LEARNING SIGNAL PROCESSING CONCEPTS AND PSYCHOACOUSTICS IN THE QUICKSIG DSP ENVIRONMENT

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ABSTRACT

In this paper a Computer-Based Education (CBE) system is described that is built on top of the QuickSig DSP programming environment. As two CBE applications we discuss the implementation and use of "Introduction to Signal Processing" and "Fundamentals of Psychoacoustics" that exploit modern multi and hypermedia features on new computers. The signal processing that is applied, the sound I/O, the graphical user interface, and the CBE navigation system are presented.

1. BACKGROUND AND MOTIVATION

Computer Based Education (CBE) means (in a broad sense) using computers in education for all kinds of purposes*). This includes both the learning process itself and the educational administration. Typically, CBE applies hypermedia tools to accomplish these tasks.

Even though CBE can be applied in almost any education, it is not always meaningful. Using CBE must be in some way more useful than ordinary teaching methods. That is why the starting point for every CBE-project is to think if CBE can give something "extra" or perhaps a "better way" in the means of education compared to ordinary methods. However, with Signal Processing and especially with Digital Signal Processing, CBE can be very fruitful.

Another important aspect is the quality of CBE, which includes both the quality of the contents and the quality of the application, namely its user interface. Getting high quality requires careful planning. Luckyly, a number of design methods for CBE exist. These methods help defining the contents and the logical structure as well as the interface itself. The key idea is that the emphasis of a CBE-application should allways be on the subject, not on the program itself! Both the structure and the interface should support that principle. In our project we have used the Lifländer design method [1], based on the studies of learning by Engeström [2], and also some ideas of the Lessonware [3].

*) Another commonly used term is Computer Aided Learning (CAL), which means the same as CBE

1.1 Hypermedia and DSP

Advanced multi and hypermedia features of new computers are particularly well suited to support the education of signal processing concepts. Efficient real-time or non-real-time processing of audio and speech signals is already made possible without any or with minimal extra hardware. CD-quality (16-bit) sound input and output are standard features in many multimedia computers. High-quality graphical interface, including video and animation, adds to the means of user interfacing.

Exploitation of these features in learning the basic concepts and applications of signal processing is the topic of this paper. As a part of a more general CBE-project at the Helsinki University of Technology, the Acoustics Laboratory has been developing computerized education for fundamentals of signal processing and psychoacoustics.

Signal processing applications demand a lot of computer capacity (and often some special hardware). Many traditional DSP tools have therefore been designed to be efficient but they are not necessarily very flexible. On the other hand, CBE-applications demand great flexibility of the environment and good tools for interface design. What is needed is a flexible yet efficient environment for both signal processing and CBE. QuickSig [4] fulfills these requirements.

2. THE QUICKSIG ENVIRONMENT

QuickSig is an object-oriented signal processing and algorithm development environment [4]. It provides signal processing tools for many application domains but it is also easily extendible to meet requirements of CBE since it runs on top of the Lisp language and CLOS (Common Lisp Object System) [5].

Signals and related concepts are represented as objects and operations are typically implemented by method functions. A wide variety of DSP functionality, such as filtering, transforms (FFT), etc., are built-in features in QuickSig. An optional real-time DSP extension QuickC30 [6], based on the use of TMS320C30 processor, is also available. The full QuickSig software environment runs on Apple Macintosh computers. CD-quality (16-bit) stereo sound input and output are available on some models.

The QuickSig graphical user interface includes graphical presentations for signals and related objects as well as direct manipulation of them by mouse and keyboard actions. The interface has been extended to support CBE applications. A display window may easily be composed of signal presentations, various controls (such as buttons), texts and figures by using a layout script syntax. A collection of basic hypermedia functions like HotWords, links to other documents and even applications, as well as on-line navigation and help have been added thus making QuickSig also a good authoring environment. A User Statistics Database for administrative purposes is also integrated in the system.

3. CBE-APPLICATION "INTRODUCTION TO SIGNAL PROCESSING"

"Introduction to Signal Processing" is a CBE-application that is designed to give 2nd-3rd year university students a practical and yet general enough view on signal processing especially from the point of view of audio and speech signals. Later on we intend this to be tailored to an early (first year) introduction. The practical approach taken here has proven to be useful in undergraduate education, see for example [7]. "Heavy" mathematics is included, but only as extra material. In this tutorial step it is more important to explain concepts with examples and exercises. The "inside theory" of signal processing is left for more specific courses.

The key to this CBE-application is the model illustrated in Fig. 1. It serves as a base for the application, being simultaneously a system model of (audio) DSP and the "Main Menu" for the application. Each block is a Hot Area that, when clicked, leads to an independent session of that subtopic.

The subtopics covered by the application are:

- Signals, their properties and representation (graphical presentations, numerical and function representations).

- A/D and D/A-conversion: sampling, quantization and coding.

- Signal Processing: general principles (linear/nonlinear, analog/digital etc).

- Spectrum Analysis: spectrum and its properties, calculating spectrum and windowing.

- Filters: impulse response and transfer function, convolution and digital filters.

Each subtopic is presented by one or several pages of concept explanation, examples, and exercises. Students can study these subtopics in any order they wish and they can also use on-line navigation map or searchable subject index for determining individual study paths.

The main difference between this application and other computer-assisted laboratory courses like [8] and [9] is that all the information, examples, and exercises have been integrated into one interactive system making it easy and convenient to use.
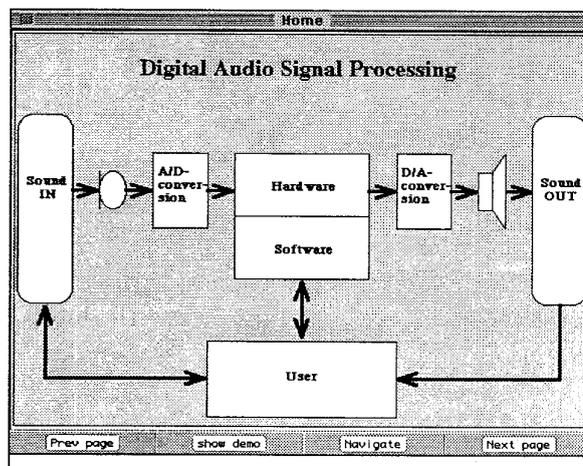

Fig. 1. The system model for CBE-application "Introduction to Signal Processing", including the "Main Menu" view and navigation buttons.
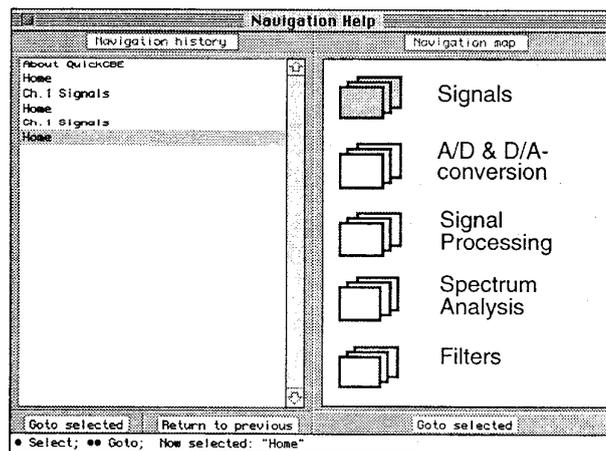

Fig. 2. Navigation Help window. Left panel is for showing the navigation history and right panel is a navigation map (top level items of the application in this case).

The introduction to and the online documentation of the QuickSig system itself is also implemented in a similar CBE-framework as the application.

4. PSYCHOACOUSTICS EXPLORATIONS

"Fundamentals of Psychoacoustics" is a very challenging and rewarding CBE-topic. Psychoacoustics (or auditory psychophysics) is a discipline where the extremely complex behavior of the human auditory system, abstracted from the physiological details, is described [10]. In principle the topic represents relatively exact knowledge but, due to inherent complexity and nonlinearity of hearing, it is impossible to
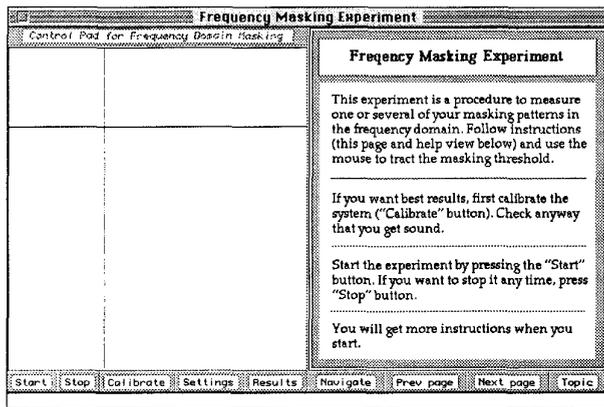
Fig. 3. Window for measuring the frequency domain masking effect. Left pane has the level and frequency cursors (intentionally minimal hints and no history curve). Right hand pane gives instructions to the user.

Fig. 4. Result of frequency domain masking effect measurement (solid line) and theoretical curve (dotted line).

cover simply by analytical mathematics. Just reading textbooks leaves the concepts too abstract and the traditional experiments are cumbersome and time consuming. Using computers and multimedia opens entirely new approaches to learning psychoacoustics.

The basic idea is to include, in addition to the fundamental theory, a rich set of easy-to-do demonstrations and exercises. Topics such as perception of pitch, loudness, timbre, and duration, concepts such as masking, critical band, loudness formation, roughness, consonance and dissonance, distortion and combination tones, concepts related to spatial hearing, etc., open an endless area of interesting experiments that support learning the theory of psychoacoustics.

In our system the student is first given a compact tutorial of the selected concept by text and figures. The next step is to follow a predefined procedure where the computer collects data on the perception of the parameter or feature under study. Then the result is shown as a curve to compare to a "standard pattern" found in textbooks. The experiment may be repeated now when the student already has a conception on the idea of the procedure.

Finally, if the student is willing to explore deeper into the given concept, he or she may generate variations of the experiment to view different aspects of the phenomenon, The concept under study is learnt in an illustrative and concrete way. Whenever available, a mathematical model or the behavior of a DSP-based auditory model may be compared to the data from the experiment.

Currently our psychoacoustics package includes a set of the most fundamental concepts. For example, the masking effect [10] is first demonstrated in the frequency domain. By using mouse the student tracks the detection threshold of a test tone when a narrow-band noise of constant level is used
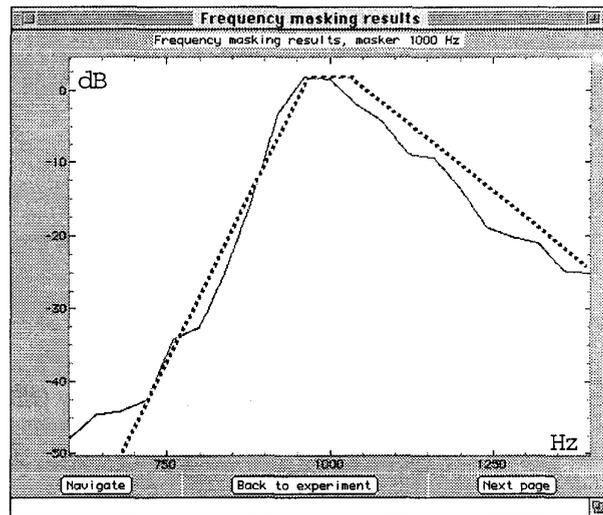
as a masker. A control pad (left-hand side in the window of Fig. 3) is used for setting the test tone level (horizontal cursor), in repetition of a pair of masker vs. masker + test tone, until the detection threshold is found. Instructions to proceed are given in the right-hand side pane of the window. By a mouse click, the next test tone frequency is generated. This is repeated until the frequency range of interest is covered. Finally, the measured masking level curve will be shown along with a "standard pattern" (Fig. 4).

During the experiment a pair of pad cursors give hints about the relative level setting and the progress of test tone frequency but no curve is given until the whole experiment is finished. Next, a similar experiment is carried out in the time domain, whereby moving the time position of the test tone the concepts of pre and postmasking are demonstrated. Finally, some more complex cases in the time or frequency domain may be explored.

In this application we have experimented with two versions of the program. By using the standard QuickSig the experiments sometimes are somewhat time-consuming because the updating of the test signals between listening periods takes time to compute. In another case we utilized the QuickC30 and the TMS320C30 signal processor that minimizes all waiting periods. It is found important to try to avoid any delays in order to keep students interested in the task. The new RISC-based computers (e.g., PowerPC-based Macintosh) are capable to do this efficiently without extra hardware.

OTHER FEATURES AND FUTURE WORK

We have experimented with other DSP-related problem domains as well. Speech processing is an excellent topic for a CBE program. For example, an articulatory speech synthesizer shows the vocal tract profile where the user can move the articulators and hear the generated voice [11], [12]. An object-oriented database [13] gives an intuitive graphical access to transcribed and stored speech. Model-based sound synthesis of musical instruments [14], sound effects used in effect processors, and real-time simulation of room acoustics combined with headphone auralization [15] are other closely related topics.

So far our aim has been of tutorial kind. By a hands-on-like approach the student learns the functionality and operability of the basic concepts, resembling experimentation with ordinary (physical) equipment. The next step after the tutorial level should be to learn the role of a constructor (and straightforward designer). The best way, already widely utilized, is the block-diagram (or patch editor) approach, where the user has DSP operations as building blocks that may be wired into data-flow diagrams. In our system we have a patch editor and real-time execution environment for the TMS320C30 processor. With the advent of fast RISC-based computers a single machine will be able to do the whole task in real time.

The final step in a CBE-program should be to integrate mathematics, already known from math courses, to the tutorial and constructional levels, in order to educate a master designer in signal processing. To support this level, integration of mathematics toolboxes, such as Mathematica and MATLAB, to QuickSig CBE is needed.

SUMMARY

In this paper we have described an environment for computer-based education applications, built on top of the QuickSig DSP software system and used to build CBE-applications. Two specific application domains are discussed in more detail: "Introduction to Signal Processing" and "Psychoacoustics". Special attention is paid to exploit modern hyper and multimedia features such as 16-bit sound, advanced graphical user interface with direct manipulation of objects, and navigation maps.

The new generation RISC-processor computers, even portable ones, will allow for high-quality real-time audio DSP without extra hardware. They are thus wery vell suited to signal processing education purposes. The main challenge that remains is to develop high-quality software covering the many subtopics related to signal processing.

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