D6385A

All Digital Telephone Answering Device with True FULL Duplex SpeakerPhone includes Slave CODEC Operation for Digital Telephone Applications

General Description

The D6385A chip is a digital speech/signal processing subsystem that implements all functions of TRUESPEECH® speech compression and voice prompts, telephone line signal processing, memory management, FLEXISPEECH™ and True FULL Duplex SpeakerPhone™ for an all digital answering machine. In addition, the D6385A has the ability to operate as a CODEC, in slave mode, for use in Digital Telephone applications such as DECT. The D6385A 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.

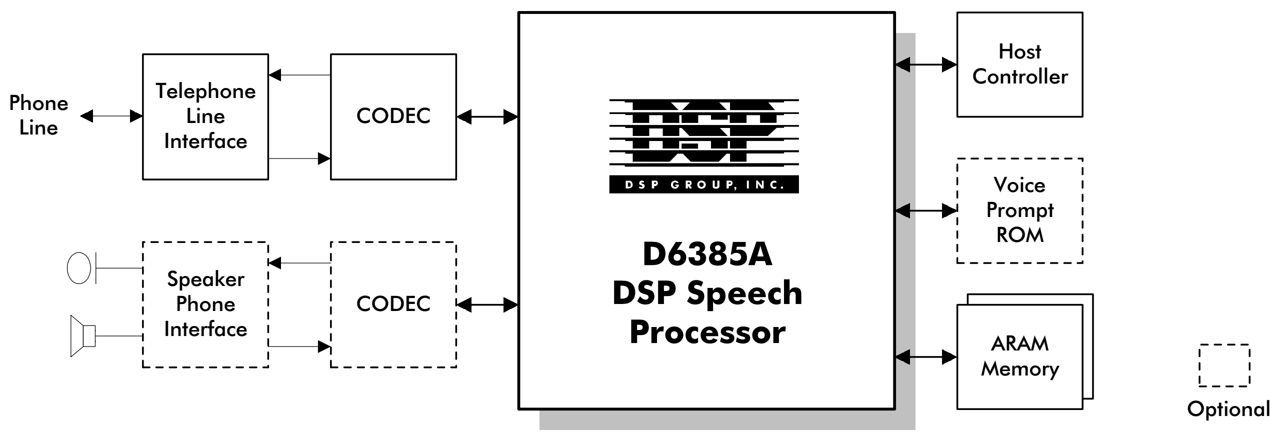

Figure 1. D6385A System Block Diagram

Key Features

- High-quality, variable low rate TRUESPEECH digital speech compression allowing 14-15 minutes of recording time per each 4 Mbit ARAM (Audio Grade DRAM)
- TRUESPEECH, natural-sound voice prompting, for Day/Time stamp and voice instructions
- Flexible storage of incoming messages (ICM), outgoing messages (OGM). Supports multiple OGMs and multiple mailboxes
- DTMF generation and detection with near-end echo cancellation for superior performance
- Storage of 32 telephone numbers for speed dialing
- Easy Host software conversion from the D6305B
- FLEXISPEECH variable speed, natural sound playback (50%-200%)
- Selectable Master/Slave CODEC operation
- Supports “offset playback” for jumps within a message
- True FULL Duplex SpeakerPhone with both acoustical and near-end echo cancellation
- Programmable thresholds for VOX, DTMF, and call progress
- Digital Volume Control
- Two programmable call progress tone filters
- Simple 8-bit Host Hardware Interface

Device Configuration and System Components

STANDARD COMPONENTS

- D6385A-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:

- D0000-35B/C* - 4Mb ARAM message memory (SOJ), up to four devices per system, or
- D0000-36B/C* - 16Mb ARAM message memory (SOJ), up to two devices per system

OPTIONAL SYSTEM COMPONENTS

- EPROM/ROM - Voice Prompt storage, each 32Kx8 device supports up to 35.4 seconds of Voice Prompts (up to 64K)

* Note: D0000-35C or D0000-36C are required if FLEXISPEECH is desired

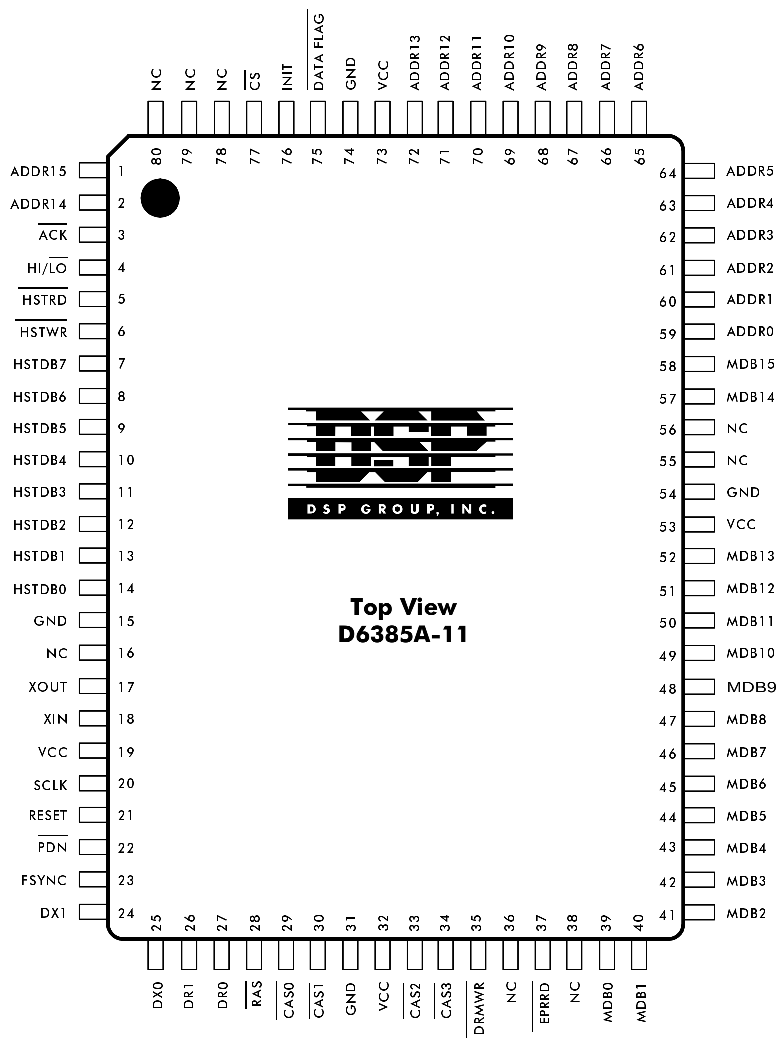

Figure 2. D6385A-11 Pin Diagram

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.

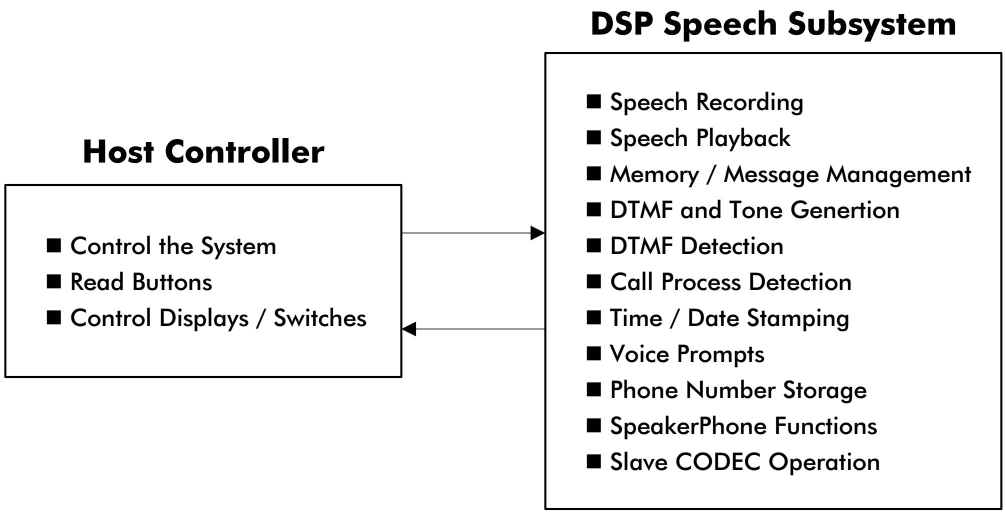

Figure 3. System Functions

Simple Hardware Interface

The hardware interface between the D6385A 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 D6385A and the D6385A will respond with Status information. Once a Command is issued the D6385A will use the ACK pin to indicate that the Status is available to be read. The hardware interface between the D6385A and Host is shown in Figure 4 below.

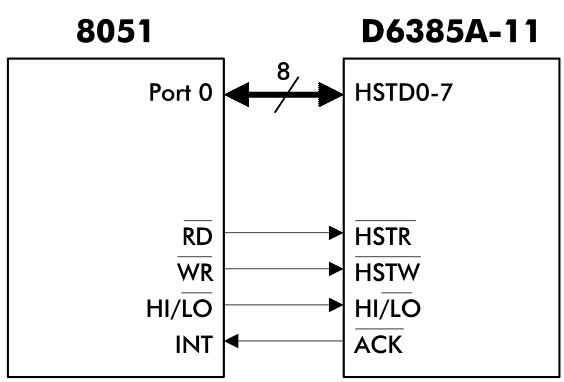

Figure 4. D6385A / Host Interface

Benefits of the D6385A

- High-Quality, variable low rate TRUESPEECH digital speech compression allowing 14-15 minutes of recording time in 4 Mbits of ARAM
- TRUESPEECH natural sounding voice prompts and time/date stamping allow design of a high quality and professional sounding product
- Operation as a Slave CODEC
  — Reduces the system cost by eliminating the need for an extra CODEC
- 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, DTFM and Call Progress allows flexibility in design for various countries and different applications
- The True Full-Duplex SpeakerPhone capability allow a very professional sounding SpeakerPhone to be added to your product with a very minimal additional cost

DSP Group, Inc.
Headquarters:
3120 Scott Boulevard
Santa Clara, CA 95054
Tel: (408) 986.4300
Fax: (408) 986.4490

Japan Office:
DSP Group
Koizumi Building, Sixth Floor
1-29-1, Nishi Gotanda
Shinagawa-Ku
Tokyo 141, Japan
Tel: (81) 3.5496.1611
Fax: (81) 3.5496.1615

Asia Pacific Technical Support Center:
Room 401, Nan Fung
Commercial Center
No. 19, Lam Lok Street
Kowloon Bay
Kowloon, Hong Kong
Tel: (852) 795.7421
Fax: (852) 795.7839

DSP Group Europe:
18 rue de l'effort mutuel
91300 Massey
France
Tel: (33) 0768.6754
Fax: (33) 1.6010.5187

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