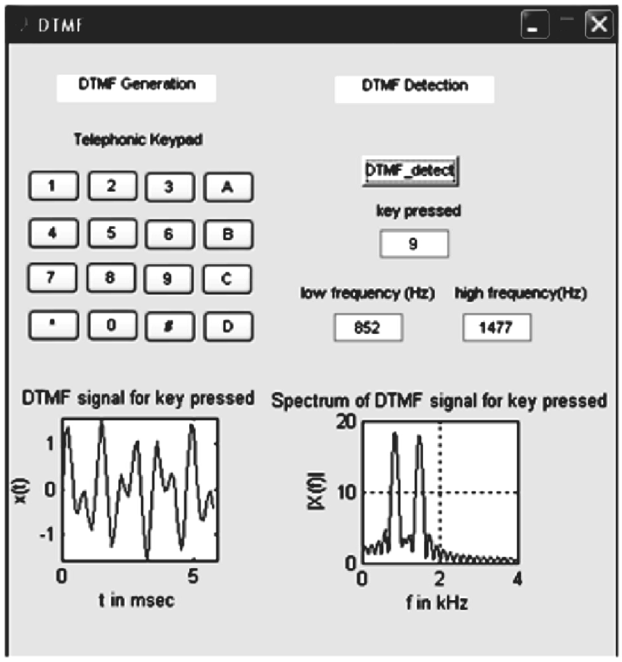


Figure 5: GUI developed to demonstrate the generation and detection of DTMF signals

components mentioned earlier in the received signal are computed. The two frequencies corresponding to the relatively high strength indicate the key pressed. The GUI developed for DTMF generation and detection is shown in Figure 5.

Left pane of the Figure 5 shows DTMF generation. Time and frequency domain representations of the DTMF generated can be seen in the figure. Right pane of Figure 5 shows the DTMF detection. Once the user presses the push button named "DTMF_detect", decoding is performed by using Goertzel's algorithm and key pressed is displayed along with the values of the two frequencies.

A. Generating echo effect and Reverberation effects

When a sound wave travelling in air hits a surface, part of it gets absorbed by the surface and the rest of it gets reflected. Human brain keeps a sound in memory for about 0.1 s. If the reflected sound reaches the ears within 0.1 s., it gets added to the original sound to form a prolonged sound wave. This results in reverberation. If the reflected sound reaches the ears after 0.1 s, the sound is known as echo. As the speed of the sound in air is 343 m/s, an echo is heard if the distance between the source of sound and the surface is greater than 17 m, else it results in reverberation. In this work, we have simulated echo and reverberation effects using Matlab GUI. Echo is generated by adding the signal to its delayed and attenuated version. If x[n] is the original sound, single echo y[n] is generated by using Equation (3) where, α is the attenuation factor less than 1 and R is delay in samples.

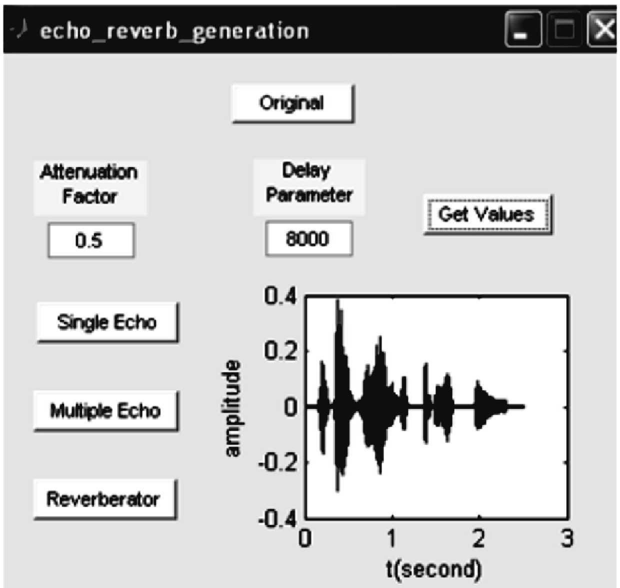


Figure 6: GUI developed to demonstrate the generation of echo and reverberation effects

$$y[n] = x[n] + \alpha x[n - R] \tag{3}$$

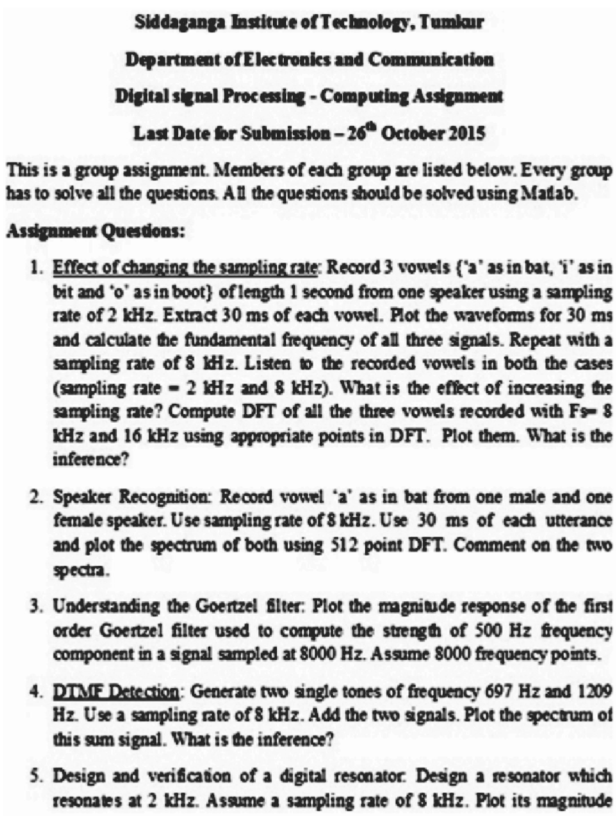
Similarly multiple echos are generated by adding original signal to different delayed versions with different attenuation factors. As the delay increases, attenuation factor increases as seen from Equation (4) which is used to generate N-1 echoes.Similarly multiple echos are generated by adding original signal to different delayed versions with different attenuation factors. As the delay increases, attenuation factor increases as seen from Equation (4) which is used to generate N-1 echoes.

$$y[n] = x[n] + \alpha x[n - R] + \alpha^2 x[n - 2R] + ----- + \alpha^{(N-1)} x[n - (N-1)R] \tag{4}$$

If number of delayed versions with different attenuation factors are added where delay is less than 0.1 s, the resulting effect is reverberation [4] which is similar to the effect that is felt in the concert hall. Figure 6 shows the GUI for generating single, multiple echo and reverberator. User can vary the values of α and R and listen to the differences in single and multiple echoes generated. To generate the reverberation effects the parameters are fixed.

5. Computing Assignments To Increase The Depth Of Understanding

To make students familiar with the concepts, computing assignments were given. Students were grouped such that there were 8 members in each group. The groups were formed such that every group had both slow and fast learners. The intention was to make slow learners interact with others. A part of the assignment page is shown in Figure 7. Seven



Siddaganga Institute of Technology, Tumkur

Department of Electronics and Communication

Digital signal Processing - Computing Assignment

Last Date for Submission – 26th October 2015

This is a group assignment. Members of each group are listed below. Every group has to solve all the questions. All the questions should be solved using Matlab.

Assignment Questions:

1. Effect of changing the sampling rate: Record 3 vowels {'a' as in bat, 'i' as in bit and 'o' as in boot} of length 1 second from one speaker using a sampling rate of 2 kHz. Extract 30 ms of each vowel. Plot the waveforms for 30 ms and calculate the fundamental frequency of all three signals. Repeat with a sampling rate of 8 kHz. Listen to the recorded vowels in both the cases (sampling rate = 2 kHz and 8 kHz). What is the effect of increasing the sampling rate? Compute DFT of all the three vowels recorded with Fs= 8 kHz and 16 kHz using appropriate points in DFT. Plot them. What is the inference?
2. Speaker Recognition: Record vowel 'a' as in bat from one male and one female speaker. Use sampling rate of 8 kHz. Use 30 ms of each utterance and plot the spectrum of both using 512 point DFT. Comment on the two spectra.
3. Understanding the Goertzel filter: Plot the magnitude response of the first order Goertzel filter used to compute the strength of 500 Hz frequency component in a signal sampled at 8000 Hz. Assume 8000 frequency points.
4. DTMF Detection: Generate two single tones of frequency 697 Hz and 1209 Hz. Use a sampling rate of 8 kHz. Add the two signals. Plot the spectrum of this sum signal. What is the inference?
5. Design and verification of a digital resonator: Design a resonator which resonates at 2 kHz. Assume a sampling rate of 8 kHz. Plot its magnitude

Figure 7: A part of the computing assignment page

questions were given and each group was asked to solve all the questions but present one question picked randomly.

6. Assessment Of Students' Performance With The Modifications Introduced

To learn the effect of these modifications introduced in the teaching-learning process, we analyzed the students' performance in the end semester exam. End semester exam will be conducted for maximum of 100 marks where part-A will be 1 or 2 mark questions for 20 marks overall and part B will be having 5 questions one from each of the 5 modules. Students have to answer 4 out of 5 questions in part-B and while part-A is compulsory which comprises questions from all the modules. Table 2 shows the result analysis of two consecutive academic years (AY) 2014-15 and 2015-16. The techniques mentioned in the paper were implemented for the AY 2015-16. Increase in the % pass can be noticed from Table 2. Also, because of the additional efforts put in teaching design of FIR filters, Hilbert transformer and differentiator, one can see the improvement in the students' performance in the answering the questions related to this. Also, use of GUI to demonstrate the applications and computing assignments have increased the number of students who attempt the questions related to applications has increased in the AY 2015-16.

We also conducted t-test and p-test to validate the results (marks scored by the students for the questions related to FIR filter module) statistically. t-value is calculated by using Equation (5).

$$t-value = \frac{\overline{x_1} - \overline{x_2}}{\sqrt{\left(\frac{(N_1-1)s_1^2 + (N_2-1)s_2^2}{df}\right)\left(\frac{1}{N_1}+\frac{1}{N_2}\right)}} \quad (5)$$

where, $\overline{x_1}$ and $\overline{x_2}$ are average marks, $s1^2$ and $s2^2$ are the variances of marks scored by N_1=60 and N_2=53 students in the AY 2014-15 and 2015-16 respectively and degree of freedom, df=N_1+N_2-2. p-value is calculated using t-value and df. We found that t-value = 6.1394 and p-value <0.00001 at the significant level of 0.05. This shows that % of marks obtained by the students for the questions related to FIR filter module in the AY 2015-16 is significantly greater than that obtained in the AY 2014-15.

Table 2: Result Analysis Of Two Consecutive Academic Years

Sl. No.	Academic Year	No. of Students Appeared for the Exam	% of Pass	% of Marks obtained in the module related to FIR Filter	No. of students who have attended module related to applications of DSP
1	2014-15	**60**	**88.17**	**49.32**	07
2	2015-16	**53**	**98.11**	**70.74**	22

7. Conclusion

This work makes an attempt to improve the students' performance in the subject 'Digital Signal Processing” by introducing two techniques. Individual assignment with same task but different data was given to overcome the mistakes in calculations. Also, computing assignment was given in groups which involved coding and also resulted in improved clarity in the concepts. GUIs were shared with the students and made the students to run the programs any number of times with the different data parameters while studying the DSP applications for better understanding. Analyzing the results over two consecutive years, we conclude that for the course DSP, it is important for students to practice problem solving as well as use simulations to gain clarity in the concepts.

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