EASYTAD Chip for an all Digital Telephone Answering Device with Super Low Rate Coder and True FULL Duplex SpeakerPhone

General Description

The D6455A/B chip is a digital speech/signal processing subsystem that implements all functions of TRUESPEECH® speech compression and voice prompts, telephone line signal processing, flash memory management and True FULL Duplex SpeakerPhone™ for an all digital answering machine. The D6455A/B is fully controlled by the system Host through a simple interface protocol. The Host controller provides activation and control of all system functions such as speech recording and playback, DTMF and call progress tone detection, DTMF and tone generation, and voice prompting.

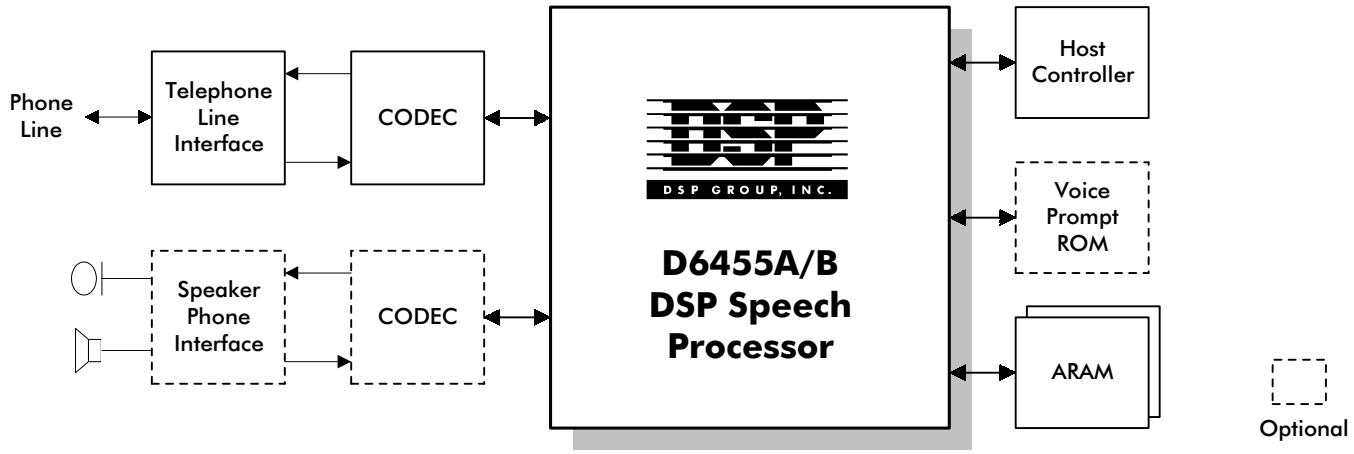

Figure 1. D6455A/B System Block Diagram

Key Features

- High-quality, new super low rate TRUESPEECH digital speech compression allowing 25-27 minutes of recording time per each 4 Mbit of memory
- Flexible storage of incoming messages (ICM), outgoing messages (OGM). Supports multiple OGMs and multiple mailboxes
- TRUESPEECH, natural-sound voice prompting, for Day/Time stamp and voice instructions
- DTMF generation and detection with near-end echo cancellation for superior performance
- Simple 8-bit Host Interface
- Fully compatible with the D6305B. Host software compatible, pin-for-pin replaceable
- Works with 256Kx4 (1 Mbit)*, 1Mx4 (4 Mbit) or 4Mx4 (16 Mbit)* ARAM's
- True FULL Duplex SpeakerPhone with both acoustical and near-end echo cancellation
- Programmable Thresholds for VOX, DTMF, and Call Progress
- Digital Volume Control
- Supports "offset playback" for jumps within a message
- Low power consumption
- Special commands for production line testing

* Supported in D6455B only

Device Configuration and System Components

STANDARD COMPONENTS

- D6455A/B-11 Digital Telephone Answering Device (TAD) processor (80-pin PQFP)—1 each

ADDITIONAL SYSTEM COMPONENTS

These are supplied by the customer according to DSP Group's specifications.

- D0000-29 Analog I/O Interface (16-pin DIP)—1 each, 2 for SpeakerPhone

One of the following:

- D0000-37C 1-Mbit ARAM Message Memory (SOJ)—Up to four devices per system (D6455B only)
- D0000-36C 16-Mbit ARAM Message Memory (SOJ)—Up to two devices per system (D6455B only), or
- D0000-35C 4-Mbit ARAM Message Memory (SOJ)—Up to four devices per system

OPTIONAL SYSTEM COMPONENTS

EPROM/ROM - For Voice Prompt storage an external EPROM/ROM (access time, 300 ms or less) is required -each 32K x 8 block supports 65 seconds of Voice Prompt (up to 64K).

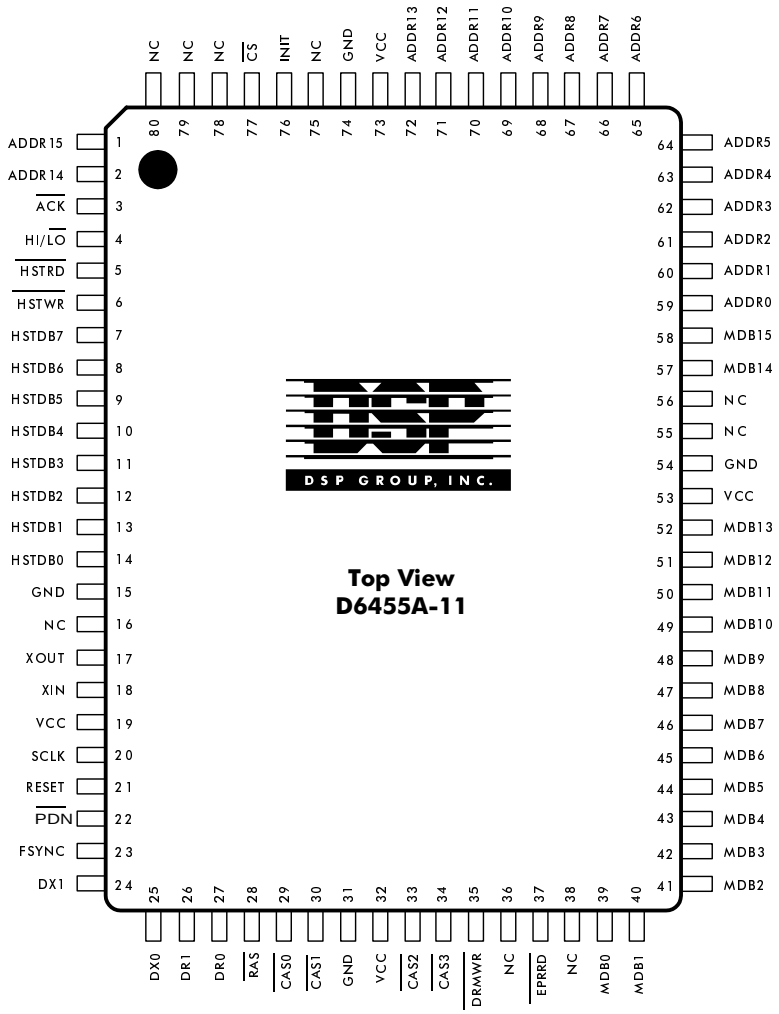

Figure 2. D6455A/B Pin Diagram

System Functions

All of the speech processing tasks are done by the DSP Speech Subsystem. This allows the use of a very low cost microcontroller to be used for basic control of the system. The Host need only send high level commands to perform functions such as Record Message or Delete Message and the operation will be performed by the DSP subsystem which will report the status of the operation to the Host. All memory interface and management will be taken care of by the DSP requiring the Host to only handle high level system functions. A summary of the functions performed by the DSP Speech Subsystem and Host Controller are shown in Figure 3, below.

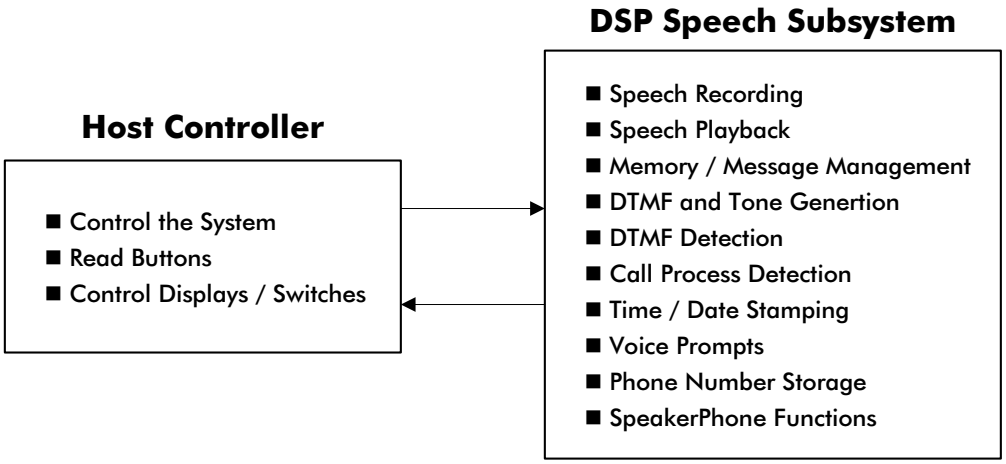

Figure 3. System Functions

Simple Hardware Interface

The hardware interface between the D6455A/B and the Host Controller is a simple one requiring only an 8 bit parallel port and 4 handshake lines. The Host writes high level Commands to the D6455A/B and the D6455A/B will respond with Status information. Once a Command is issued the D6455A/B will use the ACK pin to indicate that the Status is available to be read. The hardware interface between the D6455A/B and Host is shown in Figure 4 below.

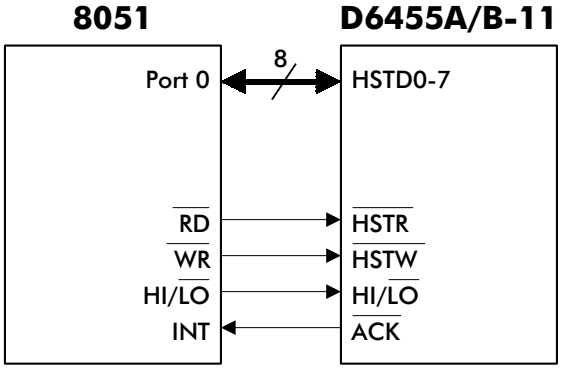

Figure 4. D6455A/B / Host Interface

Benefits of the D6455A/B

- New Super Low Rate TRUESPEECH digital speech compression allowing 25-27 minutes of recording time in only 4 Mbits of memory. This reduces the memory requirement or makes more recording time available in your design
- Allows flexibility in design for features such as multiple mailboxes and multiple OGM's allowing design of a product that is truly a Personal Voice Mail System
- The Host selectable thresholds for VOX, DTMF, and Call Progress allows for flexibility in design for various countries and different applications
- TRUESPEECH natural sounding voice prompts and time/date stamping allow design of a high quality and professional sounding product
- The True Full-Duplex SpeakerPhone capability allow a very professional sounding SpeakerPhone to be added to your product with a very minimal additional cost
- Flexibility of memory configuration, in the D6455A, allows the use of most readily available and cost effective memory

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