USING DSP HARDWARE TO CONTROL YOUR WORLD

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ABSTRACT

In recent years, more and more students have been designing and implementing small systems using real-time DSP hardware. Given the decreasing cost and the increasing capability of DSP starter kits (DSKs) and evaluation modules (EVMs), these projects are being used by greater numbers of educators as a valuable pedagogical tool. To help keep the cost relatively low, most DSKs and EVMs are designed to work with audio input and output signals. Digital input/output pins are sometimes available, but the ability to easily control a number of electrical loads totaling several hundred watts does not readily exist. Yet DSP control of significant electrical loads can add depth and interest to many student projects.

This paper will discuss the design, construction, and use of a very compact, dual-tone multiple-frequency (DTMF) based decoder and power switching device. These devices have been successfully used by a number of undergraduate and graduate students to allow their DSP algorithms to *control their world*.

1. INTRODUCTION

For several years now, we 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 DSP-based systems in real-time [1–6]. These efforts have emphasized the fact that DSP is much more than just a collection of theories and problem solving techniques. Students can easily be motivated to explore and implement DSP-based systems in an environment where they are limited only by their imagination. This process can be facilitated through real-time demonstration programs such as winDSK and winDSK6 [7].

This paper describes the addition of DTMF-based control of a power switching device to allow more than just algorithms to be implemented in real-time. When using the basic configuration of low cost DSKs or EVMs, the available real-time outputs are typically limited to:

- Audio signals
- Light emitting diodes (LEDs) that can be turned on, off, or caused to flash
- Digital input/output pin (logic control - 0 or 5 volt output)
- Screen output to the host computer's monitor
- Or with a great deal of effort, stand-alone host computer applications with textual or graphical display windows

While these outputs are informative, the ability to turn on and off an electric motor, a light, or some other appliance would add considerably more interest for many students. For example, if a student was developing a voice/speaker recognition algorithm which had as its primary function the ability to recognize the designated user's voice and turn on or off a fan when the phrase "fan. . . on" or "fan. . . off" was detected, wouldn't it be a much more fulfilling project and demonstration if a fan actually turned on or off instead of one of the previously mentioned computer-based simulated responses? We believe that not only is such a system inspiring to the student who developed the DSP-based algorithm, but that it also significantly affects anyone who observes the completed system in operation.

During the design of the DTMF decoder and power switch box the system requirements changed several times as we gained a better idea of how we would use the device. The final system requirements follow:

1. Capable of being controlled by any DSP or microprocessor that is capable of controlled generation of DTMF signals
2. Sturdy yet compact enclosure with only 3 cabled connections. These connections are:
   (a) DC power for the decoder board
   (b) DTMF control signal from the DTMF signal generator (DSP/microprossor-based DTMF generation device)
   (c) AC power to supply the switched outlets that subsequently feed the switched appliances

Fig. 1. Photograph of the DTMF decoder and switch box (rear view).

Fig. 2. Photograph of the DTMF decoder and switch box (front view, cover removed).

3. Capable of independently controlling up to 5 appliances (due to the size of the enclosure, only 4 electrical outlets were installed)

4. Power on/system status indication

5. Upgradeable to serial port (RS-232) control

2. DTMF DECODER AND SWITCH BOX THEORY OF OPERATION

The DTMF decoder and switch box (rear view), as shown in Figure 1, is designed to permit a DSP device to control high power external loads by issuing commands encoded as audio DTMF signals. The DTMF decoder and switch box receives those commands, and controls the connected loads accordingly. A small microcontroller directs the overall operation of the DTMF decoder and switch box, using a DTMF decoder integrated circuit to perform the actual DTMF audio tone pair decoding. Five independent relays permit the control of loads requiring up to 10A at 250VAC or 32VDC. During normal operation, a front panel status LED flashes at a constant rate while the unit is waiting for a command packet to be detected. The front panel of the DTMF decoder and switch box is shown in Figure 2. Once the start of command packet has been detected, the LED will remain on (constantly) until either the command packet has been fully received, an error is detected in the packet, or a four second timeout period expires. A functional block diagram of the system is shown in Figure 3.

When a valid tone pair is detected by the DTMF decoder, an interrupt is signalled to the microcontroller. The microcontroller then reads the tone pair code from the decoder, and places the symbol in an internal queue for further processing. The decoder is capable of detecting DTMF tone pairs over a wide range of amplitudes, but is optimized for line-level (+/-1V) signals. Tones should have a minimum 40ms duration and there should be at least 60ms between tones. The syntax for the DTMF command packet is:

Command	DTMF Sequence
Turn individual relay ON	* X 1 #
Turn individual relay OFF	* X 0 #
Turn all relays ON	* A 1 #
Turn all relays OFF	* A 0 #

Note: Replace the X in the individual relay commands with ‘1’ through ‘5’ to control relays 1 through 5, respectively.

If a complete and valid packet is received within the 4 second timeout window, the corresponding relay(s) are energized or de-energized as appropriate. If the timeout period expires before a complete sequence is detected, or an erroneous sequence is detected, the system discards the received data and immediately resumes looking for the start of a valid packet.

The optional RS-232 module works similarly to the DTMF module, except that the RS-232 packets are constructed from the ASCII characters corresponding to the DTMF symbols to be received. Once received, incoming RS-232 characters are placed in a separate queue for processing, and are acted on if a valid packet is received within the timeout period. The system is capable of handling simultaneous DTMF and RS-232 commands by implementing a first-started, first-executed packet handling algorithm. The RS-232 port and internal arrangement of the DTMF decoder and switch box is shown in Figure 4.

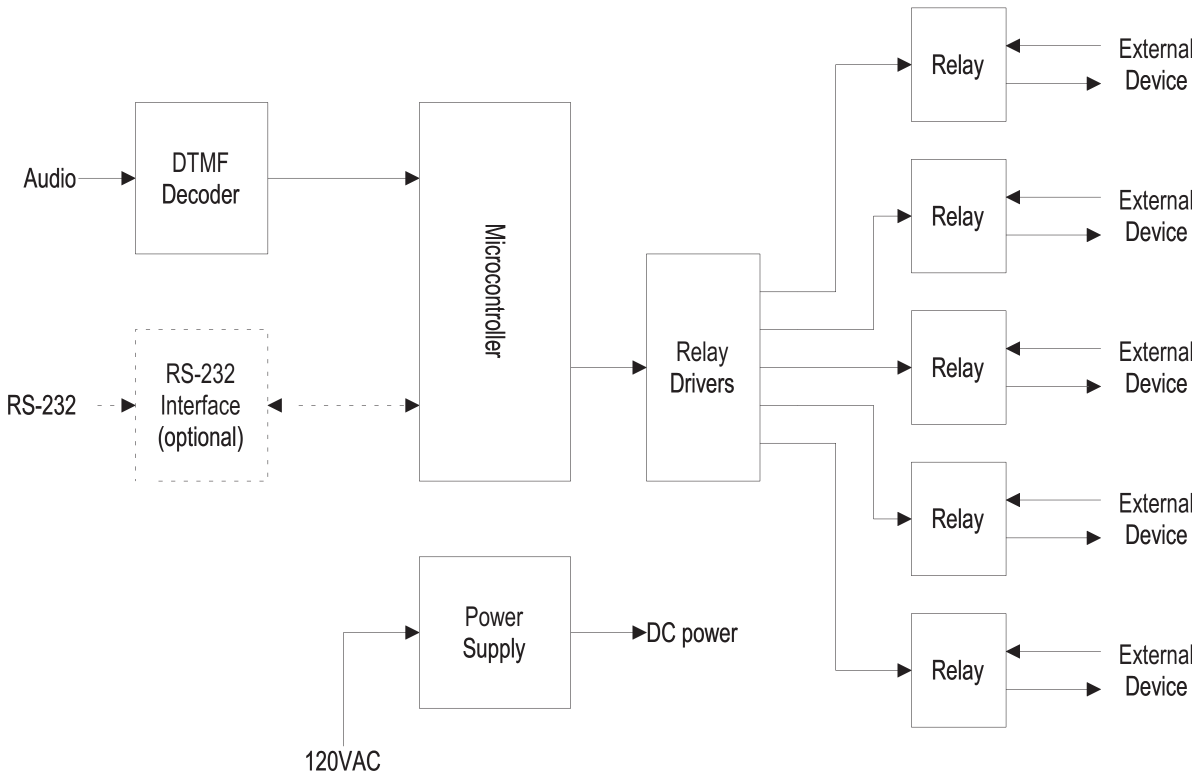

Fig. 3. DTMF decoder and switch box functional block diagram.

Fig. 4. Photograph of the DTMF decoder and switch box (side view, cover removed).

3. A FEW APPLICATIONS

Although not an all inclusive list, we have used, or plan to use, the DTMF decoder and power switch box for the following applications:

1. As described earlier, a voice/speaker recognition algorithm which has as its primary function the ability to recognize the designated user's voice and turn on or off an appliance when the correct phrase is detected. Since multiple switched outlets are available, up to 4 appliances can be controlled without circuit modification.

2. Access control system. This project gathers biometric signals from a person desiring entry into a building or room. The student develops biometric identification algorithms that once properly identified, open or unlock a door. The opening or unlocking is accomplished using a channel of the DTMF decoder and switch box.

3. Priority-based caller ID system. This project uses a telephone coupler to connect one of the channels of a DSP board's input (analog-to-digital converter, ADC) to the telephone line. We used a modified COMREX TCB-1 manual coupler to protect the ADC from the approximately 90 volts associated with the telephone ring signal. The modification involved adding a coupling capacitor to prevent the telephone system from sensing that the phone was off hook. Preventing the telephone system from sensing the off hook condition is required since failure to do so will result in the ter-

mination of the caller ID signaling.

For this system, the student develops an FSK demodulator that maps the incoming caller ID signal back to ASCII characters (the caller ID information). Priority telephone numbers, which are stored in a separate ASCII-file, are compared to the incoming caller ID information. File matches are used to activate a channel of the DTMF decoder and switch box. This could be in the form of a light, an alarm, or some other novel electrical appliance.

4. Home annunciator system. This project allows a hearing impaired person to take advantage of audio-based household alarms and warnings. Typical alarms and warnings include, fire/smoke detectors, door bells, telephone, and appliance indicators. For this system, the student develops algorithms to detect these audio signals. Detected signals are used to activate a channel of the DTMF decoder and switch box. This could be in the form of a light, an alarm, or some other novel electrical appliance.

5. Home alarm system. For this system, the student develops algorithms to detect the sound of breaking glass. The detected "breaking glass" signal is used to active a channel of the DTMF decoder and switch box. This could be in the form of a light, an alarm, or some other novel electrical appliance.

6. Stage lighting control system. For this system, the student develops a beamforming algorithm that determines the most likely position of a moving sound source on a theatrical stage. Using the 4 channel DTMF decoder and switch box, up to 4 directional stage lights (spotlights) can be controlled. Only the stage light that illuminates that portion of the stage closest to the location of the sound source will be energized.

4. CONCLUSIONS

We have described the development of our DTMF decoder and power switch box. This system has added a significant power handling capability to any DSP or microprocessor capable of controlled generation of DTMF signals. These boxes have been exceedingly well received by both our undergraduate and graduate students.

The winDSK6 software used to demonstrate the effectiveness of the DTMF decoder and power switch boxes is for educational, non-profit use, and we invite user suggestions for improvement. See `http://eceserv0.ece.wisc.edu/~morrow/software/`. Interested parties are also invited to contact the authors via e-mail.

5. REFERENCES

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