REAL-TIME DSP FOR ADAPTIVE FILTERS: A TEACHING OPPORTUNITY

Michael G. Morrow

Dept. of Elec. and Comp. Engineering
University of Wisconsin - Madison, WI
morrow@ieee.org

Cameron H. G. Wright

Dept. of Elec. and Comp. Engineering
University of Wyoming, WY
c.h.g.wright@ieee.org

Thad B. Welch

Dept. of Elec. and Comp. Engineering
Boise State University, ID
t.b.welch@ieee.org

ABSTRACT

Engineering educators have found that well-chosen demonstrations of digital signal processing (DSP) can result in improved comprehension and better retention of topics for the majority of our students. In this paper, we discuss the use of a demonstration using real-time DSP to implement a basic adaptive filter for noise cancellation, utilizing newly-available DSP hardware from Texas Instruments. A description of the new hardware platform, the demonstration used, and results from a short survey administered to the students regarding the demonstration are provided.

***Index Terms*—** digital signal processing, adaptive filters, engineering education

1. INTRODUCTION

Digital signal processing (DSP) has become one of the "must know" topics that many employers expect of new electrical and computer engineering (ECE) graduates. It has been found that a true understanding of many fundamental DSP topics can be more fully realized by a student when they attempt to implement various DSP algorithms in real-time (typically in C), when compared to non-real-time implementations with tools such as MATLAB or LabVIEW [1]. In order to help students successfully transition from theory to real-time practice, there needs to be both a pedagogical method and an infrastructure in place to support them and target as many modes of learning as are reasonably possible.

This basic idea of using demonstrations to enhance learning for DSP is not new. Many engineering educators have recognized the need for, and reported on the results of using, interactive learning and demonstrations for this important subject area [2–6]. For many years, the authors of this paper have been suggesting and providing proven DSP teaching methodologies, hardware and software solutions, and DSP tools that have helped motivate students and faculty to implement real-time DSP-based systems to improve education in signal processing and related topics [7–18].

These efforts to promote the use of demonstrations and hands-on experiences for students have emphasized the fact that DSP is much more than just a collection of theories and problem solving techniques, and that hands-on experience with real-time hardware is extremely beneficial; it not only helps students master various DSP concepts but also improves student retention rates. Using real-time DSP as the catalyst, students can be more effectively motivated to explore and implement a wide variety of DSP topics in an environment in which they are limited only by their imagination.

The preferred pedagogical approach favored by the authors is a three-step method of teaching DSP [19]. First, we teach the theory along with interesting and motivating real-time demonstrations. We then have students implement a particular concept using MATLAB, until they are comfortable with the basic topic. Finally, we have them "de-vectorize" their MATLAB code and convert it to C, in order to compile and run it in real-time on high-performance DSP hardware.

There are several choices of hardware the professor may make if using this approach. For the very popular Texas Instruments (TI) processors, the Spectrum Digital C6713 DSK, the Logic PD Zoom OMAP-L138 Experimenters Kit (ZEK), and the new Texas Instruments OMAP-L138 Low Cost Development Kit (LCDK) are all highly capable platforms currently available for real-time DSP, and all can be used effectively with the recommended pedagogical method. The LCDK is the newest of the three, having been introduced in the latter part of 2012. Its capabilities are equal to or greater than the other two platforms, and it is the lowest in cost (only $195 as of this writing). One capability germane to this paper is the access to not only "line in" audio but also to "microphone in" audio that can support generic stereo microphones or dual monaural microphones.

2. THE LCDK BOARD FOR DSP

We have used many hardware platforms over the years to support our approach of incorporating demonstrations and hands-on experiences for DSP students. Since TI has been consistently supportive of DSP educators, we have almost always used their boards with either fixed- or floating-point Texas Instruments (TI) processors, such as the C50, C31, C6201, C6211, C6711, C6713, and most recently the multi-core OMAP-L138 (which includes both a C6748 core and an ARM926 core). Of these processors, several are now only of historical interest, while the boards based on the C6713 and the OMAP-L138 remain our primary targets of interest.

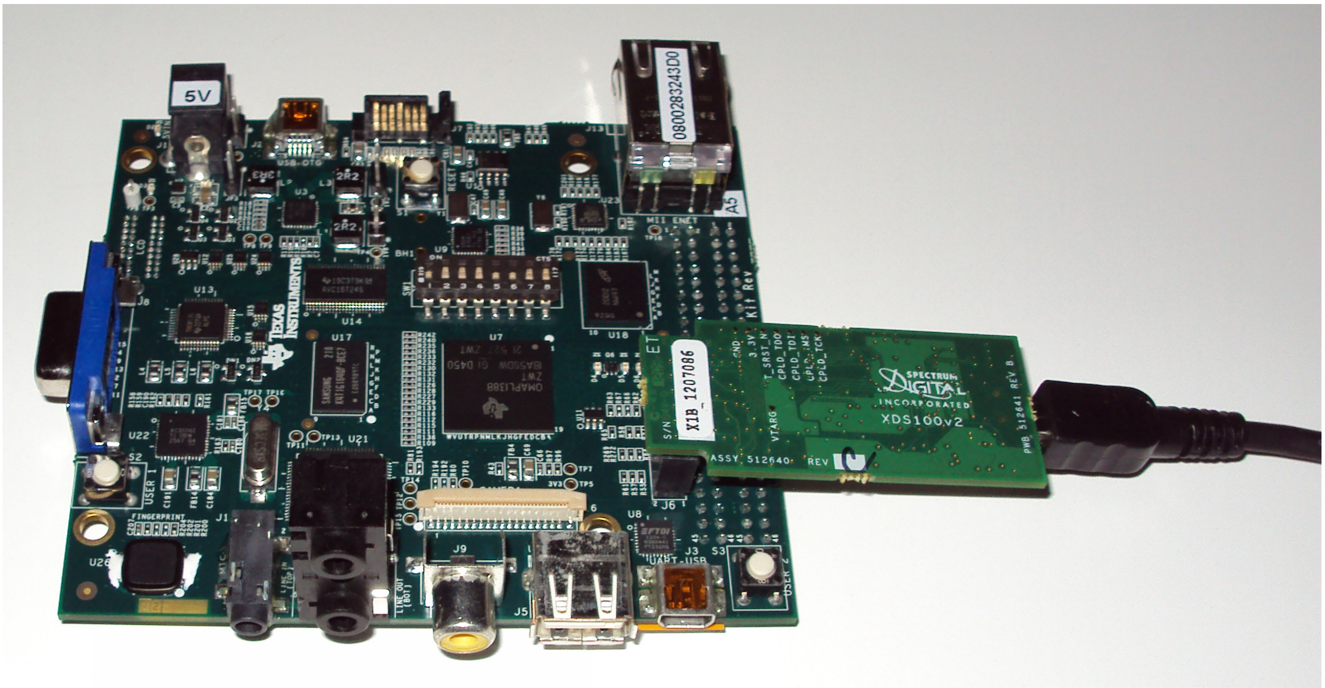

Fig. 1. The new Low Cost Development Kit (LCDK) from Texas Instruments, with a multi-core OMAP processor and multiple I/O options.

As mentioned in the previous section, the newest board based on the OMAP-L138 is called the Low Cost Development Kit (LCDK) [20], shown in Fig. 1. A specific comparison of the LCDK with the C6713 DSK and the Logic PD Zoom OMAP-L138 Experimenters Kit (ZEK) can be found at `http://www.rt-dsp.com/2nd_ed/board_comparison.pdf`. In general, the LCDK is superior in most ways to the C6713 DSK and equally (if not more) capable than the ZEK. For teaching purposes, the LCDK is our preferred choice. One of many reasons for that choice is that while the ZEK supports only audio "line in" and "line out," the less expensive LCDK also supports "microphone in," which allows more flexibility for demonstrations.

Another advantage of the LCDK over the ZEK is more subtle. While both OMAP-L138 boards use the identical audio codec chip, the two manufacturers chose to integrate that codec into the overall board design in different ways. In particular, the power supply decoupling of the codec chip is much better on the LCDK, and this is important to some applications. See the comparison on the `rt-dsp.com` site mentioned previously for plots and a discussion of the noise related to this design difference. To be fair to both boards, it should also be noted that in order to program the LCDK in C (using Code Composer Studio from TI), you need an additional item: an inexpensive XDS100 emulator, since it is not part of the main board as it is with the ZEK. These emulators are available at a suggested retail price of $79, from TI's eStore or a variety of third-party vendors.

The adaptive filter demonstration described in this paper was accomplished using the LCDK.

3. ADAPTIVE NOISE CANCELLATION: A DEMO

One application of this now-available stereo microphone capability, that is useful for teaching DSP to students, is adaptive noise cancellation using an adaptive digital filter [21–23]. As shown in Fig. 2, two signals are provided to the system. The upper signal contains "signal plus noise" while the lower signal contains "correlated noise." The adaptive filter (the box labeled with the transfer function $H_k(z)$ in the figure) uses the error signal to adjust its transfer function in real-time so as to optimally cancel out the noise at the output. The theory is straightforward, and will not be repeated in any greater detail here.

For the scenario we present to our students, we are trying to duplicate the noisy environment of a firefighter at the scene of an emergency. The "signal plus noise" represents the combined signals from the firefighter's helmet-mounted microphone (where the "signal" is his/her voice, and the "noise" is from a chainsaw running in the background being used to clear debris). The "correlated noise" signal represents just the chainsaw's signal, as detected by a second microphone located on the firefighter, but nowhere near his/her mouth. The adaptive filter's purpose is to enhance the voice signal so that the firefighter may communicate effectively, for example, when using a radio.

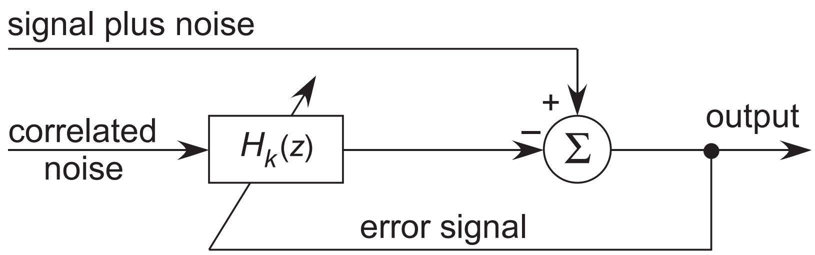

Fig. 2. A basic block diagram of an adaptive filter used for noise cancellation.

4. CLASSROOM RESULTS

A three-week filter design module was included in our traditional senior-level DSP course during the Fall 2012 semester. Since a thorough coverage of filter design can easily span an entire academic year, and adaptive filters is routinely taught as a separate course, we decided to devote a single class period to adaptive filters to help whet our student's appetite for additional DSP content. This additional content could be in the form of additional coursework, senior design projects, follow-on graduate studies, or research.

Given the severe time constraint, only the noise canceling adaptive filter configuration was discussed. The fact that an almost unlimited number of adaptive structures exist was also mentioned to motivate additional student interest.

Following our established pedagogy, we reviewed the underlying theory, conducted MATLAB simulations, and then ported the algorithm into C for execution in real-time on a LCDK. Student opinions of this topic were measured using a five-point Likert-scale survey. The item most pertinent to this paper stated,

"The adaptive filter demonstration helped me understand the underlying concepts." The allowed responses were,

1. strongly disagree
2. disagree
3. undecided
4. agree
5. strongly agree.

The average response of the 14 survey participants to this question was 3.86, with a standard deviation of 1.03. Of the 14, only two students circled the "2 - disagree" response and none circled the "1 - strongly disagree" response.

5. CONCLUSIONS

The majority of our students definitely agreed with our belief that the adaptive filter demonstration helped them understand the underlying concepts. This was in spite of the fact that only one class period was spent on this topic. A similar survey question asking about the efficacy of DSP demonstrations in general averaged a score of 4.29 on the same scale. This certainly is in line with our anecdotal information regarding the teaching effectiveness of demonstrations in the classroom.

It should be noted that this adaptive filter demonstration will not work on either the C6713 DSK or the ZEK unless external microphone preamplifiers are used to connect the two microphones to the line input. Accessing the dual microphone inputs on the LCDK does require a stereo-to-dual-mono splitter cable, but this is a very small inconvenience compared to having to buy or assemble two external microphone preamplifiers.

We strongly encourage faculty who teach DSP to incorporate demonstrations and hands-on experience with real-time hardware for their students. We have made various resources widely available to help in this endeavor [24, 25].

6. REFERENCES

[1] C. H. G. Wright, T. B. Welch, D. M. Etter, and M. G. Morrow, "Teaching DSP: Bridging the gap from theory to real-time hardware," *ASEE Comput. Educ. J.*, pp. 14–26, July–September 2003.

[2] C. S. Burrus, "Teaching filter design using MATLAB," in *Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing*, pp. 20–30, Apr. 1993.

[3] R. F. Kubichek, "Using MATLAB in a speech and signal processing class," in *Proceedings of the 1994 ASEE Annual Conference*, pp. 1207–1210, June 1994.

[4] R. G. Jacquot, J. C. Hamann, J. W. Pierre, and R. F. Kubichek, "Teaching digital filter design using symbolic and numeric features of MATLAB," *ASEE Comput. Educ. J.*, vol. VII, pp. 8–11, January–March 1997.

[5] J. H. McClellan, C. S. Burrus, A. V. Oppenheim, T. W. Parks, R. W. Schafer, and S. W. Schuessler, *Computer-Based Exercises for Signal Processing Using* MATLAB 5. MATLAB Curriculum Series, Upper Saddle River, NJ (USA): Prentice Hall, 1998.

[6] J. W. Pierre, R. F. Kubichek, and J. C. Hamann, "Reinforcing the understanding of signal processing concepts

using audio exercises," in *Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing*, vol. 6, pp. 3577–3580, Mar. 1999.

[7] C. H. G. Wright and T. B. Welch, "Teaching DSP concepts using MATLAB and the TMS320C31 DSK," in *Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing*, Mar. 1999. Paper 1778.

[8] M. G. Morrow and T. B. Welch, "winDSK: A windows-based DSP demonstration and debugging program," in *Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing*, vol. 6, pp. 3510–3513, June 2000. (invited).

[9] M. G. Morrow, T. B. Welch, C. H. G. Wright, and G. W. P. York, "Demonstration platform for real-time beamforming," in *Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing*, May 2001. Paper 1146.

[10] C. H. G. Wright, T. B. Welch, D. M. Etter, and M. G. Morrow, "Teaching hardware-based DSP: Theory to practice," in *Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing*, vol. IV, pp. 4148–4151, May 2002. Paper 4024 (invited).

[11] T. B. Welch, R. W. Ives, M. G. Morrow, and C. H. G. Wright, "Using DSP hardware to teach modem design and analysis techniques," in *Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing*, vol. III, pp. 769–772, Apr. 2003.

[12] T. B. Welch, M. G. Morrow, and C. H. G. Wright, "Using DSP hardware to control your world," in *Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing*, vol. V, pp. 1041–1044, May 2004. Paper 1146.

[13] T. B. Welch, C. H. G. Wright, and M. G. Morrow, "Caller ID: An opportunity to teach DSP-based demodulation," in *Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing*, vol. V, pp. 569–572, Mar. 2005. Paper 2887.

[14] T. B. Welch, C. H. G. Wright, and M. G. Morrow, "Teaching rate conversion using hardware-based DSP," in *Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing*, vol. III, pp. 717–720, Apr. 2007.

[15] C. H. G. Wright, M. G. Morrow, M. C. Allie, and T. B. Welch, "Enhancing engineering education and outreach using real-time DSP," in *Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing*, vol. III, Apr. 2008.

[16] T. B. Welch, C. H. G. Wright, and M. G. Morrow, "Software defined radio: inexpensive hardware and software tools," in *Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing*, pp. 2934–2937, Mar. 2010.

[17] M. G. Morrow, C. H. G. Wright, and T. B. Welch, "winDSK8: A user interface for the OMAP-L138 DSP board," in *Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing*, pp. 2884–2887, May 2011.

[18] M. G. Morrow, C. H. G. Wright, and T. B. Welch, "Old tricks for a new dog: An innovative software tool for teaching real-time DSP on a new hardware platform," *ASEE Comput. Educ. J.*, pp. 64–69, October–December 2011.

[19] T. B. Welch, C. H. G. Wright, and M. G. Morrow, *Real-Time Digital Signal Processing: From MATLAB to C with C6x DSPs*. Boca Raton, FL (USA): CRC Press, 2nd ed., 2012.

[20] Texas Instruments, "L138/C6748 Development Kit (LCDK)," 2012. `http://processors.wiki.ti.com/index.php/L138/C6748_Development_Kit_(LCDK)`.

[21] S. D. Stearns, *Digital Signal Processing with Examples in MATLAB*. Boca Raton, FL (USA): CRC Press, 2003.

[22] A. D. Poularikas and Z. M. Ramadan, *Adaptive Filtering Primer with MATLAB*. Boca Raton, FL (USA): CRC Press, 2006.

[23] S. Haykin, *Adaptive Filter Theory*. Upper Saddle River, NJ (USA): Prentice Hall, 1996.

[24] "RT-DSP website." `http://www.rt-dsp.com`.

[25] Educational DSP (eDSP), L.L.C., "DSP resources for TI DSKs," 2012. `http://www.educationaldsp.com/`.