EASYTAD™ Chip for an all Digital Telephone Answering Device with True FULL Duplex SpeakerPhone® and PCM Voice Prompt

General Description

The D6365B/C 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 D6365B/C is fully controlled by the system Host through a simple interface protocol. The Host processor 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.

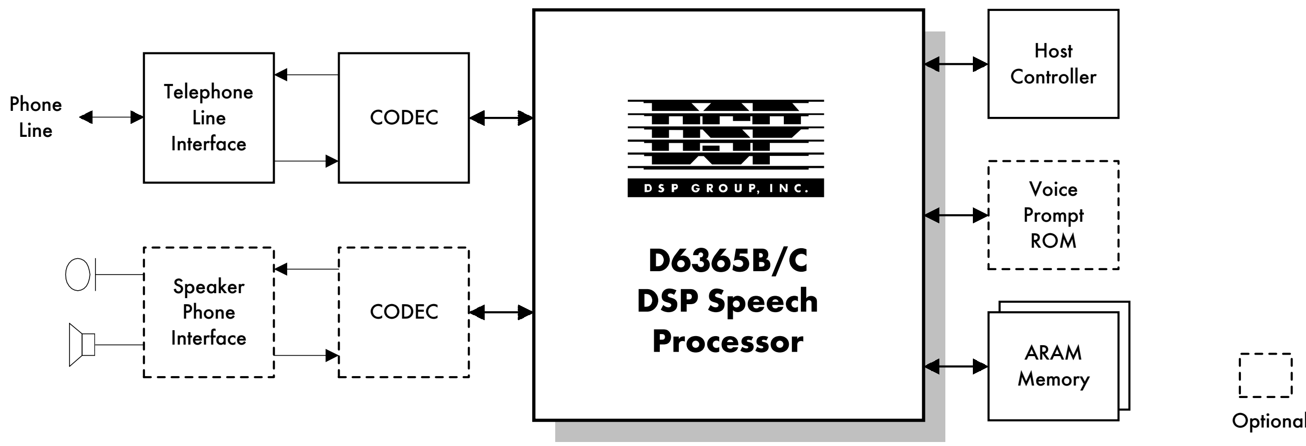

Figure 1. D6365B/C System Block Diagram

Key Features

- High-quality, low rate TRUESPEECH digital speech compression allowing 15-17 minutes of recording time per each 4 Mbit of memory
- Flexible storage of incoming messages (ICM) and outgoing messages (OGM), supporting multiple OGMs and multiple mailboxes.
- DTMF generation and detection with near-end echo cancellation for superior performance
- FLEXISPEECH™ variable speed, natural sound playback (50% - 200%)
- Supports "offset playback" for jumps within a message
- Fully compatible with the D6365A
- CID/General purpose storage (D6365C) and Telephone Number storage
- True FULL Duplex SpeakerPhone with both acoustical and near-end echo cancellation
- Flexible combination of the TRUESPEECH and PCM voice prompt in one storage ROM (up to 1 Mbyte) allows high quality and long voice prompt playback time
- Caller ID and CID on Call Waiting detection (Bell 202 and V.23) (D6365C)
- Works with 1M x 4 (4 Mbit) or 4M x 4 (16 Mbit) ARAM's (fast page mode and EDO)
- Programmable sensitivity of the DTMF, VOX, CAS, and CPT detectors
- Digital volume control
- 3.3V or 5V operation
- PCM recording (D6365C)

Device Configuration and System Components

STANDARD COMPONENTS

- D6365B/C-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

- Codec — 1, 2 for SpeakerPhone

One of the following:

- 4 Mbit ARAM Fast Page Mode and EDO — Up to four devices per system, or
- 16 Mbit ARAM Fast Page Mode and EDO (2K and 4K refresh) — Up to two devices per system

OPTIONAL SYSTEM COMPONENTS

- External EPROM/ROM for Voice Prompt storage is required (up to 1Mbyte is supported).

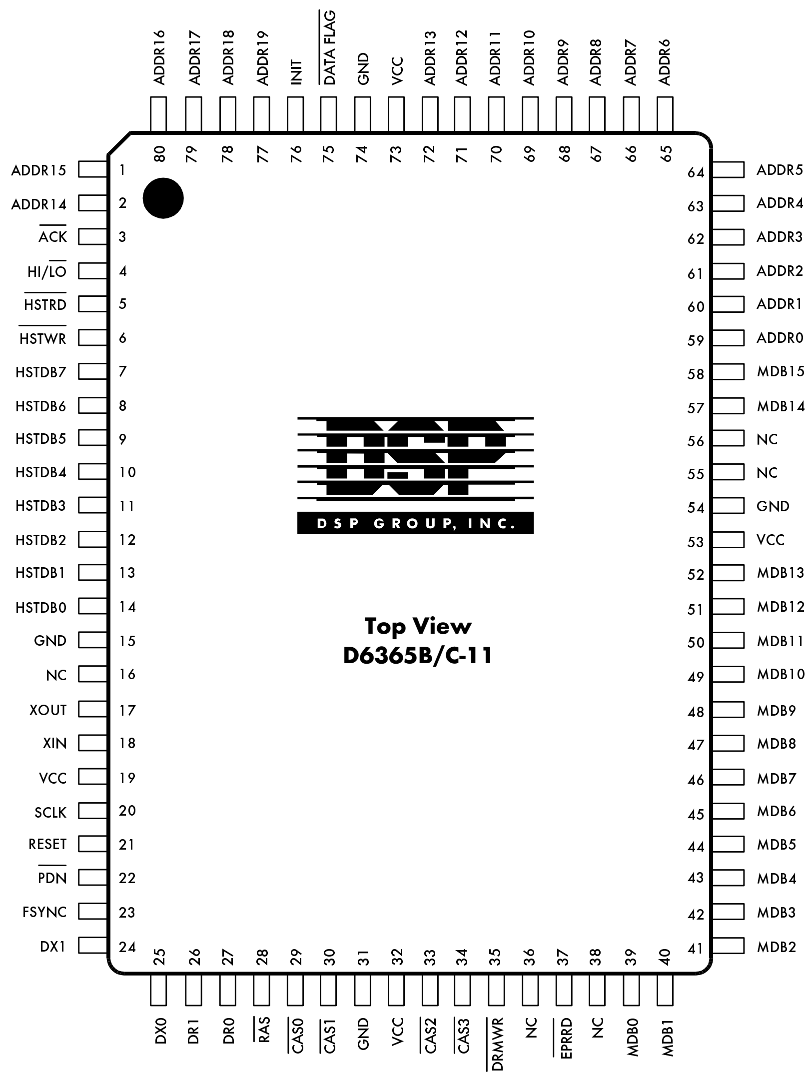

Figure 2. D6365B/C-11 Pin Diagram

System Functions

All of the speech and signal processing tasks are done by the D6365B/C. This allows the use of a very low cost microcontroller to be used for basic control of the system. The Host needs to send high level commands to perform functions such as Record Message, Playback, or Delete Message. The operation is performed by the D6365B/C which reports the status of the operation to the Host. All memory interface and management is performed by the D6365B/C, requiring the Host to only handle control functions. A summary of the functions performed by the D6365B/C and Host Controller are shown in Figure 3, below.

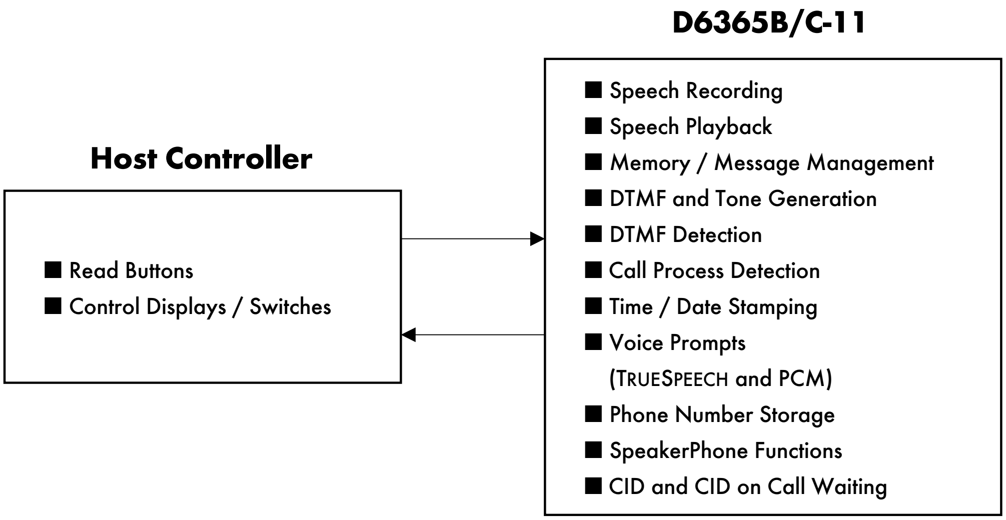

Figure 3. System Functions

Simple Hardware Interface

The hardware interface between the D6365B/C and the Host Controller is simple, requiring only an 8-bit parallel port and 4 handshake lines. The Host writes high level commands to the D6365B/C, and the D6365B/C responds with status information. Once a command is issued, the D6365B/C uses the ACK pin to indicate that the status is available to be read. The hardware interface between the D6365B/C and Host is shown in Figure 4 below.

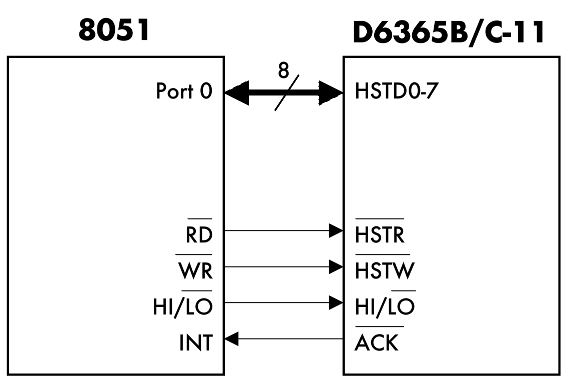

Figure 4. D6365B/C - Host Interface

Benefits of the D6365B/C

- Low Rate TRUESPEECH digital speech compression allows 15-17 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, enabling the design of a product that is truly a Personal Voice Mail System.
- The Host-selectable sensitivity of the DTMF, VOX, CAS, and CPT detectors allow for flexibility in design for various countries and different applications.
- Supports 1M x 4 and 4M x 4 (4K and 2K refresh) EDO and Fast Page mode ARAMs.
- Up to 1 Mbyte of the voice prompt storage allows a flexible combination of the TRUESPEECH and PCM prompts.
- Up to 2 min 25 sec of the PCM voice prompts, or up to 20.8 min of the TRUESPEECH voice prompts, or any combination of both is available.
- The True FULL Duplex SpeakerPhone capability allows a professional sounding speakerphone to be added to your product with very minimal additional cost.
- The Caller ID and Caller ID on Call Waiting eliminates the need for extra components to include these important features in your design (D6365C).
- PCM recording and playback allows music and sound effects recording (D6365C).

DSP Group, Inc.:
3120 Scott Boulevard
Santa Clara, CA 95054
Tel: (408) 986.4300
Fax: (408) 986.4490
http://www.dspg.com

DSP Group Europe:
18 rue de l'effort mutuel
91300 Massey
France
Tel: (33) 6.0768.6754
Fax: (33) 1.6010.5187

Nihon DSP Group, K.K:
Yasuda Kasai Building – 2nd Floor
2-3-1 Higashi Gotanda
Shinagawa-Ku, Tokyo 141
Japan
Tel: (81) 3.3449.7851
Fax: (81) 3.3449.8006

DSP Solutions, LTD:
Unit 1923, 19/F Metro Centre 1
32 Lam Hing Street, Kowloon Bay
Kowloon, Hong Kong
Tel: 852.2750.7325
Fax: 852.2305.0640

DSP Technology, LTD:
#A-1216, Champs-Elysse Center
889-5, Daechi-Dong, Kangnam-Ku
Seoul, 135-280, Korea
Tel: 822.554.7494
Fax: 822.554.7495

DSP Application, LTD:
12F-12, 79, Hsin Tai Wu Road, Sec. 1
Hsi Chih, Taipei Hsien
Taiwan, R.O.C.
Tel: 886.2.698.4320
Fax: 886.2.698.4133

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