A Study on Implementation of Voice Control System Using DSP Chip

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ABSTRACT

In this research, we design and implement a voice control system and use it to control a robot. The voice control system is implemented on a TMS320C6713 DSK board, which is a digital signal processor circuit board from Texas Instruments. The speech feature vector consists of 12 linear prediction derived cepstral coefficients, log energy and their first and second order derivatives. The automatic speech recognition system used in this study is based on the hidden Markov model. The Matlab and Simulink are used as a high level system development environment in this study. The system designed as a Simulink model consists of several interconnected modules, each of which is either represented as a block of embedded function or a built-in block in Simulink. We use the Embedded Coder and the Real-time Workshop to convert the Simulink model the associated Matlab programs into C language programs. Using the Code Composer development environment, the C language programs are compiled and linked into an executable program and then downloaded into the DSK board. The recognition result is then used to drive an infrared remote controller to send a control signal to control the robot. The parameters of the speech models are obtained from offline training. The modules of the real-time automatic recognition system include a voice acquisition and digitization unit, a framing unit, an end-point detection unit, a feature extraction unit, a data buffering unit, a core speech decoding unit, and a control signal output unit. Experimental results show that the voice control system can effectively control the robot.

Keywords : digital signal processor、automatic speech recognition、hidden Markov Model、voice control

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