D6351B

Flash TAD Chip for an all Digital Telephone Answering Device with True FULL Duplex SpeakerPhone® and Caller ID Detection

General Description

The D6351B 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 D6351B 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.

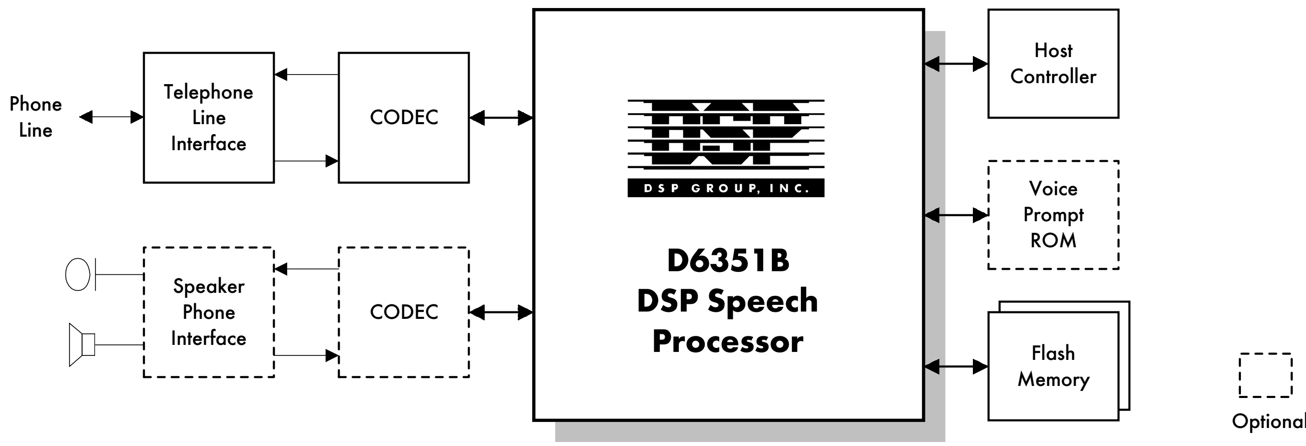

Figure 1. D6351B System Block Diagram

Key Features

- High-quality, variable low rate TRUESPEECH digital speech compression allowing 13-15 minutes of recording time per each 4 Mbit Flash Memory
- Flexible storage of incoming messages (ICM) and outgoing messages (OGM), supporting 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
- Digital volume control
- Supports message tag modification
- Works with 4Mb or 16Mb Flash Memory Devices: Samsung® KM29N040 and KM29N1600T
- Caller ID and CID on Call Waiting (Bell 202 and V.23)
- True FULL Duplex SpeakerPhone with both acoustical and near-end echo cancellation
- Programmable sensitivity of VOX, DTMF, CAS, and Call Progress tone detectors
- Supports "offset playback" for jumps within a message
- FLEXISPEECH™ variable speed, natural sound playback (50%-200%)
- 5V and 3.3V operation
- μ-law and A-law support
- Selectable slave on master mode; 7.2 KHz and 8 KHz sampling mode

Device Configuration and System Components

STANDARD COMPONENTS

- D6351B-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.

- μ-law or A-law serial coder — 1 each, 2 for SpeakerPhone

One of the following:

- Samsung KM29N040 (44-pin TSOP II) 4 Mb per device*, up to four devices per system
- Samsung KM29N1600 (44-pin TSOP II) 16 Mb, single device

* Optionally, the system will support a 32K or 64K x 8 EPROM/ROM (access time 300 ms or less) for Voice Prompt storage. This option is only available in systems with a single 4 Mbit flash memory device.

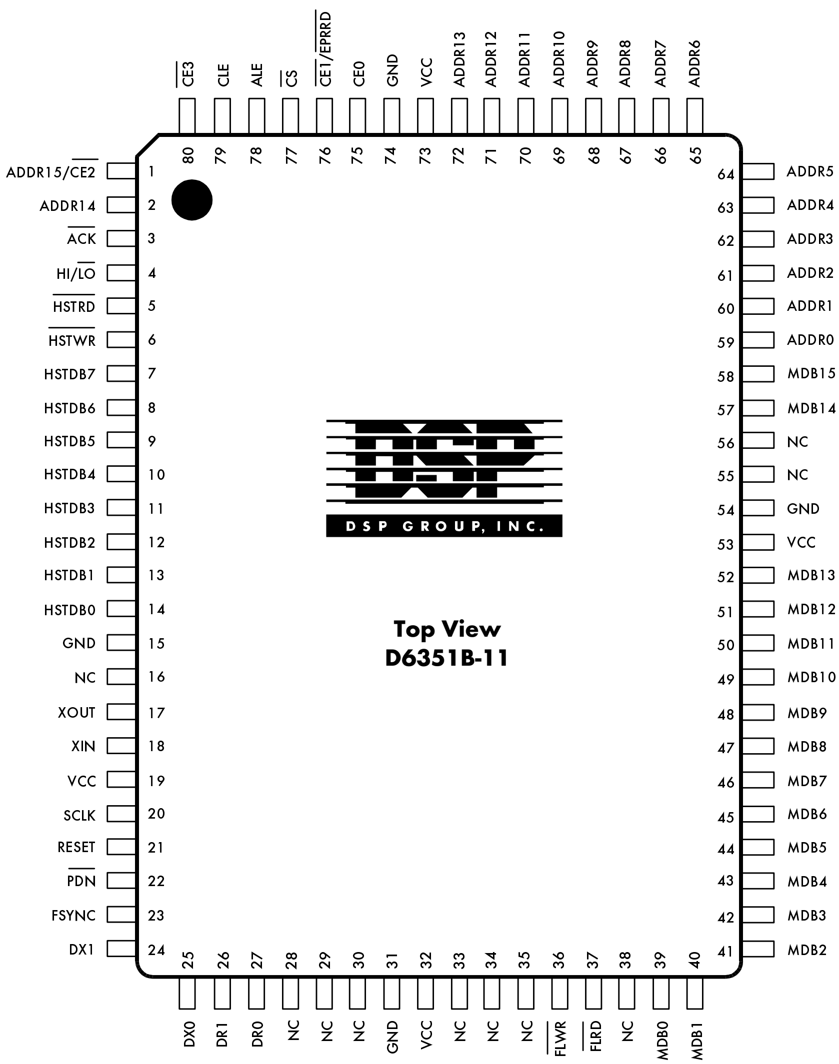

Figure 2. D6351B-11 Pin Diagram

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 is performed by the DSP subsystem which reports the status of the operation to the Host. All memory interface and management is 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.

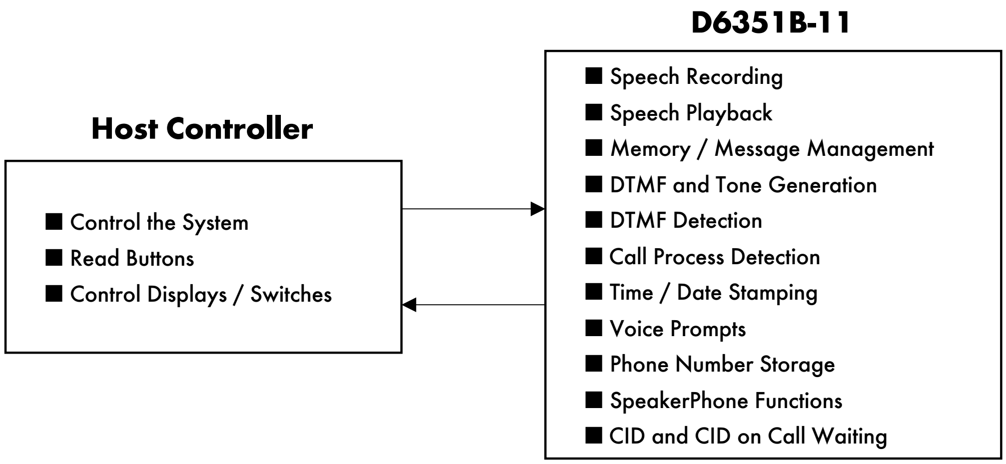

Figure 3. System Functions

Simple Hardware Interface

The hardware interface between the D6351B 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 D6351B, and the D6351B responds with status information. Once a command is issued, the D6351B will use the ACK pin to indicate that the status is available to be read. The hardware interface between the D6351B and Host is shown in Figure 4 below.

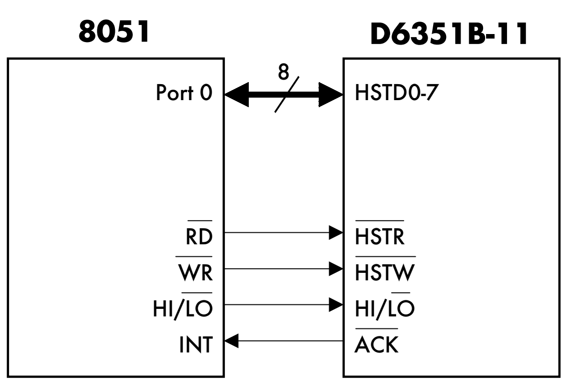

Figure 4. D6351B / Host Interface

Benefits of the D6351B

- High-Quality, variable low rate TRUESPEECH digital speech compression allowing 13-15 minutes of recording time in only 4 Mbits of memory.
- Flash Memory support
  - — Allows use of readily available memory.
  - — Reduces the system cost by eliminating the need for battery back-up in power failure situations.
  - — Allows storage of voice prompts in Flash memory, eliminating the need for external ROM.
- Allows flexibility in design for features such as multiple mailboxes and multiple OGMs, allowing design of a product that is truly a Personal Voice Mail System.
- The Host-selectable thresholds for VOX, DTMF, CAS, and Call Progress detection makes 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 professional sounding speakerphone to be added to your product with a 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.

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