TEXAS INSTRUMENTS

MSP53C691 Speech Synthesizer

User's Guide

December 2000

MSDS Speech

SPSU020

IMPORTANT NOTICE

Texas Instruments and its subsidiaries (TI) reserve the right to make changes to their products or to discontinue any product or service without notice, and advise customers to obtain the latest version of relevant information to verify, before placing orders, that information being relied on is current and complete. All products are sold subject to the terms and conditions of sale supplied at the time of order acknowledgment, including those pertaining to warranty, patent infringement, and limitation of liability.

TI warrants performance of its semiconductor products to the specifications applicable at the time of sale in accordance with TI's standard warranty. Testing and other quality control techniques are utilized to the extent TI deems necessary to support this warranty. Specific testing of all parameters of each device is not necessarily performed, except those mandated by government requirements.

Customers are responsible for their applications using TI components.

In order to minimize risks associated with the customer's applications, adequate design and operating safeguards must be provided by the customer to minimize inherent or procedural hazards.

TI assumes no liability for applications assistance or customer product design. TI does not warrant or represent that any license, either express or implied, is granted under any patent right, copyright, mask work right, or other intellectual property right of TI covering or relating to any combination, machine, or process in which such semiconductor products or services might be or are used. TI's publication of information regarding any third party's products or services does not constitute TI's approval, warranty or endorsement thereof.

Preface

Read This First

About This Manual

This user's guide provides information on the MSP53C691 mixed signal processor. This information includes architecture overview, detailed architecture description, assembly language instruction set, code development tools, and customer information.

How to Use This Manual

This document contains the following chapters:

- Chapter 1—Introduction
- Chapter 2—MSP53C691 Hardware Description
- Chapter 3—MSP53C691 Software Description
- Chapter 4—MSP53C691 Timing Considerations
- Appendix A—Designing the Master Microcontroller Software
- Appendix B—FM Synthesis
- Appendix C—Editing Tools and Data Preparation
- Appendix D—Pitch and Speech Shifting for 6xx MELP
- Appendix E—Guidelines for Optimal TI Speech

Information About Cautions

This book contains cautions.

This is an example of a caution statement.

A caution statement describes a situation that could potentially damage your software or equipment.

The information in a caution is provided for your protection. Please read each caution carefully.

Related Documentation From Texas Instruments

SPSS023B	Data sheet, MSP50C614
SPSU014	User's guide, MSP50C614
SPSS028A	Data sheet, MSP50C604
SPSU014	User's guide, MSP50C604

Contents

Figures

Tables

Chapter 1

Introduction

This chapter briefly describes the features, hardware, and software of the MSP53C691 speech synthesizer.

Topic **Page**

1.1 Description

The MSP53C691 is a standard slave synthesizer from Texas Instruments that accepts compressed speech data from other microprocessors/microcontrollers and converts it to speech. This allows the TI MSP53C691 to be used with a master microprocessor/microcontroller in various speech products, including electronic learning aids, games, and toys. (When referring to the MSP53C691 device in this guide, the terms slave and MSP53C691 are used interchangeably, and the terms master and microcontroller are used interchangeably.)

The MSP53C691 supports several speech synthesis algorithms to permit tradeoffs that meet the price performance requirements of different markets. It also incorporates a single-channel FM synthesis routine for music generation combined with codebook-exited linear-predictive (CELP) coding or mixed-excitation linear prediction (MELP) coding.

The MSP53C691 is a special program that runs on the MSP50C604 device. For more information about the MSP50C604, please refer to the latest version of the MSP50C604 data sheet (literature number SPSS28A) and to the M*SP50x6x User's Guide* (literature number: SPSU014).

1.2 Features

- The device incorporates a wide range of algorithms on one chip. This range allows users to choose from low bit rate to high-quality synthesizing routines for their application. The following algorithms are included:
  - MELP v 3.4 1.0 kbps—3.5 kbps (at 8 Khz sampling rate)
  - CELP v3.4 3.0 kbps, 3.7 kbps, 4.5 kbps, 6.2 kbps, 7.7 kbps, and 11.2 kbps (at 8 kHz sampling rate)
  - ADPCM
  - Single-channel FM synthesis
  - Single-channel FM with CELP or MELP (mixed mode)
- Six-level digital gain control
- Interrupt-driven data transfer for speech or command
- Four customer-configurable I/Os
- Option for four- or eight-bit data bus
  - Eight-bit data bus with four control lines
  - Four-bit data-bus with five control lines
- Low power (less than 10 μA) sleep mode for long battery life
  - Three different sleep modes

- A choice of oscillator control
  - The internal oscillator can be controlled with a 1% resistor for low cost, or a standard 32.768-kHz crystal for higher precision.
- Speed and pitch shifting in MELP
- Stops speaking at any time (only in 4-bit mode)
- Supports sending commands to perform certain tasks while speaking (only in 4-bit mode)
- Operating voltage 3 V–5.2 V
- Direct speaker drive, 32 Ω
- Available in die form or 64-pin LQFP package option

1.3 MSP53C691 Device

The MSP53C691 is optimized to support a four-bit-wide data transfer protocol. The MSP53C691 has two status bits and three control bits which control the communication protocol between the master and the slave. The MSP53C691 also has one bit (data/command) which differentiates between command or speech data feeding into the slave.

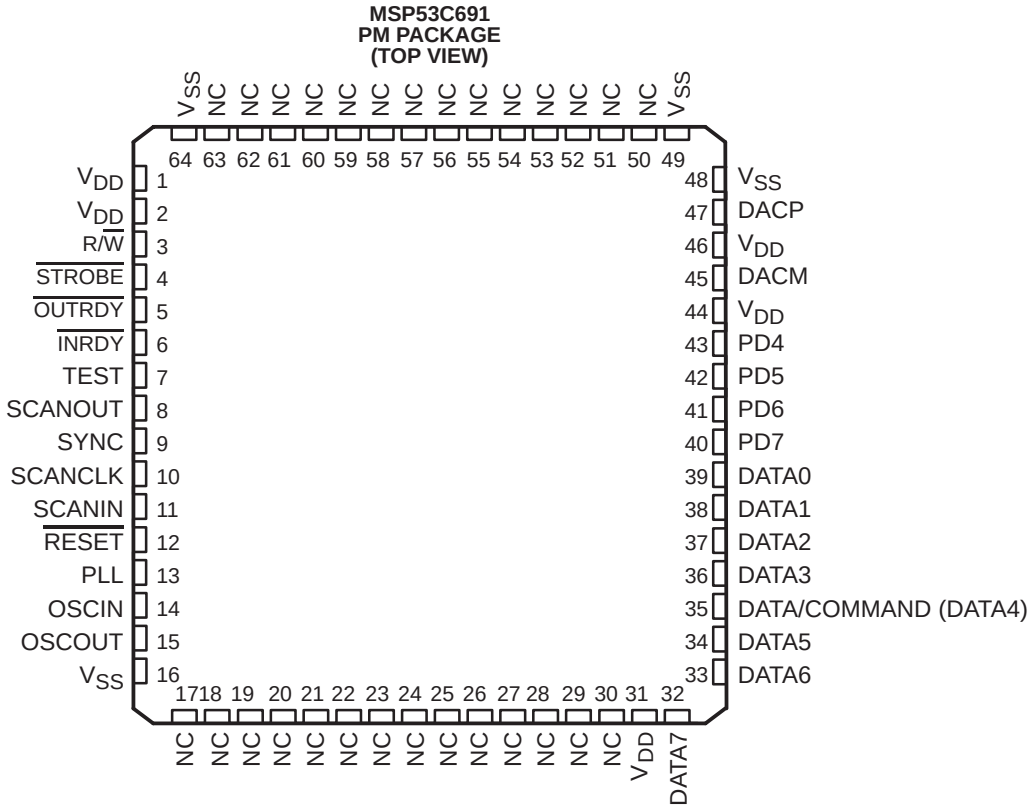
The MSP53C691 also supports the 8-bit wide data transfer but support for commands is disabled. Switching between 4-bit mode or 8-bit mode is permitted between speech data files.

	MSP53C691 (4-bit mode)	**MSP53C691 (8-bit mode)**
Number of data lines	4	8
Number of control lines	3 (strobe, R/$\overline{W}$, data/command)	2 (strobe, R/$\overline{W}$)
Number of status lines	2 (INRDY, OUTRDY)	2 (INRDY, OUTRDY)
Number of general-purpose I/O lines	4	4
Support for commands (while speaking)	Yes	No

1.4 Pin Assignments and Description

Figure 1–1 shows the pin assignments for the MSP53C691. Table 1–1 provides pin functional descriptions.

Figure 1–1. MSP53C691 Pin Assignments



NOTE: Pin 35 is DATA4 in 8-bit mode, or DATA/COMMAND in 4-bit mode.
NC – No internal connection

Table 1–1. MSP53C691 Signal Description

NAME	PIN NO.	PAD NO.	I/O	DESCRIPTION
$\overline{\text{INRDY}}$	6	6	O	An output signal from the slave to the microcontroller. A low signal indicates that the MSP53C691 is ready to accept data or command. A high signal indicates that the MSP53C691 is busy and the microcontroller must not write any data or command to it.
$\overline{\text{OUTRDY}}$	5	5	O	An output signal from the slave to the microcontroller. A low signal indicates that the MSP53C691 is ready to send data or command to the microcontroller.
$\overline{\text{STROBE}}$	4	4	I	An input signal to the slave from the microcontroller. $\overline{\text{STROBE}}$ sequences read or write operations in conjunction with the R/$\overline{\text{W}}$ signal. This signal is pulsed high-low-high for read or write operations sequencing.
R/$\overline{\text{W}}$	3	3	I	An input signal to the slave from the microcontroller. Read/write select signal which is set high for read operations or set low for write operations by the microcontroller.

Table 1–1. MSP53C691 Signal Description (Continued)

NAME	PIN NO.	PAD NO.	I/O	DESCRIPTION
DATA0–DATA3	39–36	25–22	I/O	Data bits 0 through 3 (in 4-bit or 8-bit mode)
DATA4 or DATA/COMMAND	35	21	I/O	Data bit 4 (in 8-bit mode) Data/command control bit (in 4-bit mode). Low signal indicates command and high signal indicates data.
DATA5–DATA7	34–32	20–18	I/O	Data bits 5 through 7 (8-bit mode only) Not used (4-bit mode only)
PD4–PD7	43–40	29–26	I/O	General-purpose I/O bus
Oscillator Reference Signals				
OSCOUT	15	15	O	Output of resistor/crystal oscillator
OSCIN	14	14	I	Input to resistor/crystal oscillator
PLL	13	13	O	Output of phase-lock-loop filter
Scan Port Control Signals‡				
SCANIN	11	11	I	Scan port data input
SCANOUT	8	8	O	Scan port data output
SCANCLK	10	10	I	Scan port clock
SYNC	9	9	I	Scan port synchronization
TEST	7	7	I	C604: test modes
‡NOTE: All these pins must be N.C.				
DAC Sound Output				
DACP	47	33	O	Digital-to-analog output 1 (+) that provides direct speaker drive capability
DACM	45	31	O	Digital-to-analog output 2 (–) that provides direct speaker drive capability
Initialization				
$\overline{\text{RESET}}$	12	12	I	Device initialization
Power Signals†				
V_{DD}	1, 2, 31, 44, 46†	1, 2, 17, 30, 32†	—	Processor power, +5 V nominal supply voltage
V_{SS}	16, 48, 49†, 64	16, 34†, 35, 36	—	Ground pin

† Marked pins are V_{DD} and V_{SS} connections which service the DAC circuitry. These pins tend to sustain a higher current draw. A dedicated decoupling capacitor across these pins is therefore required.

1.5 DAC Information

A two-pin push pull that can directly drive a 32-Ω speaker is used in the MSP53C691. Refer to the *MSP50x6x Mixed Signal Processor Users Guide*, literature number SPSU014, for more information on the D/A and amplifier circuit.

1.6 Algorithms Supported

- MELP: Data rates range from 1 kbps to ~ 3.5 kbps at an 8-kHz sample rate.
- CELP: Data rates can be selected from 3 kbps to 11.2 kbps at an 8-kHz sample rate.
- ADPCM
- FM: frequency modulation for one-channel musical instrument synthesis
- Mix mode: one channel FM synthesis with MELP or CELP

Chapter 2

MSP53C691 Hardware Description

This chapter describes the MSP53C691 hardware, including interface, initialization, and timing.

Topic **Page**

2.1 MSP53C691 Interface Overview

The MSP53C691 interfaces with the master microcontroller either in 4-bit or in 8-bit mode.

The MSP53C691 and the master microcontroller transfer data across four (DATA0–DATA3) or eight (DATA0–DATA7) data lines, depending upon which mode the slave is in.

In either mode the transfer of data is controlled by the two control lines: R/$\overline{W}$ and $\overline{STROBE}$. When the MSP53C691 is ready to receive data from the microcontroller, it sets $\overline{INRDY}$ low. The microcontroller sends data to the MSP53C691 by setting R/$\overline{W}$ low and then pulsing $\overline{STROBE}$ high-low-high. The MSP53C691 latches the data at the rising edge of the $\overline{STROBE}$ pulse. The MSP53C691 also sets $\overline{INRDY}$ high at the rising edge of the $\overline{STROBE}$ pulse. Setting $\overline{INRDY}$ high indicates that the MSP53C691 is not ready to receive any more data.

When the MSP53C691 is ready to send data to the microcontroller, the MSP53C691 sets $\overline{OUTRDY}$ low. The microcontroller responds by setting R/$\overline{W}$ high and then pulsing $\overline{STROBE}$ high-low-high. (The microcontroller latches the data while $\overline{STROBE}$ is low.) This informs the slave that the data has been written to the microcontroller. The MSP53C691 sets $\overline{OUTRDY}$ high at the rising edge of the $\overline{STROBE}$ pulse. Setting $\overline{OUTRDY}$ high indicates that the MSP53C691 does not have data ready to send.

Both 4-bit and 8-bit modes are controlled by commands sent to the slave. A separate bit (DATA 4) is used in 4-bit mode to differentiate between the speech data or command sent to the slave. This line is referred to as data/command line. This is discussed in detail in the software overview section.

2.2 Microprocessor Interface Description

As mentioned in section 2.1, the MSP53C691 interfaces with the master microcontroller either in 4-bit or in 8-bit mode.

2.2.1 4-Bit Mode

The interface between the microcontroller and the MSP53C691 consists of four control lines, two status lines, and four data lines.

The control lines are:

- $\overline{\text{STROBE}}$
- R/$\overline{\text{W}}$
- DATA/COMMAND
- $\overline{\text{RESET}}$

The status lines are:

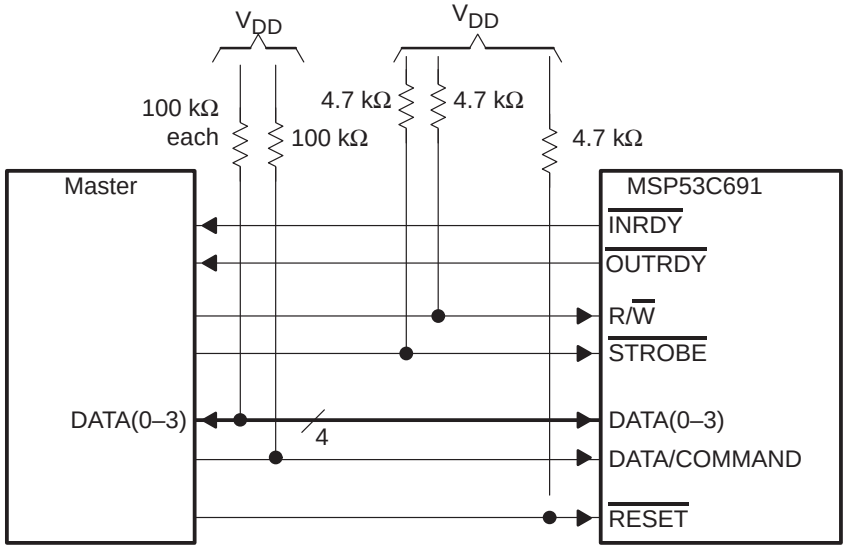
- $\overline{\text{INRDY}}$
- $\overline{\text{OUTRDY}}$

The speech data or command is transferred on lines:

- DATA0 through DATA3

Figure 2–1 shows the interfacing diagram.

Figure 2–1. MSP53C691 Interfacing Diagram—4-Bit Mode



Note:

$\overline{\text{STROBE}}$:	Active low strobe signal from microcontroller
R/$\overline{\text{W}}$	Read/write signal from microcontroller
$\overline{\text{RESET}}$	Active low reset signal from microcontroller
DATA0–DATA3	Data bits 0 through 3
PD4–PD7	General-purpose I/O bus
DACP	Output to speaker/amplifier
DACM	Output to speaker/amplifier
DATA/COMMAND	This bit determines if the data sent by the microcontroller is data or command.

2.2.2 8-Bit Mode

The interface between the microcontroller and the MSP53C691 consists of three control lines, two status lines, and eight data lines.

The control lines are:

- $\overline{\text{STROBE}}$
- R/$\overline{\text{W}}$
- $\overline{\text{RESET}}$

The status lines are:

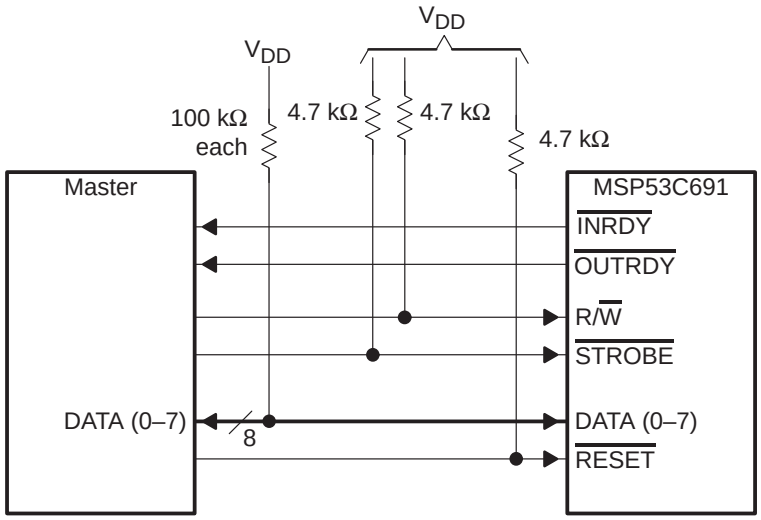
- $\overline{\text{INRDY}}$
- $\overline{\text{OUTRDY}}$

The data/command lines are:

- DATA0 through DATA7

Figure 2–2 shows the interfacing diagram.

Figure 2–2. MSP53C691 Interfacing Diagram—8-Bit Mode



Note:

$\overline{\text{STROBE}}$:	Active low strobe signal from microcontroller
R/$\overline{\text{W}}$:	Read/write signal from microcontroller
$\overline{\text{RESET}}$	Active low reset signal from microcontroller
$\overline{\text{INRDY}}$:	Active low indicates that the MSP53C691 is ready to accept data.
$\overline{\text{OUTRDY}}$:	Active low indicates that the MSP53C691 is ready to send data.
DATA0–DATA7	Data bits 0 through 7
PD4–PD7	General-purpose I/O bus
DACP	Output to speaker/amplifier
DACM	Output to speaker/amplifier

2.3 Read Operation by the Master

The process for the read operation by the master is the same in 4-bit and 8-bit modes. The read operation by the master happens when the slave wants to send something to the master. The slave initiates the read process by pulling $\overline{\text{OUTRDY}}$ low when the slave is ready.

The following events take place during the read operation:

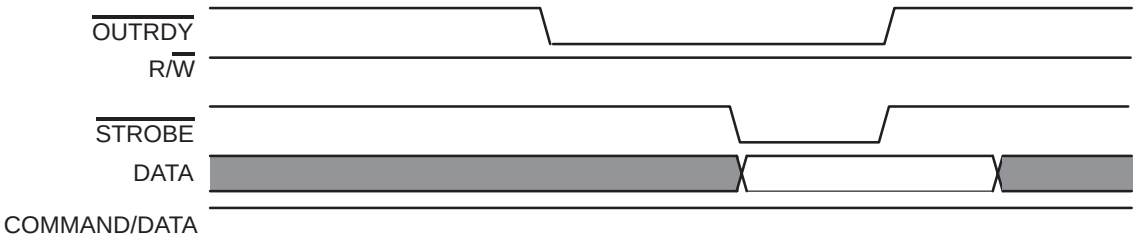
1) The MSP53C691 puts the data to be sent to the master on the internal bus.
2) The MSP53C691 sets $\overline{\text{OUTRDY}}$ low to indicate that it is ready to send data to the microcontroller.
3) The microcontroller sets R/$\overline{\text{W}}$ high to indicate a read operation.
4) The microcontroller sets $\overline{\text{STROBE}}$ low. The data is available on the external data-bus at this point.
5) The microcontroller reads the data from the bus.
6) The microcontroller sets $\overline{\text{STROBE}}$ high. The MSP53C691 also pulls $\overline{\text{OUTRDY}}$ high at the rising edge of $\overline{\text{STROBE}}$.
7) The data is taken off from the external data-bus after $\overline{\text{STROBE}}$ goes high.

The microcontroller must latch or read in the data while $\overline{\text{STROBE}}$ is low. When the microcontroller sets $\overline{\text{STROBE}}$ high, the MSP53C691 sets $\overline{\text{OUTRDY}}$ high to indicate that the data has been successfully transferred.

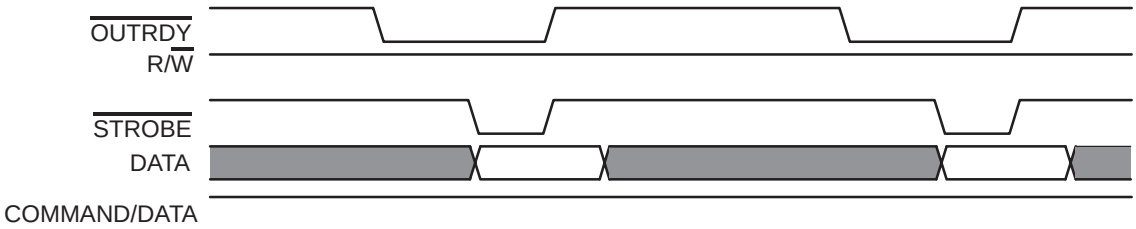
Figure 2–3 shows the sequence of events of the read operation.

Figure 2–3. Data Transfer—Read

a) Sequence of events for a single read operation:



b) Read—Two speech data transfer sequences:



2.4 Write Operation by the Master

The process for the write operation by the master is the same in 4-bit and 8-bit modes. The write operation by the master happens when the slave is ready to request data or command from the master. The slave initiates the write process by pulling $\overline{\text{INRDY}}$ low when the slave is ready to receive data.

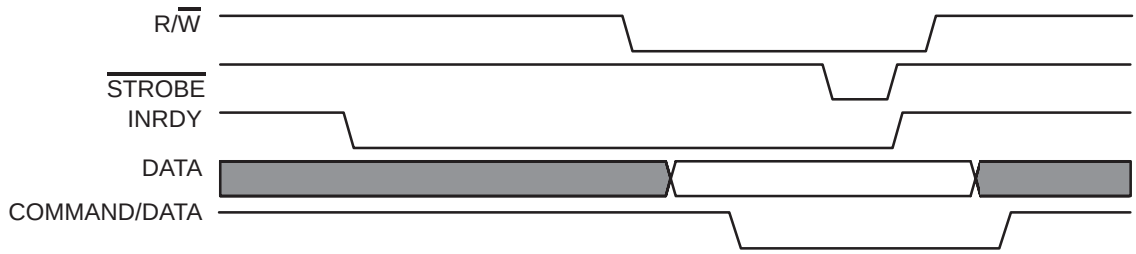
The following events take place during the write operation:

1) The MSP53C691 sets $\overline{\text{INRDY}}$ low to indicate that it is ready to receive data from the microcontroller.
2) The microcontroller sets R/$\overline{\text{W}}$ low to indicate a write operation.
3) The microcontroller puts the data in the external data-bus.
4) The microcontroller sets $\overline{\text{STROBE}}$ low after the data is valid.
5) The microcontroller sets $\overline{\text{STROBE}}$ high after a minimum of 300 ns. The MSP53C691 also pulls $\overline{\text{INRDY}}$ high at the rising edge of $\overline{\text{STROBE}}$.
6) The data is latched in the MSP53C691 at the rising edge of $\overline{\text{STROBE}}$.

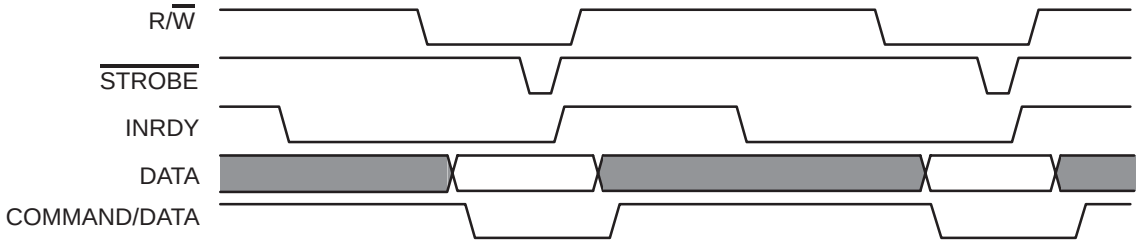
When the microcontroller sets $\overline{\text{STROBE}}$ high, the MSP53C691 sets $\overline{\text{INRDY}}$ high to indicate that the MSP53C691 is not ready to receive any more data.

Figure 2–4. Data Transfer—Write

a) Sequence of events for a single write operation



b) Write—Two speech data transfer sequences



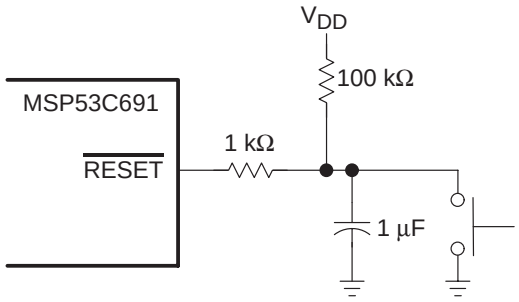
2.5 MSP53C691 Device Initialization

The $\overline{\text{RESET}}$ pin is configured as an external interrupt (see Figure 2–5). It provides the means for hardware initialization of the device. When the $\overline{\text{RESET}}$ pin is held low, the device assumes a deep sleep state and various registers are initialized. After the $\overline{\text{RESET}}$ pin is taken high, the device gets initialized and pulls $\overline{\text{INRDY}}$ low when the slave is ready to receive the oscillator selection command (see the *Software Description* chapter for a description of the oscillator selection command). The oscillator selection command (0xB) is sent through the DATA0–3 line while setting the DATA4 line low (indicating that it is a command). When the slave receives the oscillator selection command (0xB), it pulls $\overline{\text{INRDY}}$ low again to request the parameter (0x1 or 0x2) for the command. After the parameter is received by the slave, it goes through the rest of the initialization process and pulls $\overline{\text{INRDY}}$ low when the oscillator is stabilized and the slave is ready to receive data/command.

Note:

If the slave does not receive the oscillator selection command as the first command when $\overline{\text{INRDY}}$ goes low for the first time, or if the slave does not receive the subsequent parameters for the oscillator control command (0x1 or 0x2), the slave resets itself again.

Figure 2–5. MSP53C691 RESET Diagram



The MSP53C691 is considered to be properly initialized after the following events take place:

1) The microcontroller sets $\overline{\text{RESET}}$ low.
2) The microcontroller sets $\overline{\text{STROBE}}$ high throughout the initialization process.
3) The microcontroller sets $\overline{\text{RESET}}$ high.
4) The microcontroller waits for $\overline{\text{INRDY}}$ to go low.
5) The microcontroller sets R/$\overline{\text{W}}$ low.
6) The microcontroller puts 0xB (oscillator control command) on the data-bus (DATA0–3) and sets the DATA4 (DATA/COMMAND) line low.
7) The microcontroller pulls $\overline{\text{STROBE}}$ low, waits for a minimum of 300 ns, then pulls $\overline{\text{STROBE}}$ high again.

8) The slave latches the command on the rising edge of $\overline{\text{STROBE}}$.

9) The slave pulls $\overline{\text{INRDY}}$ low again to request the parameter for the oscillator control command.

10) The master sets R/$\overline{\text{W}}$ low.

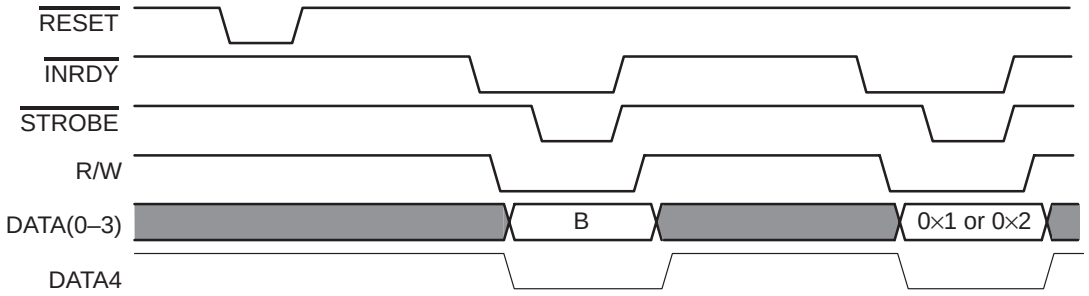
11) The master puts the command parameter for the oscillator control command (0x1 or 0x2, see Table 3–1 for details) on the data-bus (DATA0–DATA3) and sets the DATA4 (DATA/COMMAND) line low.

12) The microcontroller pulls $\overline{\text{STROBE}}$ low, waits for a minimum of 300 ns, then pulls $\overline{\text{STROBE}}$ high again.

13) The slave latches the command parameter on the rising edge of $\overline{\text{STROBE}}$.

14) The slave completes the rest of the initialization process and pulls $\overline{\text{INRDY}}$ low when ready to receive new command. See Figure 2–6 for a timing diagram of the initialization process.

Figure 2–6. Device Initialization

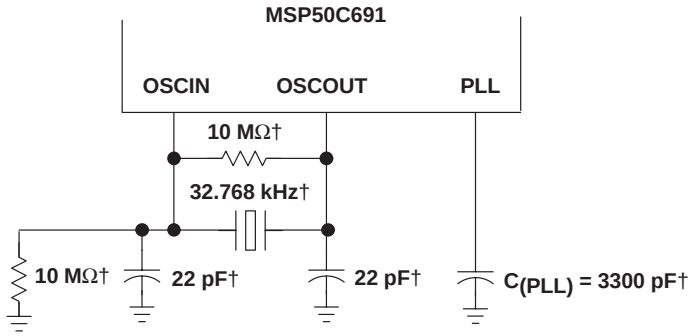


2.6 Microprocessor Interface Timing

The MSP53C691 has a self-contained clock generation system. This flexible clock generation system enables the software to control the clock over a wide frequency range. The implementation uses a phase-locked-loop (PLL) circuit that drives the processor clock to a selectable frequency between the minimum and maximum ranges. Selectable frequencies for the processor clock are spaced by 65.536 kHz. The PLL clock-reference is also selectable. between a resistor-trimmed oscillator or a crystal-referenced oscillator, see Figure 2–2. Internal and periphery clock sources are controlled separately to provide different levels of power management. Figures 2–3 and 2–4 illustrate the timing diagram for write and read operations.

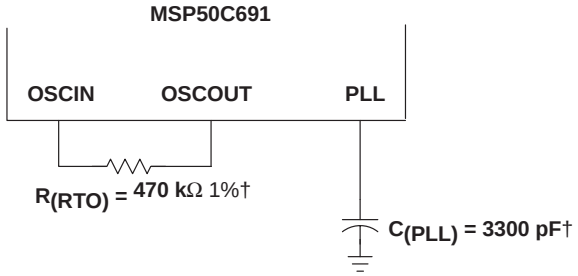
Figure 2–7. Oscillator and PLL Connection

a) Crystal Oscillator Operation Connections



† Keep these components as close as possible to the OSC_{IN}, OSC_{OUT}, and PLL pins.

b) Resistor Trim Operation Connections



† Keep these components as close as possible to the OSC_{IN}, OSC_{OUT}, and PLL pins.

Chapter 3

MSP53C691 Software Description

This chapter overviews the software and describes the commands used to program the MSP53C691.

Topic **Page**

3.1 Software Overview

The MSP53C691 is a slave device that is controlled using a formatted communications sequence that transfers commands and data streams from the master microcontroller. The commands are combined at the top level into several main groups (command header). Each main group has sublevels (level 1 and level 2) that may require several parameters, depending upon the group.

3.2 Commands and Data Streams

Two types of communications are sent to the MSP53C691: speech data streams and commands (with parameters). The command headers and the level 1 and level 2 parameters are always sent to the MSP53C691 a nibble at a time through data lines DATA0–DATA3. However, the speech data is sent over the four-bit data lines (DATA0–DATA3) in four-bit mode, while speech data is sent over the eight-bit data lines (DATA0–7) in eight-bit mode. When sending speech data in four-bit mode, the DATA4 (DATA/COMMAND) line is always pulled high to differentiate speech data from a command. The DATA4 (DATA/COMMAND) line is always pulled low while sending commands to the MSP53C691. In four-bit mode, commands can be sent to the MSP53C691 at the beginning of a speech file or at any other time while speaking. In eight-bit mode, commands can only be sent to the MSP53C691 at the beginning of each speech file. Support for commands while speaking a phrase is not available in eight-bit mode; it is only available after speaking of a phrase is complete.

- The DATA/COMMAND line defines what type of information is being sent (data or command) to the MSP53C691. The command or speech data sent to the MSP53C691 is distinguished by the following state of the DATA/COMMAND line.
  - Data = 1
  - Command = 0
- A data stream or speech data is sent to the MSP53C691 to provide the data file for speaking.
- Command codes are sent to the MSP65C691 to control functions such as speaking start or stop, toggle I/O, volume control, oscillator control, configure internal registers, etc. See Table 3–1 for a complete list of command codes.

Note:

Throughout the rest of the chapter it is assumed that data from the MSP53C691 is always read four nibbles at a time. The MSP53C691 pulls $\overline{\text{OUTRDY}}$ low when it wants to send a nibble to the master (read from the MSP53C691). It is also assumed that the data is sent to the MSP53C691 from the master only when the MSP53C691 pulls $\overline{\text{INRDY}}$ low. In four-bit mode, the DATA4 (DATA/COMMAND) bit is low when a command or its parameters are written to the MSP53C691, and the DATA4 (DATA/COMMAND) bit is high when the speech or FM data is written to the MSP53C691.

While sending speech or command in four-bit mode, the data is always sent most-significant nibble first and least-significant nibble last.

All commands and their parameters are sent with a four-bit interface. Commands and their parameters are sent to the slave through data lines DATA0–DATA3 while DATA4 (DATA/COMMAND) line is set low.

The procedures for reading from or writing to the MSP53C691 are described in the *Hardware Description* chapter.

3.3 Sequence of Command Codes and Data Streams

The sequence for sending the command codes and data streams is as follows:

- The sequence for the command codes consists of a command header, a level 1, and, sometimes, a level 2 parameter. The command header controls the operation performed by the MSP53C691. Level 1 and level 2 are two levels of command information that further define the command code. Each nibble of the command code is sent along with a 0 in the DATA/COMMAND line to indicate that it is a command. Refer to section 3.3.2, *Parameters* and Table 3–1, *Command Codes* for additional information.
- Data streams are speech data sent to the MSP53C691 from speech files processed by the SDS6000 (see Appendix C for details). Each nibble of the speech data is sent along with a 1 on the DATA/COMMAND line to indicate that it is speech data.

3.3.1 Command Header

The command header is the first information of the command code sequence sent to the MSP53C691. The command header controls the operation being performed by the MSP53C691. The MSP53C691 recognizes the following command headers:

- Command header 1—Configure internal registers
- Command header 2—Set/Clear I/O ports PD4 through PD7
- Command header 3—Read contents of I/O ports
- Command header 4—Start speaking
- Command header 5—Stop speaking

- ❑ Command header 6—Adjust volume
- ❑ Command header 7—Return status of data buffers
- ❑ Command header 8—Initiate sleep mode
- ❑ Command header 9—Receive FM data (while speaking in FM only or in mix mode)
- ❑ Command header A—Perform speed/pitch shift
- ❑ Command header B—Set up oscillator

The command headers are sent to the MSP53C691 in four-bit nibbles with a 0 in the DATA4 line in either four-bit or eight-bit mode.

CAUTION

Support for sending commands while speaking is not available in eight-bit mode. In eight-bit mode, commands can be sent to the slave between speaking phrases.

3.3.2 Parameters

For the command codes, there are two types of parameters that are sent to the MSP53C691 after sending the command header: level 1 and, if required, level 2. Level 1 and level 2 are levels of command information that further defines the command code. For example, if the command header is *Adjust the Volume* (one nibble command code 0x6), then the level 1 parameter (one nibble, 0x1 through 0x6) defines what level the volume is being set to (that is, low, medium, or high). Also, if the command header is *Configure Internal Registers* (one nibble command code 0x1), then the level 1 parameter defines which internal register address to write to (two nibbles specifying the address of the register). The level 2 parameter is the data written to the internal register (four nibbles specifying the word to be written in the register). All the command headers and Level 1 and level 2 parameters are sent to the MSP53C691 a nibble at a time with the DATA4 (DATA/COMMAND) bit set low. While interleaving commands between speech data, it is advisable to send all the nibbles of the command (command header, level 1, and level 2 parameters, if applicable) before sending more speech data or commands. Refer to Table 3–1, *Command Codes*, for additional information.

3.3.3 Return Values

The MSP53C691 can return values to the microcontroller when required. The MSP53C691 sends return values to the master either to respond to a command, to return error codes, or to provide its current status. The values returned vary in content, depending on the command code initiated and on the status of the MSP53C691. The following is a list of returned values from the MSP53C691:

Returned Values	Description
0x0055	Returned when the MSP53C691 successfully plays a speech file. In mix mode, 0x0055 is returned when the MSP53C691 has finished playing both the FM and the speech file.
0x0054	Returned in mix mode when the MSP53C691 has successfully finished playing a speech file and is waiting to start a new file or for the FM to finish playing.
0x0053	Returned in mix mode when the MSP53C691 has successfully finished playing an FM file and is waiting for the speech file to finish playing.
0x9999	Returned when an error is encountered. This error is returned when either: • An invalid command header or and invalid level 1 or level 2 parameter is issued by the master. • An invalid port address to be written to or to be read from is specified. • An invalid file format is encountered while playing multiple files In mix mode.
0x1234	Returned when there is an error recognizing the header byte in the speech file or the stack is overflowed
0x0N23	Returned (in mix mode or while speaking FM only) when the FM buffer becomes nearly empty and FM data is needed to fill up the FM buffer. N denotes the number of FM bytes needed to fill up the FM buffer.

CAUTION

The MSP53C691 waits until all of the four nibbles have been sent to the microcontroller. If the microcontroller delays in reading the data from the MSP53C691, the speech operation can be interrupted. Therefore, the microcontroller must read the data as soon as the $\overline{\text{OUTRDY}}$ goes low.

3.4 Command Codes

The valid command codes are described in Table 3–1 and paragraph 3.5. Command codes include the command header (one nibble), level 1 parameters (one or two nibbles), and, sometimes, level 2 (four nibbles) parameters. As shown in Table 3–1, there are occasions when values are returned to the slave as a result of the command that is initiated. These return values are always four nibbles in size and are sent over the DATA0–DATA3 data lines.

In four-bit mode, the commands (command header and level 1 and level 2 parameters) are sent to the MSP53C691 four bits (a nibble) at a time with DATA4 set to 0 for every transmitted nibble. In four-bit mode, commands can be sent while speaking a phrase, but in eight-bit mode commands can only be sent between speaking phrases. After receiving the command to speak in 8-bit mode, the slave expects speech data in 8–bits. No other command can be sent to the slave during speaking. The slave goes to the 4-bit mode to receive

further commands after finishing speaking (either in 4-bit or in 8-bit mode). All commands require at least one nibble of level 1 parameter. There are some commands that require more than one nibble of level 1 parameter. Some commands require more than one nibble of level 2 parameter. Some commands return four nibbles back to the master in response. For example, a return buffer status (command code 0x7) followed by a level 1 parameter (0x1 for non-FM buffer) returns four nibbles reporting the number of bytes left to fill up the buffer back to the master.

See section 3.5, Description of the Command Codes, for a description of each command.

Table 3–1. Command Codes

Command Header	Parameters Level 1	Parameters Level 2	Return Values	Description
Configure Internal Registers				
	Port address	Value to be written		
1	N_1N_2	$N_1N_2N_3N_4$	None	Write to internal registers for configuration
Set/Clear I/O Ports, PD4 Through PD7				
	PD4–PD7			
2	N	None	None	Upper four bits for port D (Note: Configuration of port D4–7 as output is required)
Read Contents of I/O Ports				
	Port address		Value	
3	N_1N_2	None	$N_1N_2N_3N_4$	Returns the port value
Start Speaking				
4	1	None	None	Speak CELP, MELP, or ADPCM in four-bit data transfer
4	2	None	None	Speak CELP, MELP, or ADPCM in eight-bit data transfer
4	3	None	None	Test mode
4	4	none	None	Play FM (Music) in four-bit mode only
4	5	none	None	Speak mixed mode in four-bit mode only (FM + CELP/MELP)
4	6	None	None	Speak multiple phrases in mix mode—see the details under the discussion of the command
4	7	None	None	Play sinewave in test mode at 1 kHz
Stop Speaking				
5	1	none	None	Stop speaking all (in mix or nonmix mode)
5	2	None	None	Stop speaking CELP/MELP only in mix mode
5	3	None	None	Stop speaking FM only in mix mode

Note: Each nibble is sent to the master with DATA4 bit set to 0, indicating that it is a command.
N—Represents one nibble.
$N_1N_2N_3N_4$—Represents four nibbles, with N_1 being the first nibble sent and N_4 being the 4th nibble sent (MSB and LSB).
The numbers in the box represent the actual value of the nibble sent to the MSP53C691.

Table 3–1. Command Codes (Continued)

Adjust the Volume				
6	1	none	none	Level 1 (lowest volume)
6	2	none	none	Level 2
6	3	none	none	Level 3
6	4	none	none	Level 4
6	5	none	none	Level 5
6	6	none	none	Level 6 (highest volume)
Command Header	**Parameters Level 1**	**Level 2**	**Return Values**	**Description**
Return Status of Data Buffers				
7	1	None	$N_1N_2N_3N_4$	Returns the number of bytes required to fill up the CELP/MELP buffer
7	2	None	$N_1N_2N_3N_4$	Returns the number of bytes required to fill up the FM buffer
Initiate Sleep Mode				
8	1	None	None	Light sleep (see Table 2–3 of SPSU014)
8	2	None	None	Mid sleep (see Table 2–3 of SPSU014)
8	3	None	None	Deep sleep (see Table 2–3 of SPSU014) Wake-up from sleep mode can be performed by putting dummy data in the bus and pulsing $\overline{\text{STROBE}}$
Receive FM Data				
9	N	None	None	Receive FM data N = number of FM bytes the master is going to send
Perform Speed/Pitch Shift				
A	0	$N_1N_2N_3N_4$	None	Slow down MELP
A	1	$N_1N_2N_3N_4$	None	Speed up MELP
A	2	$N_1N_2N_3N_4$	None	Pitch shift in MELP
A	3	None	None	Reserved (TI test code)
Oscillator Control				
B	1	None	None	Crystal mode
B	2	None	None	Resistor trim mode

Note: Each nibble is sent to the master with DATA4 bit set to 0, indicating that it is a command.
N—Represents one nibble.
$N_1N_2N_3N_4$—Represents four nibbles, with N_1 being the first nibble sent and N_4 being the 4th nibble sent.
The numbers in the box represent the actual value of the nibble sent to the MSP53C691.

3.5 Description of the Command Codes

3.5.1 Command Header 1—Configure Internal Registers

This command is used to configure the MSP53C691 internal registers. Certain values in certain registers are not allowed to be modified (see note at the end of this section). Attempting to modify these values when writing to those registers has no effect. The command header for configuring internal registers for the MSP53C691 is one nibble long and of value 1 as shown. This header is sent across the four data lines. The sequence of events to configure the internal registers is as follows:

1) The MSP53C691 pulls $\overline{\text{INRDY}}$ low to request a command or data.

2) The master sends the command header 0x1 over the data lines.

Write Operation to Configure Internal Registers				
Data Lines				
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**
0	0	0	0	1

3) The MSP53C691 pulls $\overline{\text{INRDY}}$ low again to request level 1 parameters. The level 1 parameter that defines the address of the internal register is two nibbles long in this case, see the following examples:

Most significant (first nibble)	Least significant (Second nibble)
X	X

4) The master sends the two nibbles back to back in response to the two consecutive lowerings of $\overline{\text{INRDY}}$.

Note:
When sending the two nibbles that make up the address of the register, send the first nibble (most significant) first.

Example 1: To configure pins 7 and 5 of port D as outputs and pins 6 and 4 as inputs, we need to write to the port D control register. The address for the register to be configured is 0x1C, then the two nibbles (8 bits) are written as 1C:

MSB							**LSB**
7	6	5	4	3	2	1	0
0	0	0	1	1	1	0	0

Perform the following write operation by writing the two nibbles (8 bits) MSB first, as shown. The two nibbles are written one nibble at a time—in each case the DATA4 bit is pulled low, indicating that a command is being sent.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	0	0	0	1	Firstst nibble sent
Least Significant					
0	1	1	0	0	Second nibble sent

5) After the level 1 parameters are received, the MSP53C691 pulls $\overline{\text{INRDY}}$ low again requesting the data to be written to the registers. The data is 16 bits long and is transferred to the MSP53C691 a nibble at a time (most-significant nibble first). The master must send four nibbles (for 16-bit data) back to back in response to four consecutive lowerings of $\overline{\text{INRDY}}$.

Note:
The data to be written to the configuration registers is always 16 bits long. If there are less than 16 bits to be written to the registers, then there should be leading zeroes to make it 16 bits long.

Example 2: The value to be written to the internal register is 0xA0, then the four nibbles (16 bits) are written as 0x00A0:

MSB															**LSB**
15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
0	0	0	0	0	0	0	0	1	0	1	0	0	0	0	0

Perform the following write operation by writing the four nibbles (16 bits) MSB first, as shown. The four nibbles are written one nibble at a time with the DATA4 line set to 0, indicating that a command parameter is being sent.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	0	0	0	0	First nibble sent
0	0	0	0	0	Second nibble sent
0	1	0	1	0	Third nibble sent
Least Significant					
0	0	0	0	0	Fourth nibble sent

NOTE:

- Do not write to the following addresses:
  - Port A
    - Data, address 0x00
    - Control, address 0x04

- Port B
  - Data, address 0x08
  - Control, address 0x0C
- Port C
  - Data, address 0x10
  - Control, address 0x14
- Port D (PD0 – PD3)
  - Data, address 0x18
  - Control, address 0x1C
- Port E
  - Data, address 0x20
  - Control, address 0x24
- Port F
  - Data, address 0x28
- Port G
  - Data, address 0x2C
- Timer 1 period and countdown register, address 0x3A and 0x3B
- Timer 2 period and countdown register, address 0x3E and 0x3F

Note: Refer to Table 2–2 in SPSU014 for a complete listing of the port addresses.

3.5.2 Command Header 2—Set/Clear I/O Ports, PD4 Through PD7

This command can be used to pull a general-purpose I/O pin low or high. The PD4–7 pins are defined as general-purpose I/O pins in the MSP53C691. Before sending the command, it is necessary to configure the port pins to the appropriate state (INPUT or OUTPUT) by configuring the internal control registers. This command causes the following sequence of events:

1) The MSP53C691 pulls $\overline{\text{INRDY}}$ low requesting command or data. The master sends the command header 0x2 for toggling I/O ports PD4 through PD7 of the MSP53C691 as shown in the following table:

WRITE OPERATION TO TOGGLE I/O PORTS				
DATA LINES				
DATA4	DATA3	DATA2	DATA1	DATA0
0	0	0	1	0

2) The MSP53C691 pulls $\overline{\text{INRDY}}$ low again to receive the level 1 parameter. The level 1 parameter for this command is only one nibble long and is effectively written to the upper four bits of the port (PD4–PD7). This one-nibble parameter defines which pins are being toggled (see examples below).

Example 1: If PD5 is toggled high, then the one nibble level 1 parameter is written as 0010; this sets PD5 high.

PINS			
PD7	PD6	PD5	PD4
0	0	1	0

Example 2: If none of the pins are toggled high, then the one nibble parameter is written as 0000; this sets PD4 through PD7 low.

PINS			
PD7	PD6	PD5	PD4
0	0	0	0

Note:
Port D must be set as output mode first with the configure register command for the toggle I/O command to work properly.

3.5.3 Command Header 3—Read Contents of I/O Ports

This command can be used to read back the contents of a port. The command header for this command is one nibble long and the level 1 parameter specifying the address of the port to be read is two nibbles long. This command causes the following sequence of events:

1) The MSP53C691 pulls $\overline{\text{INRDY}}$ low to request a command or data.

2) The master sends the command header 0x3 to read the port as follows:

WRITE OPERATION FOR READ CONTENTS OF I/O PORTS				
DATA LINES				
DATA4	DATA3	DATA2	DATA1	DATA0
0	0	0	1	1

3) The MSP53C691 pulls $\overline{\text{INRDY}}$ low again to receive the level 1 parameters. The level 1 parameter is two nibbles long and defines the address of the port, see the following example:

Most significant (first nibble)	Least Significant (second nibble)
X	X

4) The master sends the two nibbles in response to the two consecutive lowerings of $\overline{\text{INRDY}}$.

Note:
When sending the two nibbles, send first nibble (most-significant nibble) first.

Example 1: To read port D pins 7–4, we need to read the port D data register. The address for the port to be read is 0x18, then the two nibbles (8 bits) are written as 0x18:

MSB							LSB
7	6	5	4	3	2	1	0
0	0	0	1	1	0	0	0

Perform the following write operation by writing the two nibbles (8 bits) MSB first, as shown. The two nibbles are written one nibble at a time with the DATA4 line pulled low in each case, indicating that a command is being sent.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	0	0	0	1	First nibble sent
Least Significant					
0	1	0	0	0	Second nibble sent

5) After the command header and all the parameters are received, the command is executed by setting $\overline{\text{OUTRDY}}$ low and the four nibbles of data contained in the port register are sent (MSB first, LSB last) to the microcontroller.

6) After receiving the level 1 command parameter (port address), the MSP53C691 pulls $\overline{\text{OUTRDY}}$ low four times to return the contents of the port to the master. Each lowering of $\overline{\text{OUTRDY}}$ transfers a nibble (DATA0–DATA3) to the master—the first being the most-significant nibble.

CAUTION

The MSP53C691 waits until all of the four nibbles have been sent to the microcontroller. If the microcontroller delays in reading the data from the MSP53C691, the speech operation can be interrupted. Therefore, the microcontroller must read the data as soon as the $\overline{\text{OUTRDY}}$ goes low.

7) The MSP53C691 pulls $\overline{\text{INRDY}}$ low again to request new data or command.

Note: The contents of the following port addresses can be read:

- Port A
  - Data register (address 0x00)
  - Control register (address 0x04)

- Port C
  - Data register (address 0x10)
  - Control register (address 0x14)
- Port D
  - Data register (address 0x18)
  - Control register (address 0x1C)
- DAC
  - Data register (address 0x30)
  - Control register (address 0x34)
- IntGenCtrl register (address 0x38)
- Interrupt flag register (address 0x39)

Note: Refer to Table 2–2 in SPSU014 for a complete listing of the port addresses.

3.5.4 Command Header 4—Start Speaking

This command is used to command the MSP53C691 to start speaking. This command has several level 1 parameters to describe the mode in which the MSP53C691 speaks. The command header is one nibble long and is written 0100 as shown in the following table. The DATA4 bit is low while sending this nibble.

WRITE OPERATION FOR START SPEAKING				
DATA LINES				
DATA4	DATA3	DATA2	DATA1	DATA0
0	0	1	0	0

The command header 0100 is followed by a level 1 parameter that is one nibble long and instructs the MSP53C691 to perform the functions defined in the following table:

DATA LINES					FUNCTIONS
DATA4	DATA3	DATA2	DATA1	DATA0	
0	0	0	0	1	Speak CELP, MELP, or ADPCM in four-bit data transfer mode
0	0	0	1	0	Speak CELP, MELP, or ADPCM in eight-bit data transfer mode
0	0	0	1	1	Test mode (nibble in = nibble out)
0	0	1	0	0	Play FM (music) in four-bit data transfer mode
0	0	1	0	1	Speak mixed mode in four-bit data transfer mode (FM + CELP/MELP)
0	0	1	1	0	Speak multiple files in mix mode

The different modes of speaking are described in detail in the following subsections.

3.5.4.1 Speaking CELP, MELP, ADPCM in Four-Bit Data Transfer Mode

Commands and data are sent over a four-bit interface. When data is sent over the DATA0 through DATA3 lines, DATA4 (pin 35) becomes the DATA/COMMAND control line. The DATA/COMMAND line defines whether the information on DATA0 through DATA3 is data or commands. When DATA0 through DATA3 and DATA/COMMAND bits are received, the MSP53C691 analyzes the DATA/COMMAND bit to determine if the information is data or commands being received on DATA0 through DATA3 lines.

DATA/COMMAND LINE	DESCRIPTION
Set to 0	Command sent to the MSP53C691
Set to 1	Data sent to the MSP53C691

The following typical sequence of events takes place while playing CELP, MELP, or ADPCM in four-bit mode:

1) The MSP53C691 pulls $\overline{\text{INRDY}}$ low and waits for a command from the master.

2) The master sends the command header 0x4 (command header for speaking in four-bit mode) on the data bus. Note that writing 0x04 to the data bus ensures that the DATA4 line is low, indicating that value on data lines DATA0–DATA3 is a command.

Most Significant					
DATA LINES					
DATA4	DATA3	DATA2	DATA1	DATA0	
0	0	1	0	0	First command header sent

3) After receiving the header, the MSP53C691 pulls $\overline{\text{INRDY}}$ low again and waits for the level 1 parameter from the master.

4) The master sends the level 1 command parameter 1 to indicate the four-bit transfer mode for CELP, MELP, or ADPCM.

Most Significant					
DATA LINES					
DATA4	DATA3	DATA2	DATA1	DATA0	
0	0	0	0	1	Level 1 parameter sent

5) The MSP53C691 interprets the command and initializes itself to get ready to speak.

6) The MSP53C691 pulls $\overline{\text{INRDY}}$ low, indicating that it is ready to receive more commands or speech data.

7) The master can now start sending speech data from a file in response to each lowering of $\overline{\text{INRDY}}$. Note that, while sending speech data, the DATA4 line must be high, indicating that speech data is being sent. For example, to send the speech data byte 0x24 to the MSP53C691, the master must put 0x12 on the data bus in response to the lowering of $\overline{\text{INRDY}}$. When the MSP53C691 is again ready to receive the next nibble, the master puts 0x14 on the data bus again.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
1	X	X	X	X	First nibble sent
1	X	X	X	X	Second through X nibble sent
Least Significant					
1	X	X	X	X	Last nibble sent

8) The master can send a command (such as volume control)-between the speech data while sending speech data. In this case the master first sends the command header nibble of the command in response to $\overline{\text{INRDY}}$ low and then sends the required level 1 and/or level 2 parameter nibbles in response to each consecutive lowering of $\overline{\text{INRDY}}$. Note that the master must ensure that the DATA4 line is also low, indicating that the value in data lines DATA0–DATA3 is a command. If the master starts sending a command, then it must send all the parameters of the command back to back in response to each lowering of $\overline{\text{INRDY}}$ before reverting back to sending the rest of the speech file data again.

9) After sending the last nibble of speech data, the master can send dummy data (0xFF) to the slave in response to each consecutive lowering of $\overline{\text{INRDY}}$ until the slave detects the end-of-speech data.

10) When the MSP53C691 detects the end-of-speech, it pulls $\overline{\text{OUTRDY}}$ low.

11) The master must then read the data from the bus. Note that only the lower four bits of the bus are valid data. $\overline{\text{OUTRDY}}$ goes low three more times, sending a total of four nibbles (0, 0, 5, and 5) if speaking of the phrase was successful.

12) $\overline{\text{INRDY}}$ goes low again, indicating that the MSP53C691 is ready to receive a new command.

3.5.4.2 Speaking CELP, MELP, ADPCM in 8-Bit Data Transfer Mode

Data is sent over an eight-bit interface (DATA0–DATA7). Sending commands while speaking is not supported in this mode. When the speaking of a phrase is finished, the master can again send any command over the DATA0–DATA3 lines along with pulling DATA4 line low in response to the next lowering of $\overline{\text{INRDY}}$. In other words, the MSP53C691 goes back to the four-bit mode after it finishes speaking a phrase in eight-bit mode. The master must send the command header (4) and level 1 parameter (2) to speak a phrase again in eight-bit mode.

While sending a command in between phrases, the master must make sure that the DATA4 bit is low.

The following is a typical sequence of events that takes place while playing CELP, MELP, or ADPCM in eight-bit mode:

1) The MSP53C691 pulls $\overline{\text{INRDY}}$ low and waits for a command from the master.

2) The master sends the command header 0x04 to the MSP53C691. Note that writing 0x04 on the data bus ensures that the DATA4 line is also low, indicating that the value on data lines DATA0–DATA3 is a command.

Most Significant					
DATA LINES					
DATA4	DATA3	DATA2	DATA1	DATA0	
0	0	0	1	0	First command header sent

3) After receiving the command header, the MSP53C691 again pulls $\overline{\text{INRDY}}$ low and waits for the level 1 parameter from the master.

4) The master sends 0x2 as the level 1 parameter to indicate the eight-bit transfer mode for CELP, MELP, or ADPCM as follows:

Most Significant					
DATA LINES					
DATA4	DATA3	DATA2	DATA1	DATA0	
0	0	0	1	0	Level 1 parameter sent

5) The MSP53C691 interprets the command and initializes itself to get ready to speak.

6) The MSP53C691 pulls $\overline{\text{INRDY}}$ low, indicating that it is ready to receive speech data.

7) The master can now start sending speech data from a file in response to each lowering of $\overline{\text{INRDY}}$ over the whole eight-bit data bus (DATA0 through DATA7).

Most Significant								
DATA LINES								
DATA7	DATA6	DATA5	DATA4	DATA3	DATA2	DATA1	DATA0	
X	X	X	X	X	X	X	X	First byte sent

8) The master can not send a command while sending speech data in eight-bit mode. The master has to send the speech data in response to each lowering of $\overline{\text{INRDY}}$ by putting eight-bit speech data on the bus.

9) After finishing sending the last byte of speech data, the master can send dummy data (0xFF) to the slave to respond to each consecutive lowering of $\overline{\text{INRDY}}$ until the slave detects the end-of-speech.

10) When the MSP53C691 detects the end-of-speech, it pulls $\overline{\text{OUTRDY}}$ low.

11) The master then must read the data from the bus. Note that only the lower four bits of the bus are valid data. $\overline{\text{OUTRDY}}$ goes low three more times after that, sending a total of four nibbles (0, 0, 5, and 5) if the speaking of the phrase was successful.

12) $\overline{\text{INRDY}}$ goes low, indicating that the MSP53C691 is ready to receive a new command. At this point the slave changes back to four-bit mode. Now the master can send any command to the slave on the four-bit data bus (DATA0–DATA3) with the DATA4 line pulled low. If the master must speak again in eight-bit mode, repeat Steps 1 through 12.

Note:

The eight-bit interface is recommended for ADPCM.

The eight-bit interface does not support any interleaved commands (STOP, volume control, etc.). Hold RESET low to stop speaking in the middle of a phrase.

A header byte is embedded in the speech data to indicate which algorithm (MELP, CELP, etc.) to speak. When a certain mode of data transfer is selected (four-bit or eight-bit), the first few bytes of the speech file set up the MSP53C691 to play the speech file at the proper speed for the synthesizer algorithm. After the speech file is finished, the MSP53C691 goes back to running at 8 MIPS.

3.5.4.3 Test Mode

If the test mode is selected, data is transferred to and from the MSP53C691 (one nibble at a time). The MSP53C691 device receives one nibble of data and then sets $\overline{\text{OUTRDY}}$ low, meaning that the MSP53C691 is ready to send data. The MSP53C691 then sends the same nibble of data back to the microcontroller. $\overline{\text{INRDY}}$ is then set low, meaning that the MSP53C691 is ready to receive more data.

The following is a typical sequence of events for the test mode:

1) The MSP53C691 pulls $\overline{\text{INRDY}}$ low and waits for a command from the master.

2) The master sends the command header 0x4 on the data bus. Note that writing 0x4 to the data bus ensures that the DATA4 line is low, indicating that the value on data lines DATA0–DATA3 is a command.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	0	1	0	0	First command header sent

3) After receiving the header, the MSP53C691 pulls $\overline{\text{INRDY}}$ low again and waits for a level 1 parameter from the master.

4) The master sends 0x03 to the MSP53C691 as the level 1 parameter to indicate the test mode.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	0	0	1	1	Level 1 parameter sent

5) The MSP53C691 interprets the command and initializes itself to get ready to go to the test mode.

6) The MSP53C691 pulls $\overline{\text{INRDY}}$ low, indicating that it is ready to receive test data.

7) The master pulls R/$\overline{\text{W}}$ low, puts a nibble of test data on the bus, and pulses the $\overline{\text{STROBE}}$. When the $\overline{\text{STROBE}}$ is pulled low and then back to high, the data is latched into the MSP53C691 at the rising edge of $\overline{\text{STROBE}}$ and the MSP53C691 pulls $\overline{\text{INRDY}}$ high. Note that the data is sent on DATA0–DATA3 lines only.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
X	X	X	X	X	First nibble sent

8) When the data is latched into the MSP53C691, the MSP53C691 pulls $\overline{\text{OUTRDY}}$ low.

9) The master then must pull R/$\overline{\text{W}}$ high and pull $\overline{\text{STROBE}}$ low.

10) The master then must read the data from data lines DATA4–DATA7. Note that only the upper four bits of the bus (DATA4–DATA7) are valid data echoed back to the master. After that the master must pull $\overline{\text{STROBE}}$ high, indicating that the data has been read from the bus. $\overline{\text{OUTRDY}}$ goes high at the rising edge of the $\overline{\text{STROBE}}$.

11) The MSP53C691 pulls $\overline{\text{INRDY}}$ low again for the next data and the process repeats.

Note:
The only way to get out of the test mode is to reset the device.

3.5.4.4 Playing Single Channel FM Synthesis in Four-Bit Transfer Mode

The MSP53C691 can play single-channel FM in four-bit transfer mode. Sending commands to perform other functions (volume control, toggle I/O, etc.) is also supported in this mode. The FM data is supplied to the MSP53C691 from the master over the four-bit data lines (DATA0–DATA3). The DATA/COMMAND line is pulled high while transferring FM data. The MSP53C691 sends a request back to the master (by pulling $\overline{\text{OUTRDY}}$ low and transferring four nibbles to the master) whenever it needs more FM data. The master then must send a command back to the MSP53C691 informing how many bytes of FM data are going to be sent. Then the master can send the promised number of bytes to the MSP53C691 in response to each lowering of $\overline{\text{INRDY}}$. The following is a typical sequence of events for FM transfer:

1) The MSP53C691 pulls $\overline{\text{INRDY}}$ low and waits for a command from the master.

2) The master sends the command header 0x04 to the MSP53C691. Note that writing 0x04 to the data bus ensures that the DATA4 line is low, indicating that the value on data lines DATA0–DATA3 is a command.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	0	1	0	0	First command header sent

3) After receiving the header, the MSP53C691 pulls $\overline{\text{INRDY}}$ low to expect the level 1 parameter from the master.

4) The master sends 0x4 as the level 1 parameter to indicate four-bit transfer mode for playing FM.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	0	1	0	0	Level 1 parameter sent

5) The MSP53C691 interprets the command and initializes itself to get ready to speak FM.

6) The MSP53C691 pulls $\overline{\text{OUTRDY}}$ low, indicating that it must send a nibble back to the master.

7) The master reads the data from the bus (DATA0–DATA3).The MSP53C691 pulls $\overline{\text{OUTRDY}}$ low three more times to send a total of four nibbles. The four nibbles sent to the master are 1, 8, 2, and 3 respectively. The 1 and 8 are interpreted as 18H (24 decimal) bytes needed to fill up the FM buffer initially. The 2 and 3 are interpreted as 23, characterizing the FM data request to the master.

Most Significant				
DATA LINES				
DATA3	DATA2	DATA1	DATA0	
0	0	0	1	First nibble sent
1	0	0	0	Second nibble sent
0	0	1	0	Third nibble sent
Least Significant				
0	0	1	1	Fourth nibble sent

8) The MSP53C691 then pulls $\overline{\text{INRDY}}$ low to indicate that it is ready to receive FM data.

9) The master can now start sending FM data from a file in response to each lowering of $\overline{\text{INRDY}}$. Note that while sending speech FM, the DATA4 line should be high indicating that FM data is being sent. For instance, to send the FM data byte 0x38 to the MSP53C691, the master must put 0x13 on the data bus in response to lowering of $\overline{\text{INRDY}}$. When the MSP53C691 is again ready to receive the next nibble, the master puts 0x18 on the data bus.

Most Significant					
DATA LINES					
DATA4	DATA3	DATA2	DATA1	DATA0	
1	X	X	X	X	First nibble sent
1	X	X	X	X	Second through X nibbles sent
Least Significant					
1	X	X	X	X	Last (24X2=48th) nibble sent

Note:

It is very important to note that the master must send the exact number of FM data bytes initially requested by the MSP53C691 while filling up the buffer. If the master has finished sending the exact number of FM data bytes as requested by the MSP53C691 to initially fill up the buffer and the MSP53C691 has pulled $\overline{\text{INRDY}}$ low to request more data or command, the master can either send a command to perform any task (volume control, toggle I/O, etc.) or send a dummy command (0x00) if it does not want to send a valid command.

10) The MSP53C691 starts playing FM after the buffer is filled (that is, the first 24 (decimal) bytes from the file have been received). The MSP53C691 continues to pull $\overline{\text{INRDY}}$ low and waits for a command from the master. The master can either send a valid command to fulfill the request, or a dummy command if it doesn't have any more commands to send.

11) When the MSP53C691 needs more FM data, it pulls $\overline{\text{OUTRDY}}$ low to send a nibble back to the master. The master must read the nibble from the bus. The MSP53C691 pulls $\overline{\text{OUTRDY}}$ low three more times to transfer a total of four nibbles to the master. The four nibbles sent this time are 0, X, 2, and 3. The nibble X is interpreted as the number of FM data bytes requested by the MSP53C691. The 2 and 3 characterize the request for FM data.

Most Significant				
DATA LINES				
DATA3	**DATA2**	**DATA1**	**DATA0**	
0	0	0	1	First nibble sent
X	X	X	X	Second nibble sent
0	0	1	0	Third nibble sent
Least Significant				
0	0	1	1	Fourth nibble sent

12) The MSP53C691 pulls $\overline{\text{INRDY}}$ low after sending the four nibbles requesting the FM data from the master.

13) The master must send 9 as a command (0x09), letting the MSP53C691 know that master is going to send FM data.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	1	0	0	1	Command header sent

14) After receiving the 9 command, the MSP53C691 lowers $\overline{\text{INRDY}}$ again. In response, the master must send another nibble as level 1 parameter, letting the MSP53C691 know how many bytes of FM data it is going to send. This number must be less than or equal to the number of FM data requested by the MSP53C691. For example, the MSP53C691 may request 8 bytes of data, but the master may want to send only 6 bytes of FM data. Therefore, the master sends 0x6 to the MSP53C691.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	X	X	X	X	Level 1 parameter sent

15) After the level 1 parameter nibble has been latched, the MSP53C691 pulls $\overline{\text{INRDY}}$ low again, to get ready to receive the FM data. The master must now start sending the FM data in response to each lowering of $\overline{\text{INRDY}}$.

Note:
The master must send the exact number of FM bytes promised.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
1	X	X	X	X	First nibble sent
1	X	X	X	X	Second through X nibbles sent
Least Significant					
1	X	X	X	X	Last nibble sent

16) Once all of the promised FM bytes has been sent, the MSP53C691 pulls $\overline{\text{INRDY}}$ low again and waits for a command from the master. The master can send a valid command to the MSP53C691 (if desired) or a dummy command (0x00) if it has nothing else to send. If a valid command is to be sent, the master must ensure that DATA4 line is low while sending the command nibbles over the DATA0–DATA3 lines. When the MSP53C691 needs more FM data, it pulls $\overline{\text{OUTRDY}}$ low to let the master know the number of bytes of FM data requested, and the process is repeated.

17) After finishing sending the last nibble of FM data in the FM file, the master can send some dummy data (0xFF) to the slave in response to each consecutive lowering of $\overline{\text{INRDY}}$ until the slave detects the end-of-FM file.

18) When the MSP53C691 detects the end-of-file, it pulls $\overline{\text{OUTRDY}}$ low. The master then must read the data from the bus. $\overline{\text{OUTRDY}}$ goes low three more times after that, sending a total of four nibbles (0, 0, 5, and 5) if the speaking of the phrase was successful.

19) Then $\overline{\text{INRDY}}$ goes low again, indicating that the MSP53C691 is ready to receive a new command.

3.5.4.5 Playing Mixed Mode in 4-Bit Transfer Mode

In mixed mode, the single-channel FM is played in the background and CELP/MELP is spoken in the foreground.

The MSP53C691 can play mixed mode in four-bit transfer mode. This mode also supports sending commands to perform other functions (volume control, toggle I/O, etc.). The CELP/MELP and FM data is supplied to the MSP53C691 from the master over the four-bit data lines (DATA0–DATA3). The DATA/COMMAND line is pulled high while transferring both the CELP/MELP and the FM data. The master sends CELP/MELP data to the MSP53C691 by default. The MSP53C691 sends a request back to the master (by pulling $\overline{\text{OUTRDY}}$ low and transferring four nibbles to it) whenever it needs more FM data. The master must then send a command (with parameter) back to the MSP53C691, indicating how many bytes of FM data it is going to send. Then the master can send the promised number of bytes to the MSP53C691 in response to each lowering of $\overline{\text{INRDY}}$, until the promised number of FM data-bytes is sent. The following is a typical sequence of events for mixed mode transfer:

1) The MSP53C691 pulls $\overline{\text{INRDY}}$ low and waits for a command from the master.

2) The master sends the command header 4 to the MSP53C691. Note that writing 0x04 to the data bus ensures that the DATA4 line is low, indicating that the value on the data lines DATA0–DATA3 is a command.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	0	1	0	0	First command header sent

3) After receiving the command header, the MSP53C691 pulls $\overline{\text{INRDY}}$ low again to request the level 1 parameter from the master.

4) The master sends 0x5 to the MSP53C691 as the level 1 parameter to indicate the four-bit transfer mode for playing mixed mode.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	0	1	0	1	Level 1 parameter sent

5) The MSP53C691 interprets the command and initializes itself to get ready to speak the mixed mode.

6) When the MSP53C691 is ready, it pulls $\overline{\text{INRDY}}$ low and waits for the CELP/MELP data. The master must start sending the CELP/MELP data to the MSP53C691 from the speech file. The MSP53C691 first fills up the buffer with the CELP/MELP data before starting to speak the phrase. The master must send the CELP/MELP to the MSP53C691 in response to each $\overline{\text{INRDY}}$ low to fill up the buffer. The buffer to store CELP/MELP data in the MSP53C691 is 68 bytes long. Note that the DATA4 line must be high while sending the CELP/MELP speech data, indicating that CELP/MELP data is being sent. For instance, to send the speech data byte 0xAC to the MSP53C691, the master must put 0x1A on the data bus when $\overline{\text{INRDY}}$ goes low. When the MSP53C691 is again ready to receive the next nibble, the master puts 0x1C in the data bus.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
1	X	X	X	X	First nibble sent
1	X	X	X	X	Second through X nibbles sent
Least Significant					
1	X	X	X	X	Last ($68X2=136^{th}$) nibbles sent

7) When the CELP/MELP buffer is full, the MSP53C691 pulls $\overline{\text{OUTRDY}}$ low, indicating that it must send a nibble back to the master. The master reads the data from the bus (DATA0–DATA3). The MSP53C691 lowers

$\overline{\text{OUTRDY}}$ again three more times, sending a total of four nibbles altogether (back to back) in this way. The four nibbles sent to the master are 1, 8, 2, and 3 respectively. The 1 and 8 are interpreted as 18H (24 decimal) bytes needed to fill up the FM buffer initially. The 2 and 3 are interpreted as 23, characterizing the FM data request to the master.

Most Significant				
DATA LINES				
DATA3	**DATA2**	**DATA1**	**DATA0**	
0	0	0	1	First nibble sent
1	0	0	0	Second nibble sent
0	0	1	0	Third nibble sent
Least Significant				
0	0	1	1	Fourth nibble sent

8) The MSP53C691 then pulls $\overline{\text{INRDY}}$ low again to indicate that it is ready to receive FM data.

9) The master can now start sending FM data from a file in response to each lowering of $\overline{\text{INRDY}}$. Note that the DATA4 line must be high while sending FM, indicating that FM data is being sent. For instance, to send the FM data byte 0x38 to the MSP53C691, the master must put 0x13 on the data bus in response to the lowering of $\overline{\text{INRDY}}$. When the MSP53C691 is ready to receive the next nibble, the master puts 0x18 on the data bus.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
1	X	X	X	X	First nibble sent
1	X	X	X	X	Second through X nibbles sent
Least Significant					
1	X	X	X	X	Last (24X2=48th) nibble sent

Note:

It is very important to note that the master must send the exact number of FM data bytes initially requested by the MSP53C691 while filling up the buffer. If the master has finished sending the exact number of FM data bytes requested by the MSP53C691 to initially fill up the buffer, and the MSP53C691 has pulled $\overline{\text{INRDY}}$ low to request more data or command, the master can either send a command to perform any task (volume control, toggle I/O, etc.) or CELP/MELP data.

10) The MSP53C691 starts playing FM after the buffer is filled (that is, the first 24 (decimal) bytes from the file has been received). The MSP53C691 still continues to pull $\overline{\text{INRDY}}$ low and waits for speech data or command from the master. The master can either send a valid command or send CELP/MELP speech data.

11) When the MSP53C691 needs more FM data, it pulls $\overline{\text{OUTRDY}}$ low to send a nibble back to the master. The master must read the nibble from the bus. The MSP53C691 sends a total of four nibbles to the master by pulling $\overline{\text{OUTRDY}}$ low three more times and the master must read the four nibbles as previously described in step 7. The four nibbles sent this time are 0, X, 2, and 3. X is the number of FM data bytes requested by the MSP53C691. The 2 and 3 indicate a request for FM data.

Most Significant				
DATA LINES				
DATA3	**DATA2**	**DATA1**	**DATA0**	
0	0	0	0	First nibble sent
X	X	X	X	Second nibble sent
0	0	1	0	Third nibble sent
Least Significant				
0	0	1	1	Fourth nibble sent

12) The MSP53C691 pulls $\overline{\text{INRDY}}$ low again after sending the four nibbles requesting the FM data from the master.

13) In response, the master must send 9 as a command (0x09), letting the MSP53C691 know that master is going to send FM data.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	1	0	0	1	Command header sent

14) After receiving the command 9, the MSP53C691 lowers $\overline{\text{INRDY}}$ again. In response, the master must send another nibble as the level 1 parameter, letting the MSP53C691 know how many bytes of FM data it is going to send. This number must be less than or equal to the requested number of FM data by the MSP53C691. For example, the MSP53C691 may request 8 bytes of data, but the master may wish to send only 6 bytes of FM data. Therefore, the master sends 0x6 to the MSP53C691.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	X	X	X	X	Level 1 parameter sent

15) After the level 1 parameter nibble has been latched, the MSP53C691 again pulls $\overline{\text{INRDY}}$ low to get ready to receive the FM data. The master must start sending the FM data in response to each lowering of $\overline{\text{INRDY}}$.

Note:
The master must send the exact number of FM data promised to the MSP53C691.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
1	X	X	X	X	First nibble sent
1	X	X	X	X	Second through X nibbles sent
Least Significant					
1	X	X	X	X	Last nibble sent

16) After all the promised number of FM data has been sent, the MSP53C691 again pulls $\overline{\text{INRDY}}$ low and waits for a command or CELP/MELP data from the master. The master can send a valid command to the MSP53C691 (if it is desired) or send CELP/MELP data. If a valid command is to be sent, ensure that the DATA4 line is low while sending the command nibbles over the DATA0–DATA3 lines. When the MSP53C691 needs more FM data, it pulls $\overline{\text{OUTRDY}}$ low and the process of step 7 is repeated.

Note:
If the master starts sending a command, it must send all the nibbles of the command back to back before starting to send CELP/MELP data or responding to an FM request.

17) If the master has sent the last nibble of a file to the MSP53C691 buffer, then it can send dummy commands (0x00) in response to $\overline{\text{INRDY}}$ low if it is not servicing an FM request or does not have any valid commands to send.

Note:
If the master wishes to speak another phrase, it must wait to finish the current phrase. As mentioned earlier, the master can send dummy commands (0x00) if it not servicing an FM request or sending valid commands.

18) When the MSP53C691 finishes speaking a phrase, it pulls $\overline{\text{OUTRDY}}$ low to send a nibble to the master. The master must read the nibble. The MSP53C691 pulls $\overline{\text{OUTRDY}}$ low three more times sending a total of four nibbles (0, 0, 5, and 4) to the master if the MSP53C691 has successfully finished speaking the phrase.

19) If the master wishes to speak another phrase at this point, it must send the command header 0x4 to the slave.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	0	1	0	0	Command header sent

20) After the 0x4 has been received, the MSP53C691 pulls $\overline{\text{INRDY}}$ low again requesting the level 1 parameter from the master.

21) The master must now send 6 as the level 1 parameter to the MSP53C691, letting it know that it wishes to speak multiple files.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	0	1	0	1	Level 1 parameter sent

22) When the MSP53C691 receives the command, it pulls $\overline{\text{INRDY}}$ low again, requesting the CELP/MELP data for the next phrase to speak.

23) The master must now start sending the speech data for the next phrase. The MSP53C691 first fills the buffer with the CELP/MELP data supplied by the master. In the meantime, the MSP53C691 continues playing FM. The master must send the speech data as long as the $\overline{\text{INRDY}}$ is being pulled low. When the buffer is full, the MSP53C691 pulls $\overline{\text{OUTRDY}}$ low to request for more FM data.

Note:
It is very important that the master sends all the speech data to fill up the buffer as fast as possible in order to avoid interruption in the playing of FM.

Most Significant					
DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
1	X	X	X	X	First nibble sent
1	X	X	X	X	Second through X nibbles sent
Least Significant					
1	X	X	X	X	Last (24X2=48th) nibble sent

24) After filling up the buffer, the MSP53C691 starts playing the new phrase. The MSP53C691 now pulls $\overline{\text{INRDY}}$ low requesting more CELP/MELP data or commands or pulls $\overline{\text{OUTRDY}}$ low to request more FM data, and the process described in steps 11 through 16 is repeated.

25) If the master is ready to speak another phrase after this, it must follow the process described in steps 18 through 25.

26) If the master is not ready to speak more phrases after the current phrase, it must not send the command header 4 and the level 1 command 6 to start

another phrase after receiving the four nibbles (0, 0, 5, and 4) from the MSP53C691. After that, the master only needs to respond and send FM data when requested by the slave. If there is no more FM data to send, the master sends valid or dummy commands (0x00) in response to the lowering of $\overline{\text{INRDY}}$.

27) After finishing sending the last nibble of FM data in the FM file, the master can send dummy data (0xFF) to the slave in response to each consecutive lowering of $\overline{\text{INRDY}}$ by the MSP53C691 until the FM finishes playing.

28) When the MSP53C691 finishes playing FM, it pulls $\overline{\text{OUTRDY}}$ low.

29) The master must read the data from the bus. Note that only the lower four bits of the bus are valid data. The $\overline{\text{OUTRDY}}$ goes low three more times after that, sending a total of four nibbles (0, 0, 5, and 3) if the playing of FM was successful.

Note:
If the FM is finished while the MSP53C691 is still speaking a phrase in MELP/CELP, then the MSP53C691 sends 0, 0, 5, and 3 to let the master know that the playing of FM has ended and continues to pull $\overline{\text{INRDY}}$ low to receive the MELP/CELP speech data. The master must continue to send the CELP/MELP data to the MSP53C691.

30) When both the FM and the CELP/MELP files have ended, the MSP53C691 sends four more nibbles (0, 0, 5, and 5) to the master to indicate that the playing for mixed mode has ended.

31) After that $\overline{\text{INRDY}}$ goes low, indicating that the MSP53C691 is ready for a new command.

Note:

The master cannot restart a new FM file after an FM file has finished playing while a speech file is in progress. However, the master can issue a new command to speak in four-bit mixed mode after both the FM and the CELP/MELP file has ended.

While playing FM and CELP in mixed mode, only a new CELP file can be restarted. In a similar way, only a new MELP file can be restarted while playing FM and MELP. A new MELP file cannot be restarted while playing FM and CELP. A new CELP file cannot be restarted while playing FM and MELP.

3.5.4.6 Playing Sinewave in Test Mode

If the test mode to play a pure sine wave is selected, the MSP53C691 plays a pure sine wave tone. The tone comes over the DAC pins of the MSP53C691. If a 32-Ω speaker is directly connected to the output, a pure sine wave tone can be heard. The following is a typical sequence of events for the test mode:

1) The MSP53C691 pulls $\overline{\text{INRDY}}$ low and waits for a command from the master.

2) The master sends 0x4 as command header on the data bus. Note that by writing 0x04 to the data bus ensures that the DATA4 line is also low, indicating that the value on the data lines DATA0–DATA3 is a command.

Most Significant					
DATA LINES					
DATA4	DATA3	DATA2	DATA1	DATA0	
0	0	1	0	0	First command header sent

3) After receiving the command header, the MSP53C691 pulls $\overline{\text{INRDY}}$ low again and waits for the level 1 parameter from the master.

4) The master must send 0x7 as the level 1 parameter to indicate the test mode.

Most Significant					
DATA LINES					
DATA4	DATA3	DATA2	DATA1	DATA0	
0	0	1	1	1	Level 1 parameter sent

5) The MSP53C691 interprets the command and initializes itself to get ready to go to the test mode.

6) The MSP53C691 outputs a pure sine wave tone of 1 kHz frequency through its DAC pins.

3.5.5 Command Header 5—Stop Speaking

The MSP53C691 can stop speaking at any time while speaking either CELP/MELP only in four-bit mode, playing FM only in four-bit mode, or speaking both CELP/MELP and FM in mixed mode.

The sequence of events for stopping speaking while playing CELP/MELP or FM only is as follows:

1) The MSP53C691 pulls $\overline{\text{INRDY}}$ low while speaking a phrase requesting a command, speech, or FM data.

2) The master sends the command header 0x5 to the MSP53C691. The DATA/COMMAND line is pulled low while sending the header to indicate a command.

DATA LINES					
DATA4	DATA3	DATA2	DATA1	DATA0	
0	0	1	0	1	Command header set

3) The MSP53C691 pulls $\overline{\text{INRDY}}$ low again requesting the level 1 command.

4) The master sends 1 as the level 1 command to the MSP53C691.

DATA LINES					
DATA4	DATA3	DATA2	DATA1	DATA0	
0	0	0	0	1	Level 1 command set

5) The MSP53C691 stops speaking the phrase and pulls $\overline{\text{OUTRDY}}$ low to send a nibble back to the master. The master must read the nibble from the MSP53C691.

6) The MSP53C691 pulls $\overline{\text{OUTRDY}}$ low three more times sending a total of four nibbles (0, 0, 5, and 5), indicating the successful stop of the speaking of the phrase and the command.

7) The MSP53C691 then pulls $\overline{\text{INRDY}}$ low and waits for another command from the master.

The sequence of events for stopping speaking CELP/MELP or FM while playing mixed mode is as follows:

1) The MSP53C691 pulls $\overline{\text{INRDY}}$ low while speaking a phrase requesting a command or speech or FM data.

2) The master sends a command header 0x5 to the MSP53C691. The DATA/COMMAND line is pulled low while sending the header to indicate a command.

DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	0	1	0	1	Command header set

3) The MSP53C691 pulls $\overline{\text{INRDY}}$ low again requesting the level 1 command.

4) The master sends the level 1 command 0x2 to the MSP53C691 if it is ready to stop speaking CELP/MELP but must continue playing FM.

DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	0	0	1	0	Level 1 command sent

The master sends the level 1 command 0x3 to the MSP53C691 if it is ready to stop playing FM but must continue speaking CELP/MELP.

DATA LINES					
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	0	0	1	1	Level 1 command sent

5) The MSP53C691 stops speaking the phrase or FM and pulls $\overline{\text{OUTRDY}}$ low to send a nibble back to the master. The master must read the nibble from the MSP53C691.

6) The MSP53C691 pulls $\overline{\text{OUTRDY}}$ low again sending a total of four nibbles: either 0, 0, 5, and 4 indicating the successful stopping of the speaking of the phrase and the command, or 0, 0, 5, and 3 indicating the successful stopping of FM and completion of the command. Once a phrase is stopped, the master can restart another phrase by sending the command 4 followed by 6 (as described in the *Playing Mixed Mode In Four Bit* section) in response to lowering of $\overline{\text{INRDY}}$ by the MSP53C691 requesting the command.

7) The MSP53C691 then pulls $\overline{\text{INRDY}}$ low waiting for another command from the master.

3.5.6 Command Header 6—Adjust the Volume

The MSP53C691 has six levels of volume control from the highest to lowest. The default is the highest volume. The volume can be changed at anytime while speaking a phrase or FM or in mixed mode. The sequence of events for sending the volume command is as follows:

1) The MSP53C691 pulls $\overline{\text{INRDY}}$ low to receive a command or CELP/MELP data. The master sends the command header 0x6 for volume control.

DATA LINES				
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**
0	0	1	1	0

2) When the command header is latched into the MSP53C691, the MSP53C691 pulls $\overline{\text{INRDY}}$ low again requesting the level 1 parameter. The master sends any value from 1 through 6 representing the desired volume (1 being the lowest, 6 being the highest).

DATA LINES					FUNCTIONS
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	0	0	0	0	Level 1—Lowest volume x(0x0)
0	0	0	0	1	Level 2 volume x(0x1)
0	0	0	1	0	Level 3 volume x(0x2)
0	0	0	1	1	Level 4 volume x(0x3)
0	0	1	0	0	Level 5 volume x(0x4)
0	0	1	0	1	Level 6 volume x(0x5)
0	0	1	1	0	Level 7—Highest volume x(0x6)

3) After the level 1 parameter is received, the MSP53C691 adjusts the volume and then pulls $\overline{\text{INRDY}}$ low, requesting more commands or speech data.

Note:
Each level change in volume control represents 6 dB of volume difference.

3.5.7 Command Header 7—Return Buffer Status

The MSP53C691 can be queried to return the status of the CELP/MELP or FM buffer at anytime while speaking. The MSP53C691 returns four nibbles representing the number of empty bytes in the buffer. The sequence of events for returning the status of a buffer is as follows:

1) The MSP53C691 pulls $\overline{\text{INRDY}}$ low, requesting command or speech data.

2) The master sends the command header 0x7 as follows:

DATA LINES				
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**
0	0	1	1	1

3) After the command header is received and latched into the MSP53C691, the MSP53C691 pulls $\overline{\text{INRDY}}$ low again, requesting the level 1 command.

4) The master can send the level 1 command 1 (if requesting CELP/MELP buffer status) or 2 (if requesting FM buffer status).

DATA LINES				
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**
0	0	0	X	X

5) In return, the MSP53C691 pulls $\overline{\text{OUTRDY}}$ low to send a nibble back to the master. The master reads the nibble from the MSP53C691. The MSP53C691 pulls $\overline{\text{OUTRDY}}$ low again sending a total of four nibbles: 0, 0, X_1, and X_2. X_1X_2 represent the number of empty bytes in the buffer in hexadecimal format.

3.5.8 Command Header 8—Initiate Sleep

The MSP53C691 can be put into sleep mode when it is not speaking or even during speaking. The MSP53C691 can be put into three levels of sleep modes: light, mid, and deep. For details about the specification for the three different sleep modes, consult the data sheet listed in Table 2–9 of the SPSU014 manual. The sequence of events in sleep mode is as follows:

1) The MSP53C691 pulls $\overline{\text{INRDY}}$ low requesting for command or more speech data.

2) The master sends the command header 0x8 to the MSP53C691.

DATA LINES				
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**
0	1	0	0	0

3) The MSP53C691 pulls $\overline{\text{INRDY}}$ low again to request a level 1 command. The master sends a 1, 2, or 3, depending upon the level of sleep.

DATA LINES				
DATA4	DATA3	DATA2	DATA1	DATA0
0	X	X	X	X

4) After receiving the level 1 command, the MSP53C691 pulls $\overline{\text{INRDY}}$ low and then goes to the requested level of sleep.

Note:

When the master is ready to wake up the MSP53C691, it must send a dummy byte to the MSP53C691 to respond to the $\overline{\text{INRDY}}$ low.

The wakeup time from light sleep is the lowest. The MSP53C691 takes more time to wake up from mid and deep sleep because ramp-up and stabilization of the clock is required in mid and deep sleep.

The device draws less than 10 μA of current In deep sleep.

3.5.9 Command Header 9—Receive FM Data

The command header 9 is used in mixed mode to send the requested amount of FM data to the MSP53C691. For detailed description of the command, see the *Playing Mixed Mode in Four-Bit Mode* section.

3.5.10 Command Header A—Perform Speed Shift or Pitch Shift

This command changes the speed or the pitch of a file being played in MELP. The speed or pitch shift gives a very interesting flavor to the audio quality of MELP. See Appendix D, *Speed and Pitch Shifting in 6xx MELP*. The sequence of events to perform pitch and speed shift is as follows:

1) The MSP53C691 pulls $\overline{\text{INRDY}}$ low, requesting for command or speech data while playing MELP.

2) The master sends the command header 0xA to indicate the pitch or speed shift.

DATA LINES				
DATA4	DATA3	DATA2	DATA1	DATA0
0	1	0	1	0

3) After receiving the command header, the MSP53C691 pulls $\overline{\text{INRDY}}$ low again requesting a level 1 parameter from the master. The master must send the level 1 parameter to the MSP53C691: 0 for slowing down MELP, 1 for speeding up MELP, or 2 for pitch shifting.

DATA LINES					FUNCTIONS
DATA4	**DATA3**	**DATA2**	**DATA1**	**DATA0**	
0	0	0	0	0	Slow down MELP x(0x0)
0	0	0	0	1	Speed up MELP x(0x1)
0	0	0	1	0	Pitch shifting x(0x2)
0	0	0	1	1	Reserved x(0x3)

4) After receiving the level 1 command, the MSP53C691 pulls $\overline{\text{INRDY}}$ low again requesting the first of the four nibbles characterizing the amount of speed or pitch shift needed (level 2 command). The MSP53C691 pulls $\overline{\text{INRDY}}$ low again three more times after that requesting the remaining three nibbles.

5) After all the four nibbles has been received, the MSP53C691 changes the pitch or speed of the speech file as requested.

6) After that the MSP53C691 pulls $\overline{\text{INRDY}}$ low again to request another command or MELP data.

Refer to appendix D, *Speed and Pitch Shifting in 6xx MELP* for details.

Chapter 4

MSP53C691 Timing Considerations

This chapter discusses general timing constrains and waveforms.

Topic **Page**

4.1 General Constraints

The sound quality of the speech produced by the MSP53C691 is sensitive to the timing of speech data transfers from the master microprocessor. The speech data is stored in a circular buffer. The buffer sizes are 26 bytes for FM and 68 bytes for CELP, MELP, and ADPCM. If the data is written to the slave too infrequently, the buffer becomes empty and the synthesis process stops or becomes corrupted. It is very important that the master supplies the speech data to the slave as soon as it is requested.

Each of the synthesis algorithms use the data in the buffer at various rates. In general, the slave requests speech data (sets $\overline{\text{INRDY}}$ low) as soon as there are at least two bytes empty in the buffer. The rate at which the slave requests speech data depends upon two things: 1) The rate at which the data is consumed by the synthesizer algorithm (that is, the bit rate of the algorithm) and 2) The rate at which the slave queries the buffer to check for empty bytes. The slave generally requests new data in bursts; that is, while processing a frame the slave requests new data as soon as there are two empty bytes in the CELP/MELP buffer. Once the processing of a frame is done, the slave ceases to request data until it is time to process the next frame. Once the MSP53C691 has pulled $\overline{\text{INRDY}}$ low to request new data, the data must be loaded quickly until the buffer is again full and the slave stops pulling $\overline{\text{INRDY}}$ low. It is highly recommended that the slave read and write status pins ($\overline{\text{INRDY}}$ and $\overline{\text{OUTRDY}}$) be tied to an interrupt of the master for most efficient data transfer. A polling method can be used to find the status of the $\overline{\text{INRDY}}$ and $\overline{\text{OUTRDY}}$ pins; however, it is not a suggested method for mix mode operation or for playing high bit rate CELP or ADPCM.

The master can send commands to the slave in four-bit mode while speaking. However, this feature makes the precise characterization of timing more difficult because of the widely varied command set. The number of commands that can be sent to the slave while speaking varies and depends on the command being sent and on the the bit rate of the synthesizer.

The following sections show timing waveforms which have been found to work with tested data sets. It is often difficult for the system designer to exactly replicate the timing shown in these sections. The timing must be adjusted to optimize the sound quality for the specific system being designed. In each case, only the speech data was sent to the slave without any interleaved commands.

The following general considerations must be observed:

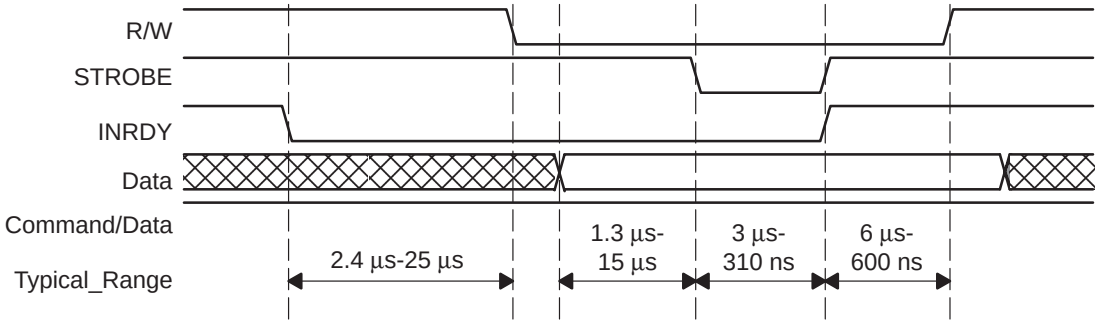
- Use interrupt service routines to send and receive data. This provides a fast response and allows speech data to be sent even while the master is performing other tasks.
- Avoid disabling the interrupts tied to $\overline{\text{INRDY}}$ and $\overline{\text{OUTRDY}}$. If it is necessary to turn off the interrupts tied to $\overline{\text{INRDY}}$ and $\overline{\text{OUTRDY}}$, make sure that they are not kept off so long that the slave's buffer underflows. The MSP53C691 has a 68-byte internal buffer to store incoming speech data. If CELP data for 11.2 kbps at 8-kHz sampling rate is sent, the buffer can hold approximately 48 ms of speech. If the speech file is all voiced or has a higher sampling rate, the buffer can hold an even shorter duration of speech. It is important to consider the state of the buffer before turning off the interrupts.

- ❑ Keep the interrupt service routines small so that a new interrupt service request does not occur while the master is processing the previous request.
- ❑ The number of commands that can be sent to the slave depends upon various factors, mainly the bit rate of the speech file. Sending too many commands back to back can cause buffer underflow. It is best to space the commands out in time if possible.
- ❑ Keep the strobe pulses as short as possible, but not shorter than shown in the following waveforms.
- ❑ The master can, by default, drive R/W high, except when writing to the slave.
- ❑ Commands can be interleaved between speech data, but the master should send all the parameters of the command (if a command is initiated) before starting to send speech data again.
- ❑ It is recommended that the master send the commands at the even boundary of the speech nibbles.
- ❑ Use pullup resistors in the data bus to keep the data bus from floating.
- ❑ Ensure that the master to the slave have a common ground connection

4.2 MSP53C391 Timing Waveforms

The data request frequency of the slave depends upon the bit rate and the type of synthesizer algorithm being played. A typical range of the timing waveforms for various synthesizers is provided below.

4.2.1 CELP/MELP



The average time between $\overline{\text{INRDY}}$ low for CELP speech data transfer is 152 µS.

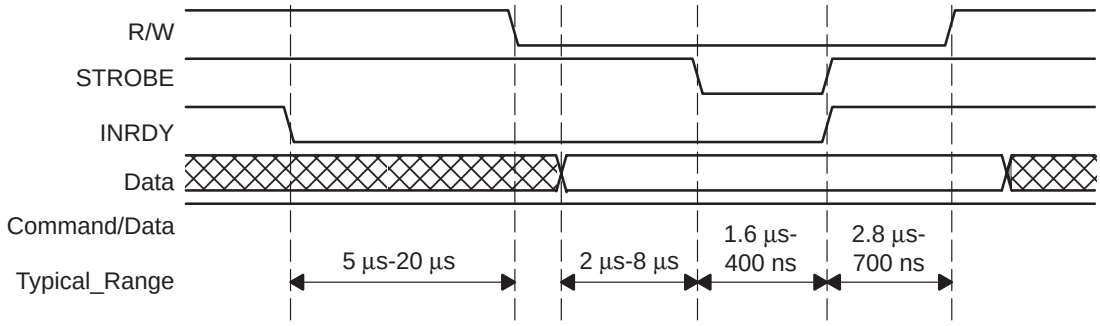
The average time between $\overline{\text{INRDY}}$ low for MELP speech data transfer is 122 µS.

The average time between $\overline{\text{INRDY}}$ low for FM speech data transfer only is 183 µS.

The average time between $\overline{\text{INRDY}}$ low for ADPCM speech data transfer is 30.5 µS.

NOTE: The time between $\overline{\text{INRDY}}$ low is less than the stated average while a command is being transferred.

4.2.2 MIX Mode



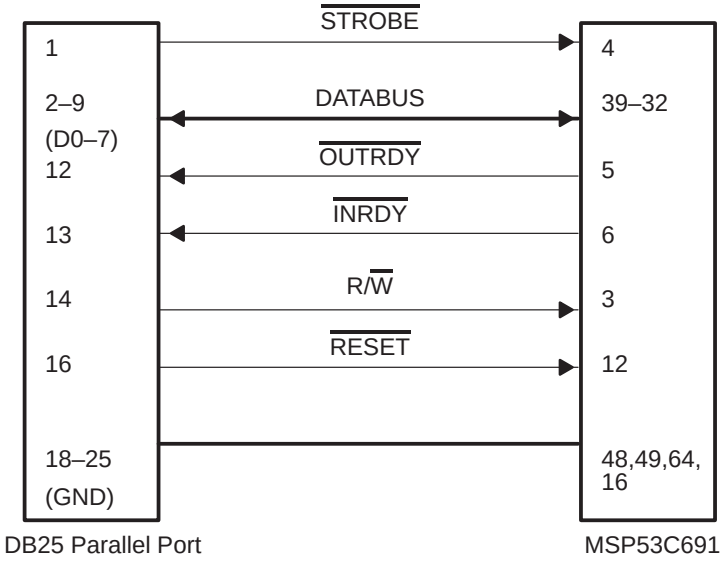
The average time between $\overline{\text{INRDY}}$ low for speech data transfer for mixed mode for speaking CELP and FM is 30.5 µS.

The average time between INRDY low for speech data transfer mixed mode for speaking MELP and FM is 91.5 µS.

NOTE: The time between $\overline{\text{INRDY}}$ low is less than the stated average while a command is being transferred.

The parallel port MSP53C691 hardware interface connection is shown in Figure 4–1.

Figure 4–1. MSP53C691 Hardware Interface Connection



Appendix A

Designing the Master Microcontroller Software

This chapter covers the master microcontroller software design.

Topic **Page**

A.1 Master Code Flow

The design of the master code to control the slave varies widely between applications. The master's protocol for the stand-alone vocoders differs from the mixed mode's protocol. Flowcharts are provided in Figures A–1 through A–5 for playing MELP/CELP and mixed mode as an aid in writing the master code's interface. A reference code in C is also provided to interface the MSP53C691 to the PC's parallel port to play MELP/CELP. It is recommended that the suggested flow of the master code be followed as closely as possible.

A.1.1 Flowchart for the Master to Play Mixed Mode (FM+CELP/MELP)

The flowcharts for the master code for playing mixed mode are provided in Figures A–1, A–2, A–3, and A–4. The master is divided into three main routines:

- Main routine
- Interrupt service routine triggered by the falling edge of $\overline{\text{INRDY}}$
- Interrupt service routine triggered by the falling edge of $\overline{\text{OUTRDY}}$

The main routine of the master is responsible for resetting and initializing the slave. It is also responsible for selecting the oscillator by sending the oscillator selection command. The main loop can be set up to send the first command along with its parameter. Typically, this first command would be a SPEAK command. The main loop can then go into an infinite loop while waiting for an interrupt to send or receive data.

The ISR, triggered by the falling edge of the $\overline{\text{OUTRDY}}$, is responsible for receiving data from the slave. Whenever the slave has data to send to the master, it pulls $\overline{\text{OUTRDY}}$ low. The master responds to the interrupt, reads the data from the bus, and decodes the data. The slave pulls $\overline{\text{OUTRDY}}$ low three times to send a total of four nibbles after the master acknowledges the receipt of the first nibble.

The ISR, triggered by the falling edge of $\overline{\text{INRDY}}$, is responsible for sending a nibble to the slave. This ISR is the backbone of the master code and is responsible for sending commands or speech data to the slave. The ISR decides what to send based on a priority list and sends the command/data. The priority list is explained in the flowchart. The routine for sending data to the slave is described as a separate routine for clarity.

Figure A–1. Program Flow for the Master Main Routine

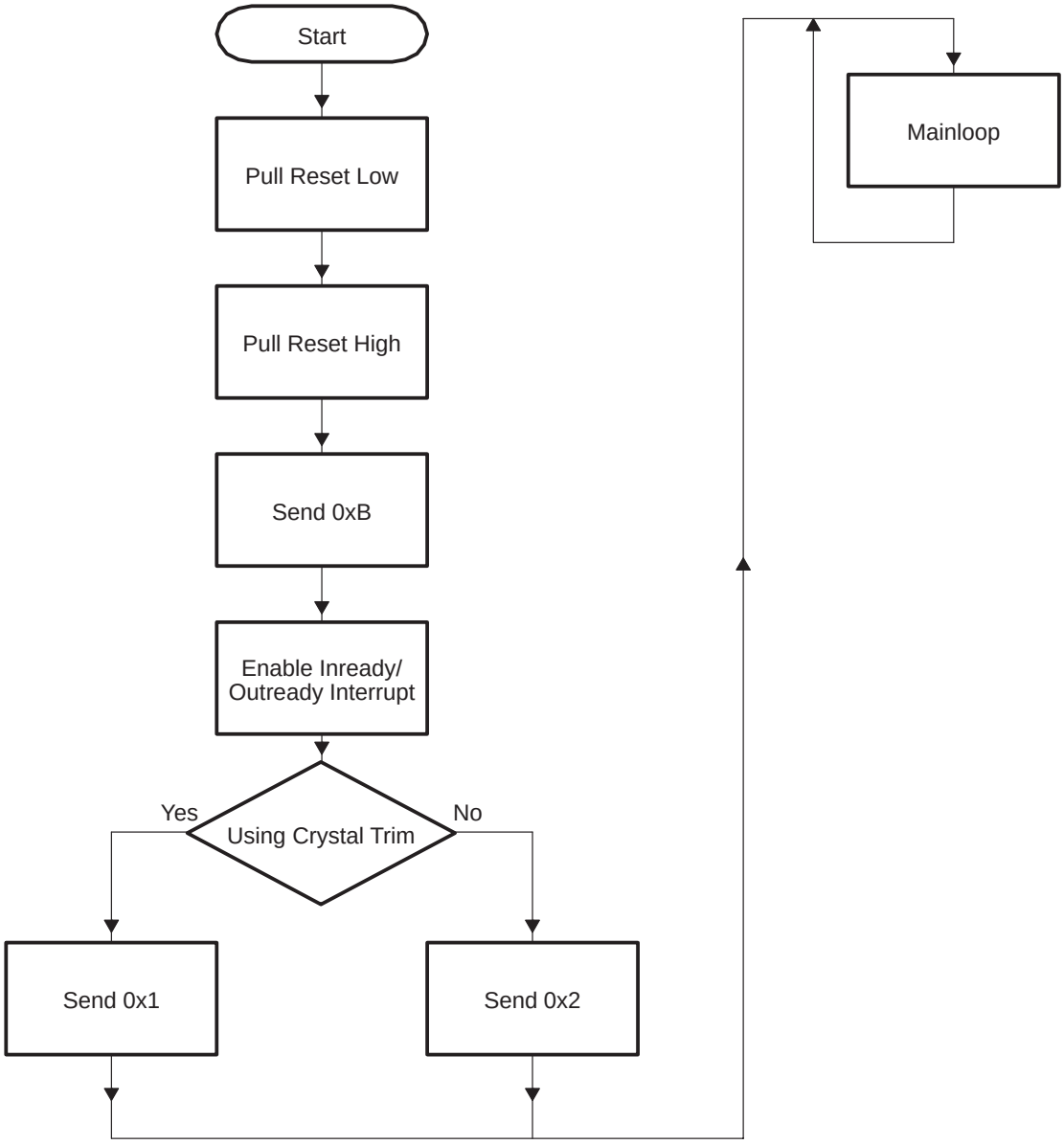


Figure A–2. Program Flow for an ISR Tied to the Falling Edge of OUTRDY

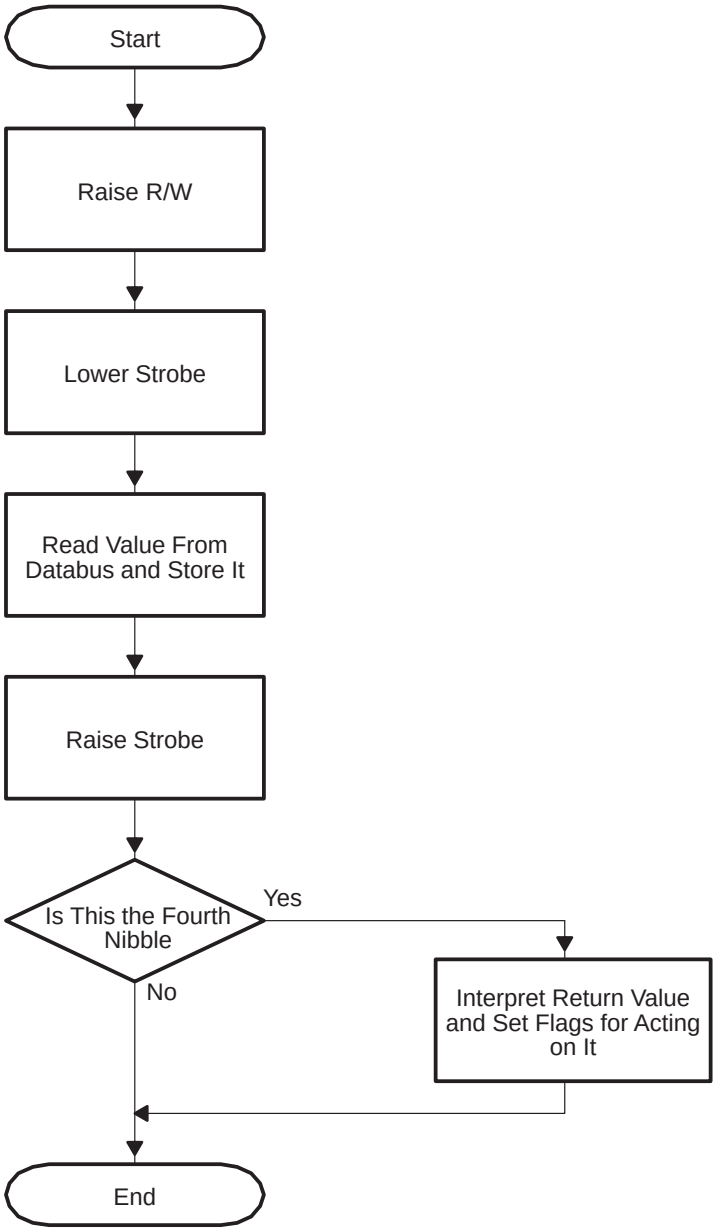


Figure A–3. Program Flow for an ISR Tied to the Falling Edge of $\overline{INRDY}$ to Play Mixed Mode

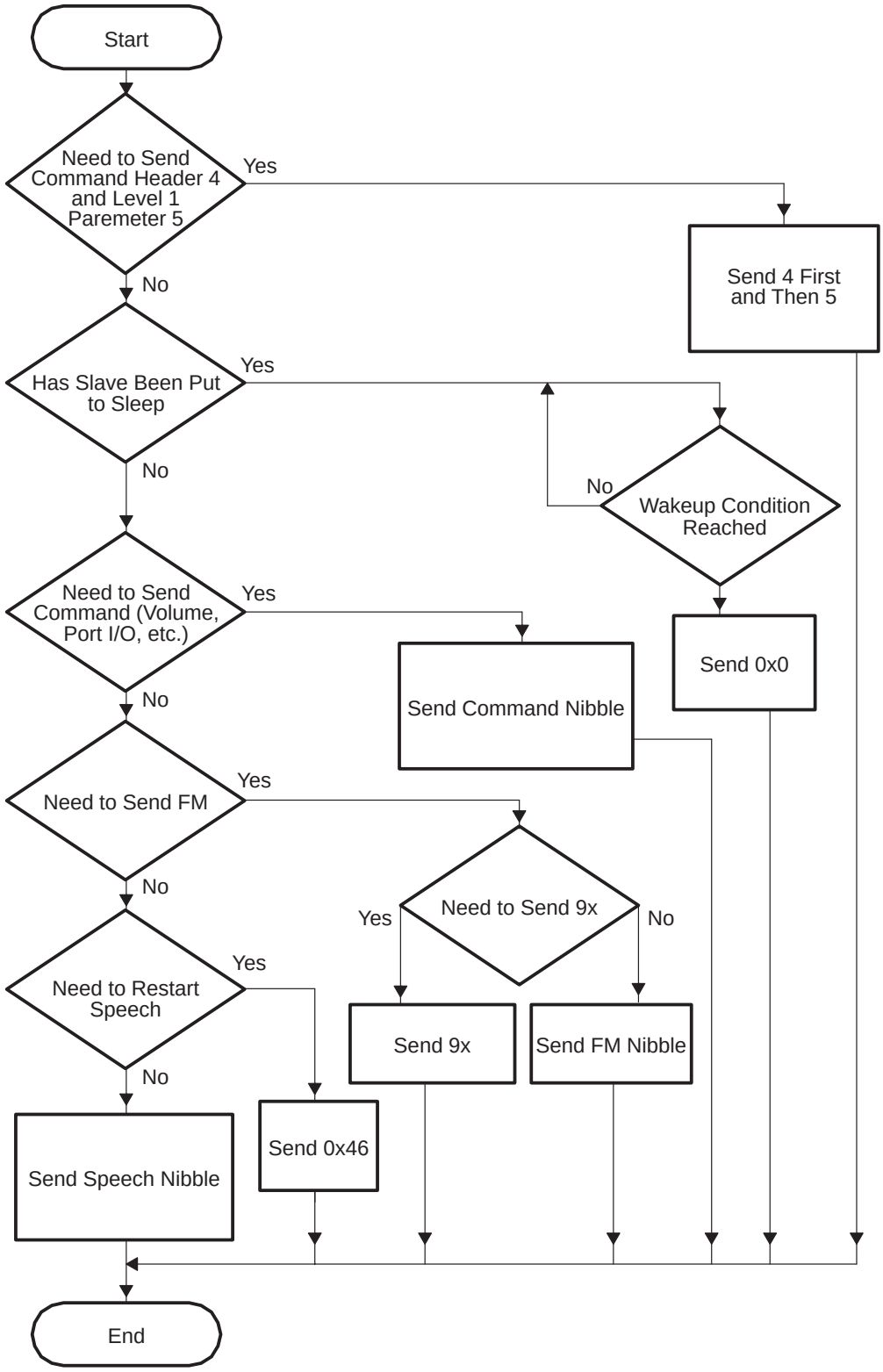
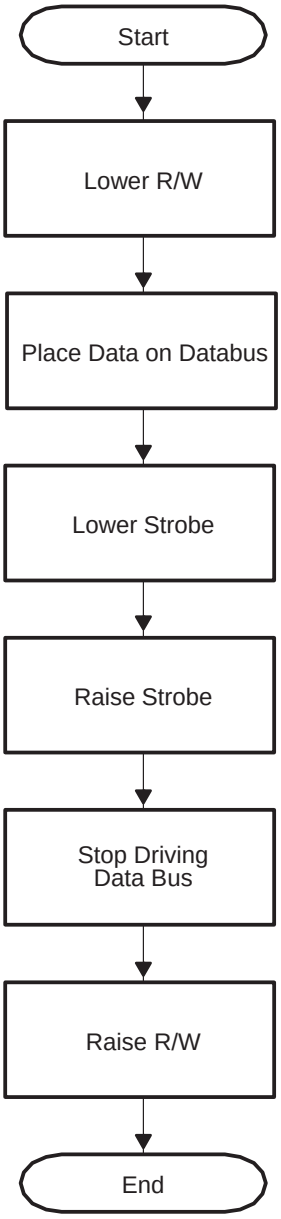


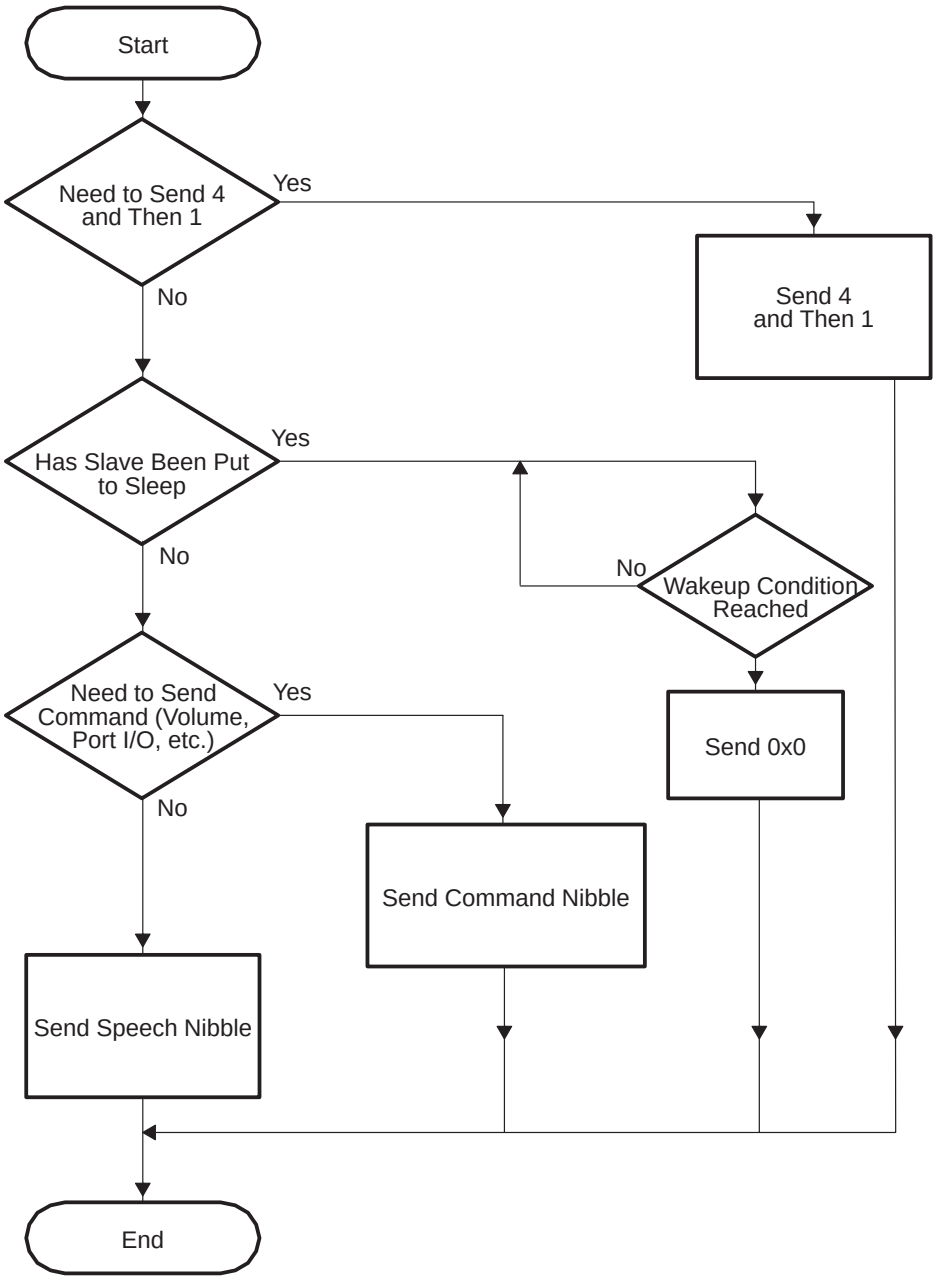
Figure A–4. Routine for Sending Data or Commands to the Slave



A.1.2 Flowchart for the Master to Play CELP/MELP

All of the flowcharts for playing mixed mode apply, except for the one associated with the ISR tied to the falling edge of $\overline{\text{INRDY}}$. Sending speech only is a subset of playing mixed mode. The previous flowcharts handle speech only. The simplified ISR associated with $\overline{\text{INRDY}}$ low is shown in Figure A–5.

Figure A–5. Program Flow for ISR Tied to the Falling Edge of $\overline{\text{INRDY}}$ to Play CELP/MELP Only



A.1.3 PC Parallel Port Reference Code to Play CELP/MELP

Section A.2 provides C reference code to play CELP/MELP files through a PC's parallel port. The C code interfaces with the MSP53C691 through the parallel port of the PC and uses a polling method to transfer data to the MSP53C691. Though the PC code uses a polling method to transfer data to the MSP53C691, it is recommended that data transfer be done through interrupt service routines in the actual master microcontroller. The PC code is a very basic code to describe the protocol and is provided for reference only. The code does not interleave commands to the MSP53C691 with speech data or recognize any error codes returned by the slave.

The following points must be considered while using the PC C code to communicate with the slave.

- The C code communicates with the slave through the parallel port. The code asks users for the address of the parallel port of their PC. Users must specify the address (in HEX) to be able to communicate properly.
- The PC's parallel port must be in bidirectional mode for the code to work. Users must make sure that the parallel port of the PC is set in either bi-directional mode or ECP mode in the BIOS.
- A schematic of the hardware connections between the parallel port and MSP53C691 is shown in Figure 4–1. Connect parallel port hardware with the MSP53C691 as described. Also, provide power and oscillator circuit to the MSP53C691 and connect a speaker or audio amplifier to the DAC pins of the MSP53C691. There are two low-cost hardware boards available through authorized TI distributors. These boards allow users to connect their PC's parallel port to the MSP53C691. Details about the hardware boards, the connection procedure, and the software is described in appendix C, *Data Preparation*.
- The PC code accepts speech files in binary format. The SDS6000 software (available from TI) takes a WAV file as an input, processes the file, and creates a compressed ASCII file either in MELP or CELP. This file must be converted to binary format if it is to be used as an input to the PC code.
- The PC code accepts the following format as the command argument :604pc filename.bin

The executable asks the user for the address of the PC parallel port. Users must specify the address of the parallel port (in HEX) to be able to communicate properly with the MSP53C691. The code then streams the speech data into the MSP53C691, and the MSP53C691 returns 0x55 when the speech file is finished playing.

The PC reference code for playing MELP or CELP is given in the *Parallel Port C Reference Code* section of this Appendix.

The downloadable .exe file is provided under device's application notes. A tool (.exe file) is also provided in the application notes which converts the .byt file created by the SDS6000 to a binary format suitable for the 604pc code.

A.2 Parallel Port C Reference Code

```
/************************************************************
* 604pc.c
*
* Sends speech files to the 604 via the parallel port. This file
* uses polling for transferring data to the slave from the PC.
* It is recommended that the interrupt service routines of the
* master microcontroller are used to transfer the data to the
* slave. The PC code is provided as a reference code to follow
* the protocol of the interface.
*
************************************************************/
#include <time.h>
#include <stdio.h>
#include <string.h>
#include <stdlib.h>
#include <windows.h>
#include <winbase.h>
// Defines
#define      DEBUG           0
#define      VERSION         ”1.00”
#define      MAXBINSIZE      524288
#define      CMDSPEAK        4
#define      CMDADPCM        8
// Define bit masks for R/W and CE
#define RW      0x02
#define CE      0x01
// Define masks for use when writing
// The mnemonics have two parts.  The first part gives the state of R/W and
// the second gives the state of CE. They are masked and inverted to produce
// the desired values when used with the parallel port control register.
#define RWHCEH  ~(0xFB)
#define RWHCEL  ~(0xFB & ~CE)
#define RWLCEH  ~(0xFB & ~RW)
#define RWLCEL  ~(0xFB & ~RW & ~CE)
// Define masks for use when reading
// The difference between the read and write masks is in bit 5, which
// controls the direction of the Data port.Setting bit 5, causes the port to
// work as input.
#define RRWHCEH  ~(0xDB)
#define RRWHCEL  ~(0xDB & ~CE)
#define RRWLCEH  ~(0xDB & ~RW)
#define RRWLCEL  ~(0xDB & ~RW & ~CE)
// Defines for calibration routine
#define      TEST_VALUE          1000
#define      TEN_USEC            10.0 * 0.000001
#define      CALIBRATE_LOOPS     100
// Type Definition
typedef      unsigned char    byte;
// Global Variables
```

```
int       timebasecount,overheadcount;
int       ParData,ParStatus,ParControl;
int       wait30us;
int       timeout;
int       numbytes;
int       thisbyte,sample;
byte      speechdata[MAXBINSIZE];
byte      txcount[MAXBINSIZE];
byte      temp_hi, temp_lo;
// Declarations
int       speak(void);
void      sendbyte(byte cmd);
int       recvbyte(void);
int       calibrate(int *tb, int *ovrhd);
void      MySleep(unsigned int sleep_time); // one of Frank's
void      DeviceReset(void);
void      setECP(void);
int main(int argc, char *argv[])
{
   FILE   *inbin;
   char   binfname[256];
   int    stat;
   printf("PC-604 Code  Version %s\n",VERSION);
   _outp(ParControl,RWHCEH); // start off with RW and CE both high
   calibrate(&timebasecount,&overheadcount);  // calibrate the timecount
   wait30us = 3 * timebasecount - overheadcount;
   timeout = 500000;
// Check the number of arguments given
   if (argc < 2)
   {
      printf("Usage : 604pc [...] <binspeechfile>\n");
      exit(0);
   }
   strcpy(binfname,argv[argc-1]);
   // Open the file
   if ((inbin = fopen(binfname,"rb")) == NULL)
   {
      printf("Error opening binary file <%s>\n",binfname);
      exit(0);
   }
// Read the number of bytes from the file
   numbytes = fread(&speechdata[0],sizeof(byte),MAXBINSIZE,inbin);
   fclose(inbin);
   printf("Read %d bytes\n",numbytes);
// Define parallel port registers
   printf("What is your Parallel port address (in HEX)?\n");
   scanf("%x",&ParData);
   ParStatus=ParData+1;      // Parallel port status register
   ParControl=ParData+2;     // Parallel port Control register
   // Set the parallel port for ECP mode for bidirectional mode
```

```
    setECP();
//  RESET the device
    DeviceReset();
// Waiting for 30ms before start polling for INRDY low is necessary
    Sleep(30);
    // Wait for INRDY to go low ––– loop here as long as INRDY is high
       do {
       stat = _inp(ParStatus) & 0x30;
       }  while ((stat & 0x10) == 0x10);   // INRDY low OUTRDY high
// Waiting for 30ms before start polling for INRDY low is necessary
    Sleep(30);
    sendbyte(0XB); // send clock command
    // Wait for INRDY to go low ––– ––– loop here as long as INRDY is high
       do {
       stat = _inp(ParStatus) & 0x30;
       }  while ((stat & 0x10) == 0x10);   // INRDY low OUTRDY high
    sendbyte(2); // send command for Crystal mode select
    SetPriorityClass(GetCurrentProcess(), HIGH_PRIORITY_CLASS);
    speak();
    SetPriorityClass(GetCurrentProcess(), NORMAL_PRIORITY_CLASS);
    return(0);
}
int speak()
{
    int    i,hi;
    int    stat;
    short int OUTRDY1,OUTRDY2,OUTRDY3,OUTRDY4;

    hi = 1;
       // Wait for INRDY to go low ––– ––– loop here as long as INRDY is high
       do {
       stat = _inp(ParStatus) & 0x30;
       }  while ((stat & 0x10) == 0x10);   // INRDY low OUTRDY high
    sendbyte(CMDSPEAK); // send speak command
    // Wait for INRDY to go low ––– ––– loop here as long as INRDY is high
       do {
       stat = _inp(ParStatus) & 0x30;
       }  while ((stat & 0x10) == 0x10);   // INRDY low OUTRDY high
       sendbyte(1); // send speak command parameter for 4 bit transfer
// loop forever –– break out when the slave would send something –– hopefully
// the termination value
    for (i=0;;i++)
    {
       if (hi == 1)  {
temp_hi = speechdata[i] & 0xF0; // read a bytes from the speech file and
                                //store
temp_hi = temp_hi >> 4; // position the high nibble so that it goes through the
                        // DATA0–3 lines
temp_hi = temp_hi | 0x10;  // Make DATA4 line high to signify speech data
temp_lo = speechdata[i] & 0x0F;  // Store the low nibble and position it for
                                 //the DATA0–3 lines
```

```
temp_lo = temp_lo | 0x10;  // Make DATA4 line high to signify speech data
       }
// Check for INRDY low or OUTRDY low ---- loop until one of them is low
          do {
          stat = _inp(ParStatus) & 0x30;
          }  while (stat == 0x30);
// If OUTRDY is low --- then read the data (4 nibbles)
             if(((stat & 0x10) == 0x10) || (stat == 0))  {
             OUTRDY1 = recvbyte();      // 1st nibble
             OUTRDY1 = OUTRDY1<<4;
// Wait for OUTRDY to go low again
             do {
             stat = _inp(ParStatus) & 0x30;
             }  while  (((stat & 0x20) != 0x0));
             OUTRDY2 = recvbyte();  // read 2nd nibble
             OUTRDY1 = OUTRDY1+OUTRDY2;
             OUTRDY1 = OUTRDY1<<4;
// Wait for OUTRDY to go low again
             do {
             stat = _inp(ParStatus) & 0x30;
             }  while  (((stat & 0x20) != 0x0));
             OUTRDY3 = recvbyte();  // read 3rd nibble
             OUTRDY1 = OUTRDY1+OUTRDY3;
             OUTRDY1 = OUTRDY1<<4;
// Wait for OUTRDY to go low again
             do {
             stat = _inp(ParStatus) & 0x30;
             }  while  (((stat & 0x20) != 0x0));
             OUTRDY4 = recvbyte();  // read 4th nibble
             OUTRDY1 = OUTRDY1+OUTRDY4;
             printf("status returned = %x\n",OUTRDY1);
          break;
       }
// Check if INRDY is low
       if ((stat & 0x10) != 0x10) {
             if (hi == 1) { // decide if the high nibble to be sent
                sendbyte(temp_hi);  // send the high nibble
                hi = 0;
                --i;   }
             else   {               // else send low nibble
                sendbyte(temp_lo);  // send low nibble
             hi = 1;      }
             }
   }
// reconfigure the parallel port as it was before
   _outp(ParData+0x402,0x00);
   return(0);
}
void sendbyte(byte databyte)
{
```

```
// RW is already low, take STROBE low, apply data, take STROBE high
   _outp(ParControl,RWLCEL);  // drop RW and STROBE
   _outp(ParData,databyte);   // send a byte of data
   _outp(ParControl,RWLCEH);  // drop RW, raise STROBE
}
int recvbyte()
{
   int    x;
   _outp(ParControl,RRWHCEH); // bit5 1, RW high, STROBE high
   _outp(ParControl,RRWHCEL); // bit5 1, RW high, STROBE high
   x = _inp(ParData);         // copy data to x
   _outp(ParControl,RRWHCEH); // bit5 1, RW high, STROBE high
   _outp(ParControl,RWHCEH);  // bit5 0, RW high, STROBE high
   return(x);
}
// This routine calibrates according to the speed of the PC
// This calibration is used to insert a predetermined amount of delay
// into the code with the MySleep routine. This routine
// and the MySleep routine doesn't have much use for this code but can
// be used with very fast PC to insert delays for microsecond precision
int calibrate(int *tb, int *ovrhd)
{
   LARGE_INTEGER PerfCount1, PerfCount2, PerfFreq;
   double Frequency;
   unsigned int time_base, overhead, total, i,return_priority;
   QueryPerformanceFrequency(&PerfFreq);
   if (PerfFreq.QuadPart == 0)
   {
printf ("This machine does not support high performance counters\n");
      exit(-1);
   }
   Frequency = 1.0/PerfFreq.QuadPart;
   time_base = (unsigned int)((TEN_USEC/Frequency) + 0.5);
   total = 0;
   for (i=0; i< CALIBRATE_LOOPS; i++)
   {
   SetPriorityClass(GetCurrentProcess(), REALTIME_PRIORITY_CLASS);
   __asm pushfd
   __asm cli
   QueryPerformanceCounter(&PerfCount1);
   MySleep(0);
   QueryPerformanceCounter(&PerfCount2);
   __asm sti
   __asm popfd
   SetPriorityClass(GetCurrentProcess(), NORMAL_PRIORITY_CLASS);
   overhead = (unsigned int)(PerfCount2.QuadPart - PerfCount1.QuadPart);
   total += overhead;
   }
#ifdef DEBUG
   printf ("MySleep (0) consumes an average of %d\n", total/CALIBRATE_LOOPS);
```

```
    printf (”MySleep (0) consumes %d ticks\n”, overhead);
    printf (”One Performance Counter tick = %g seconds\n”, Frequency);
    printf (”To wait ten usec, wait for %g ticks\n”, (TEN_USEC/Frequency));
#endif
    overhead = total/CALIBRATE_LOOPS;
    *tb = time_base;
    *ovrhd = overhead;
    return_priority=SetPriorityClass(GetCurrentProcess(),NORMAL_PRIOR-
ITY_CLASS);
    return(time_base);
}
// This routine can be used to insert a certain amount of delay in micro-
   seconds
void MySleep(unsigned int sleep_time)
{
    LARGE_INTEGER PerfCount1, PerfCount2, goal;
    int return_priority;
    QueryPerformanceCounter(&PerfCount1);
    goal.QuadPart = PerfCount1.QuadPart + sleep_time;
    do
    {
       QueryPerformanceCounter(&PerfCount2);
    } while (PerfCount2.QuadPart < goal.QuadPart);
    return_priority=SetPriorityClass(GetCurrentProcess(),NORMAL_PRIOR-
ITY_CLASS);
}
// RESET the device
    void DeviceReset()  {
    int control;
    control=_inp(ParData+2);
    _outp(ParData+2,control & 0xFB);
    Sleep(10);
    control=_inp(ParData+2);
    _outp(ParData+2,control | 0x04);
    control=_inp(ParData+2);
    }
// Set the parallel port in the ECP mode
    void setECP()  {
    _outp(ParData+0x402,0x20);
    }
```

Appendix B

FM Synthesis

This appendix discusses FM synthesis.

Topic **Page**

B.1 FM Synthesis Overview

FM synthesis is a technique for creating harmonically rich musical tones in a relatively simple manner. Generally speaking, the tones generated do not closely correspond to the tonal texture of conventional instruments; but they can be used to generate interesting and pleasant music.

The MSP53C691 can generate one channel of FM synthesis. This appendix presents the command formats used to describe the music.

B.2 FM Synthesis Format and Data Preparation

B.2.1 FM Synthesis Format and Commands

The song to be played is coded into a file in a specified format. This file contains a series of BYTE or DATA statements to specify the notes, instruments, and other details of the music.

Each command has the general form:

BYTE command, parameters,0

Where *command* indicates the action to be taken, followed by one or more modifying parameters. For example, to transpose a section of a song up by a semitone, the following command would be written:

```
BYTE RTRNS,1
```

In this example, RTRNS is the command and 1 is the modifying parameter.

As another example, to play a note, use the following command:

BYTE C1,n4,n4,127

This example commands the synthesizer to play a C note for a quarter note duration at the maximum volume.

The formats and parameters for the different commands are described in the following sections.

The various commands are defined in the file FM source file.

B.2.2 Musical Notes

Musical notes are defined as:

```
BYTE    Notevalue, TimeValue,Duration,Velocity
```

Where:

Notevalue defines the pitch of the musical note. The valid values for the note are defined in the programming code included at the end of this appendix. In general, they range from a minimum of C1 to a maximum of C6. The sequence within each octave is: C, Cs, or Db, D, Ds, or Eb, E, F, Fs, or Gb, G, Gs, or Ab, A, As, or Bb, B. For example, the C sharp in the first octave is written as Cs1. This is the same tone as Db1.

TimeValue defines the duration of the note that is played. The valid values for TimeValue are defined in the programming code included at the end of this appendix. The more common values are: n8, which defines an eighth note; N4, which defines a quarter note; N2, which defines a half note; N1, which defines a whole note.

Duration defines the sum of TimeValue and any following rest. For example, if a quarter note is followed by a quarter note rest, then it must be coded with a TimeValue=n4 and a Duration=n2.

Velocity is the relative volume of the note. Values range from 0 to 127.

B.2.3 Tempo Control

The tempo of the music is defined as:

```
BYTE    TEMPO,BPM,TimeSig,EnvelopeLen
```

Where:

TEMPO is the command that indicates a tempo change is being defined.

BPM is the new tempo. The valid values for BPM are defined in the programming code included at the end of this appendix. An example is bpm62, which indicates 62 beats per minute.

TimeSig defines the time signature of the songs. TS44 sets the time signature to 4/4 time.

EnvelopeLen defines the length of the note envelope. ENVOK sets the envelope length to normal (i.e., lasting for a whole note).

B.2.4 Transposition

Two commands are available for transposing the music (i.e., uniformly shifting the notes to higher or lower frequencies). The ATRNS command adds a specific amount to the note value. The RTRNS command adds a relative amount to the note value.

```
BYTE    ATRNS, NUM
```

Shifts the music NUM semitones from the value as written, for example:

```
BYTE    ATRNS,12
```

Shifts the music one octave above the music as written.

The RTRNS command is used to shift the music by a cumulative amount, for example in the following sequence:

```
BYTE   RTRNS,12  ;This will shift the music up by one octave
BYTE   RTRNS,12  ;This will shift the music up a second octave
BYTE   DETUNE,5     ;Add 5 to the frequency of channel 5
```

B.2.5 Adjust Output Volume

The FADER command is used to scale the volume of the notes played.

```
BYTE      FADER, InitialFaderValue,FaderInc
```

Where:

FADER is the signal whose output volume is to be adjusted by the command.

InitialFaderValue is the new sound volume.

FaderInc allows a gradual transition to the new volume. It is a signed two-byte value that specifies the incremental amount of volume change during each interval.

For example:

```
BYTE   FADER,f100p     ;Set volume to 100
DATA   NOFADER      ;Change is abrupt
```

B.2.6 Modulation Index Adjustment

It is frequently desirable to incrementally change the texture of the sound quality in the song. This can be done by changing the modulation index to get a more or less brighter tonal quality. This can be done by using the following commands:

```
BYTE   MIX1     ; Set the modulation index scale to 1
BYTE   MIX2     ; Set the modulation index scale to 2
BYTE   MIX3     ; Set the modulation index scale to 3
BYTE   MIX4     ; Set the modulation index scale to 4
BYTE   MIX5     ; Set the modulation index scale to 5
BYTE   MIX6     ; Set the modulation index scale to 6
BYTE   MIX7     ; Set the modulation index scale to 7
BYTE   MIX8     ; Set the modulation index scale to 8
BYTE   MIXUP    ; Increment the modulation index scale
BYTE   MIXDN    ; Decrement the modulation index scale
```

B.2.7 End of Song

STOPSONG is used to signal the end of the song.

```
BYTE   STOPSONG    ;Signal the end of song
```

B.2.8 Command Summary

Table B–1 summarizes the several valid commands.

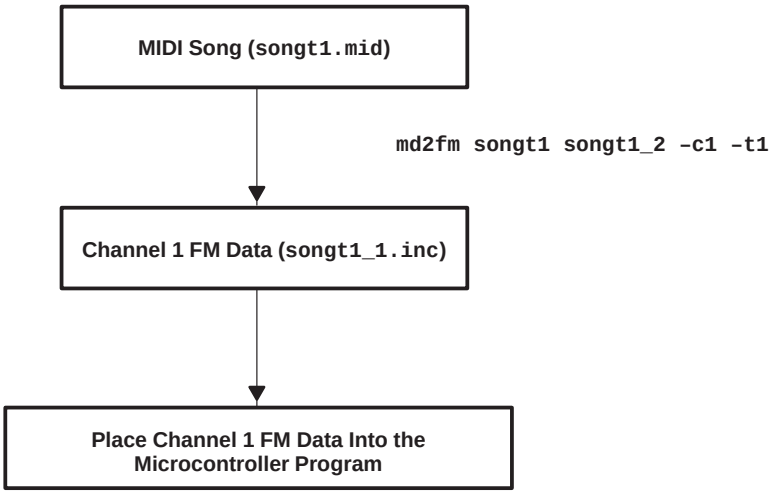
Table B–1.Command Summary

Command and Format	Description
Music Notes: Format: BYTE Note,TimeValue,Duration,Velocity Example: BYTE C1, n4, n4, 127	Note: Is the music note. It can range form C0 to C6. TimeValue: Total length of the note. n4 is 1/4 note. Duration: Length of tone generate Velocity: Note volume from 0 to 127
Tempo control the speed of music: Format: BYTE TEMPO,BPM,TimeSig,EnvelopeLen Example: TEMPO,BPM116,TS44,ENVOK	TEMPO: Tempo command. Set the song tempo in channel 1 only. BPM: Beats per minutes. TimeSig: Beats per measure. TS44 sets the time signature to 4/4. EnvelopLen: Envelop time. ENVOK sets the envelope length to normal.
Transpose: Format: BYTE ATRNS, NUM Example: BYTE ATRNS,–12 (–12 = Transpose Down an Octave)	ATRANS: Transpose command. NUM: Set the channel transpose to a signed offset.
Transpose: Format: BYTE RTRNS,NUM Example: BYTE RTRNS,7 (Add 7 Semitones to the channel's transpose offset)	RTRNS: Transpose command. NUM: Add a signed offset to the channel transpose value.
Fade control: Format: BYTE FADER,InitialFaderValue,FaderInc Example: BYTE FADER,f100p, –32 (word)	FADER: Fader command InitalFaderValue: Set initial fader value from 0 to 63. F100p is defined in FmequM2.inc FadInc: Fader increment, which is a 16-bit value. Calculate as follows: $\frac{(\text{End Fader Value} - \text{Start Fader Value}) \times 16}{\#\ \text{of Events}}$
Mix control: Format: BYTE MixLevel Example: MIX8	MixLevel: Set the modulation index value by lookup table from MIX0 to MIX15
Mix control: Format: BYTE MIXUP Example: MIXUP	MIXUP: Increment the current modulation index as set by MIXn.
Mix control Format: BYTE MIXDN Example: BYTE MIXDN	MIXDN: Decrement the current modulation index as set by MIXn.
End of the song: Format: StopSong Example: StopSong	StopSong: Stop playing the song.

B.3 Data Preparation of FM Synthesis

According to the discussion on FM data format, a song can be coded following the predefined command and formats. Alternatively, software utilities are available for converting a song from musical instrument digital interface (MIDI) formatted files to a format accepted by MSP53C691. MD2FM.exe is available for the conversion. The process for the conversion is shown in Figure B–1:

Figure B–1. FM Conversion Process



By using MD2FM, the channel 1 data of a MIDI file (.mid) can be extracted and converted to FM format (.inc). Then it can be put into the MCU program for FM synthesis.

B.3.1 MD2FM Software

MD2FM converts a MIDI format file to FM data accepted by MSP53C691. With this routine users can compose or translate music base on the MIDI format. This routine runs under the DOS environment and uses the following syntax:

```
   md2fm songt1 songt1_1 –c1 –t1
input : songt1.mid (MIDI format)
output   : songt1_1.inc (FM format)
–c1   : channel #1 to be decoded†
–t1   : track #1 to be decoded†
```

† Specify the number of tracks and channels to be extracted and converted. Only one channel within one track can be converted on the MIDI file at any one time.

Assuming that the songt1.mid contains two tracks and each has one channel music data, the FM data can be extracted as follows:

```
   md2fm songt1 songt1_1 –c1 –t1
```

The output file songt1_1.inc is the track 1 data in FM format.

Note that more than one note appearing on the same track and channel may cause error in the conversion. Thus, a song with one note is recommend for the conversion.

Since MD2FM does not support all the features of MIDI, the following must be noted:

1) Due to the limitations of the conversion program, the TIMEBASE of the .mid file must be 48 and there must not be any TEMPO/METER changes after the initial settings. Also, the conversion program does not recognise channel pressure control.

2) This routine creates a .inc file for a single channel of a single track only from a simple .mid file.

3) Since the instruments supported by TI FM synthesis are different from the MIDI editors, users must select the instruments for channel 1 and 2 according to the instrument list. Refer to section B–4 for a complete list of available instruments. A portion of the list is shown below.

```
*  Lookup values for instrument sound tables:
PatchFLT1   equ   128+0 ;FM flute tone 1
PatchBRS1   equ   128+1 ;FM brass tone 1, medium slow
                        ;attack
PatchBRS2   equ   128+2 ;FM brass tone 2, fast attack
PatchBRS3   equ   128+3 ;FM brass tone 3, slow attack
PatchTRM1   equ   128+4 ;FM brass tone trombone 1, slow
                        ;attack
PatchTRM2   equ   128+5 ;FM brass tone trombone 2, med
                        ;slow attack
PatchCLR1   equ   128+6 ;FM clarinet tone 1
PatchCLR2   equ   128+7 ;FM clarinet tone 2, brighter
                        ;than CLR1
PatchMT1a   equ   128+8 ;FM metallic tone 1a
...
...
```

For example, the line defining the instruments should be found both in channel 1 and channel 2 data as follows:

```
...
BYTE     PatchMT6i          ;AcouGrandPiano
...
```

If PIANO1 is selected for channel 1, the line must be replaced by the following line, which can be found on section B–4.

At channel 1 data,

```
...
BYTE     PatchPNO1
...
```

B.3.2 Procedure for Converting MIDI to MSP53C691 FM

The following conversion method involves the use of asmx.exe, which is a compiler for MSP50C3x. Note that is only used as a tool for converting all the symbols to actual data.

1) Extract the channel data form the MIDI file using the MIDI2FM.EXE DOS executable.

2) The new file created by the MIDI2FM executable puts a default PATCHxxx for the instrument of the MIDI file. This default patch can be replaced by

the instrument as chosen by the user from section B–4. A list of available instruments is given in this file. Replace all the instances of the PATCH with the desired instrument.

3) Run the .INC file with the executable ASMX.EXE. This creates a binary equivalent of the .INC file. This file can be supplied to the MSP53C691 as an FM file.

Note:

If you need an ASCII representation of the binary FM file, run the executable BINBYTE.EXE with the binary file as input. The executable creates an output file which is an ASCII representations of the binary file. This ASCII file can be included in your microcontroller assembly code if the assembler of your microcontroller converts the ASCII file to binary. Usage of the BINBYTE.EXE–Binbyte 0x input_binary_filename output_filename

B.4 FM II—Constant Definitions

```
************************************************
* FM II
************************************************
*  Constant Definitions
*            DC,BA98:7654,3210
*            --,----:----,----
BIT0      equ #0001  ;00,0000:0000,0001
BIT1      equ #0002  ;00,0000:0000,0010
BIT2      equ #0004  ;00,0000:0000,0100
BIT3      equ #0008  ;00,0000:0000,1000
BIT4      equ #0010  ;00,0000:0001,0000
BIT5      equ #0020  ;00,0000:0010,0000
BIT6      equ #0040  ;00,0000:0100,0000
BIT7      equ #0080  ;00,0000:1000,0000
BIT8      equ #0100  ;00,0001:0000,0000
BIT9      equ #0200  ;00,0010:0000,0000
BIT10     equ #0400  ;00,0100:0000,0000
BIT11     equ #0800  ;00,1000:0000,0000
BIT12     equ #1000  ;01,0000:0000,0000
BIT13     equ #2000  ;10,0000:0000,0000
*  Initialization defaults
*  default gain value
MAXGAIN      equ 24 ;default value, also MAX. do not exceed!
*  default Master Modulation Index Scale values
DEFSCLMIX equ 96  ;like a tone control...
*  Song interval delay values
ONESEC    equ 1*10
TWOSECS      equ 2*10
THREESECS equ 3*10
FOURSECS  equ 4*10
FIVESECS  equ 5*10
TENSECS      equ 10*10
*  FM channel Automated Fader calculations
*  Coded as:
*  BYTE   FADER, CurrFader, ((DestFader–CurrFader) * 16) / #Events
*  Init Fader with Start Gain * 16 = 384
*  So, when each new event comes along, Add FaderInc to Fader
*  (EXTSG ON) and Update Fader.
*  When calculating Loudness, use Fader / 16 * Current Signal.
*  My standard fader values
f100p     equ 63
f94p      equ 60
f87p      equ 56
f75p      equ 48
f62p      equ 40
f50p      equ 32
f37p      equ 24
f25p      equ 16
f18p      equ 12
```

```
f12p      equ 8
f9p    equ 6
f6p    equ 4
f4p    equ 3
f3p    equ 2
f2p    equ 1
f0p    equ 0
OFF    equ 0      ;use for REST event, set velocity = 0
NOFADER   equ 0   ;Fader Increment = 0.
*  Musical note index definitions
C0     equ 0
Cs0    equ 1
Db0    equ 1
D0     equ 2
Ds0    equ 3
Eb0    equ 3
E0     equ 4
F0     equ 5
Fs0    equ 6
Gb0    equ 6
G0     equ 7
Gs0    equ 8
Ab0    equ 8
A0     equ 9
As0    equ 10
Bb0    equ 10
B0     equ 11
C1     equ 12
Cs1    equ 13
Db1    equ 13
D1     equ 14
Ds1    equ 15
Eb1    equ 15
E1     equ 16
F1     equ 17
Fs1    equ 18
Gb1    equ 18
G1     equ 19
Gs1    equ 20
Ab1    equ 20
A1     equ 21
As1    equ 22
Bb1    equ 22
B1     equ 23
C2     equ 24
Cs2    equ 25
Db2    equ 25
D2     equ 26
Ds2    equ 27
Eb2    equ 27
```

```
E2      equ 28
F2      equ 29
Fs2     equ 30
Gb2     equ 30
G2      equ 31
Gs2     equ 32
Ab2     equ 32
A2      equ 33
As2     equ 34
Bb2     equ 34
B2      equ 35
C3      equ 36
Cs3     equ 37
Db3     equ 37
D3      equ 38
Ds3     equ 39
Eb3     equ 39
E3      equ 40
F3      equ 41
Fs3     equ 42
Gb3     equ 42
G3      equ 43
Gs3     equ 44
Ab3     equ 44
A3      equ 45
As3     equ 46
Bb3     equ 46
B3      equ 47
C4      equ 48
Cs4     equ 49
Db4     equ 49
D4      equ 50
Ds4     equ 51
Eb4     equ 51
E4      equ 52
F4      equ 53
Fs4     equ 54
Gb4     equ 54
G4      equ 55
Gs4     equ 56
Ab4     equ 56
A4      equ 57
As4     equ 58
Bb4     equ 58
B4      equ 59
C5      equ 60
Cs5     equ 61
Db5     equ 61
D5      equ 62
Ds5     equ 63
```

```
Eb5    equ 63
E5     equ 64
F5     equ 65
Fs5    equ 66
Gb5    equ 66
G5     equ 67
Gs5    equ 68
Ab5    equ 68
A5     equ 69
As5    equ 70
Bb5    equ 70
B5     equ 71
C6     equ 72
RST    equ 73
REST      equ 120
*  Lookup values for Instrument Sound tables
PatchFLT1 equ 128+0  ;FM Flute tone 1
PatchBRS1 equ 128+1  ;FM Brass tone 1, Medium slow attack
PatchBRS2 equ 128+2  ;FM Brass tone 2, Fast attack
PatchBRS3 equ 128+3  ;FM Brass tone 3, Slow attack
PatchTRM1 equ 128+4  ;FM Brass tone Trombone 1, Slow attack
PatchTRM2 equ 128+5  ;FM Brass tone Trombone 2, Med slow attack
PatchCLR1 equ 128+6  ;FM Clarinet Tone 1
PatchCLR2 equ 128+7  ;FM Clarinet Tone 2, brighter than CLR1
PatchMT1a equ 128+8  ;FM Metallic tone 1a
PatchMT1b equ 128+9  ;FM Metallic tone 1b
PatchMT1c equ 128+10 ;FM Metallic tone 1c
PatchMT2a equ 128+11 ;FM Metallic tone 2a
PatchMT2b equ 128+12 ;FM Metallic tone 2b
PatchMT2c equ 128+13 ;FM Metallic tone 2c
PatchMT3a equ 128+14 ;FM Metallic tone 3a
PatchMT3b equ 128+15 ;FM Metallic tone 3b
PatchMT3c equ 128+16 ;FM Metallic tone 3c
PatchMT4a equ 128+17 ;FM Metallic tone 4a
PatchMT4b equ 128+18 ;FM Metallic tone 4b
PatchMT4c equ 128+19 ;FM Metallic tone 4c
PatchMT5a equ 128+20 ;FM Metallic tone 5a
PatchMT5b equ 128+21 ;FM Metallic tone 5b
PatchMT5c equ 128+22 ;FM Metallic tone 5c
PatchMT6a equ 128+23 ;FM Metallic tone 6a, good BASS sound 1
PatchMT6b equ 128+24 ;FM Metallic tone 6b, good BASS sound 2
PatchMT6c equ 128+25 ;FM Metallic tone 6c, good BASS sound 3
PatchMT6d equ 128+26 ;FM Metallic tone 6d
PatchMT6e equ 128+27 ;FM Metallic tone 6e
PatchMT6f equ 128+28 ;FM Metallic tone 6f
PatchMT6g equ 128+29 ;FM Metallic tone 6g
PatchMT6h equ 128+30 ;FM Metallic tone 6h
PatchMT6i equ 128+31 ;FM Metallic tone 6i, hard metallic sound 1
PatchMT6j equ 128+32 ;FM Metallic tone 6j, hard metallic sound 2
PatchMT6k equ 128+33 ;FM Metallic tone 6k, hard metallic sound 3
```

```
PatchMT6l equ 128+34       ;FM Metallic tone 6l, plucked string 1
PatchMT6m equ 128+35       ;FM Metallic tone 6m, plucked string 2
PatchMT6n equ 128+36       ;FM Metallic tone 6n
PatchMT6o equ 128+37       ;FM Metallic tone 6o, plucked string 3
PatchMT6p equ 128+38       ;FM Metallic tone 6p, plucked string 4
PatchCHM1 equ    128+39    ;FM Chimes tone 1
PatchCHM2 equ    128+40    ;FM Chimes tone 2
PatchCHM3 equ    128+41    ;FM Chimes tone 3
* Extra instument   AUG 99
PatchPNO1 equ    128+42    ;FM Piano tone 1
PatchPNO2 equ    128+43    ;FM Piano tone 2
PatchEP_1 equ    128+44    ;FM Electric Piano tone 1
PatchEP_2 equ    128+45    ;FM Electric Piano tone 2 (more aggressive)
PatchBNJ1 equ    128+46    ;FM Banjo 1
PatchGTR1 equ    128+47    ;FM Guitar 1
PatchHRP1 equ    128+48    ;FM Harp 1
PatchEGT1 equ    128+49    ;FM Electric Guitar 1
PatchXYL1 equ    128+50    ;FM Xylophone
PatchVIB1 equ    128+51    ;FM Vibraphone
PatchFLT12 equ   128+52    ;FM Flute Tone 1
PatchCLR12 equ   128+53    ;FM Clarinet Tone 1
PatchOBO1 equ    128+54    ;FM Oboe 1
PatchHRN1 equ    128+55    ;FM French Horn 1
PatchBSN1 equ    128+56    ;FM Bassoon 1
PatchTBA1 equ    128+57    ;FM Tuba 1
PatchTRB1 equ    128+58    ;FM Trombone 1
PatchTPT1 equ    128+59    ;FM Trumpet 1
PatchVLN1 equ    128+60    ;FM Violin 1
PatchEBS1 equ    128+61    ;FM Electric Bass 1
*  Max # of patch codes is 64
CONTROL      equ #80 ;< is NOTE data, >= are commands and controls
*  Codes #80 to #BF are reserved for Patch Codes (packed BYTE)
PATCH     equ #80 ;usage: BYTE PATCH+patchcode, ie 80+MT6o = B7
*  Codes #C0 to #FF are reserved for Commands
CMDS      equ #C0 ;192 to 255 are Commands
*
*  Codes #C0 to #D3 available for user expansion.
*
StopSong  equ #D4 ;End of Song
GOTO      equ #D5 ;Control code for GOTO a Label
RETURN    equ #D6 ;Control code for Return from a Subroutine
GOSUB     equ #D7 ;Control code for Play a Subroutine
DETUNE    equ #D8 ;Control code for Ch2 Detune, Signed offset
TEMPOSYNC equ #D9 ;Control code for Sync Ch2/Ch1 Tempo change
MIXDN     equ #DA ;Control code for Shift MIX value DOWN
MIXUP     equ #DB ;Control code for Shift MIX value UP
RTRNS     equ #DC ;Control code for RELATIVE Transpose
ATRNS     equ #DD ;Control code for ABSOLUTE Transpose
TEMPO     equ #DE ;Control code for Tempo
FADER     equ #DF ;Control code for Set Fader
```

```
MIX0      equ#E0;Control code for Modix (Modulation Index)
MIX1      equ#E1;isolate bits 0–3 for table lookup
MIX2      equ#E2;
MIX3      equ#E3;
MIX4      equ#E4;
MIX5      equ#E5;
MIX6      equ#E6;
MIX7      equ#E7;
MIX8      equ#E8;
MIX9      equ#E9;
MIX10     equ#EA;
MIX11     equ#EB;
MIX12     equ#EC;
MIX13     equ#ED;
MIX14     equ#EE;
MIX15     equ#EF;
ENDREP    equ#F0;Control code for End Repeat
BEGREP2      equ#F1;Control code for Start Repeat, Play 2X
BEGREP3      equ#F2;Control code for Start Repeat, Play 3X
BEGREP4      equ#F3;Control code for Start Repeat, Play 4X
BEGREP5      equ#F4;Control code for Start Repeat, Play 5X
BEGREP6      equ#F5;Control code for Start Repeat, Play 6X
BEGREP7      equ#F6;Control code for Start Repeat, Play 7X
BEGREP8      equ#F7;Control code for Start Repeat, Play 8X
ENDMULTI  equ#F8;Control code for End MULTI Repeat
BEGMULTI2 equ#F9;Control code for Start MULTI, Play 2X
BEGMULTI3 equ#FA;Control code for Start MULTI, Play 3X
BEGMULTI4 equ#FB;Control code for Start MULTI, Play 4X
BEGMULTI5 equ#FC;Control code for Start MULTI, Play 5X
BEGMULTI6 equ#FD;Control code for Start MULTI, Play 6X
BEGMULTI7 equ#FE;Control code for Start MULTI, Play 7X
BEGMULTI8 equ#FF;Control code for Start MULTI, Play 8X
*  Tempo values as 1/4 notes (Beats) Per Minute, 12 Clocks per Beat
*           Index    MyPsc  Exact_BPM  Beat_Prd  Exact_Tmr  Timer
BPM61     equ0  ;20   61.035156  0.983040  196.608000  197
BPM67     equ1  ;22   67.138672  0.893673  178.734545  179
BPM73     equ2  ;24   73.242188  0.819200  163.840000  164
BPM79     equ3  ;26   79.345703  0.756185  151.236923  151
BPM85     equ4  ;28   85.449219  0.702171  140.434286  140
BPM92     equ5  ;30   91.552734  0.655360  131.072000  131
BPM98     equ6  ;32   97.656250  0.614400  122.880000  123
BPM104    equ7  ;34  103.759766  0.578259  115.651765  116
BPM110    equ8  ;36  109.863281  0.546133  109.226667  109
BPM116    equ9  ;38  115.966797  0.517389  103.477895  103
BPM122    equ10 ;40  122.070313  0.491520   98.304000  98
BPM128    equ11 ;42  128.173828  0.468114   93.622857  94
BPM134    equ12 ;44  134.277344  0.446836   89.367273  89
BPM140    equ13 ;46  140.380859  0.427409   85.481739  85
BPM146    equ14 ;48  146.484375  0.409600   81.920000  82
BPM153    equ15 ;50  152.587891  0.393216   78.643200  79
```

```
BPM159    equ 16  ;52  158.691406  0.378092   75.618462  76
BPM165    equ 17  ;54  164.794922  0.364089   72.817778  73
BPM171    equ 18  ;56  170.898438  0.351086   70.217143  70
BPM177    equ 19  ;58  177.001953  0.338979   67.795862  68
BPM183    equ 20  ;60  183.105469  0.327680   65.536000  66
BPM189    equ 21  ;62  189.208984  0.317110   63.421935  63
BPM195    equ 22  ;64  195.312500  0.307200   61.440000  61
BPM201    equ 23  ;66  201.416016  0.297891   59.578182  60
BPM208    equ 24  ;68  207.519531  0.289129   57.825882  58
BPM214    equ 25  ;70  213.623047  0.280869   56.173714  56
BPM220    equ 26  ;72  219.726563  0.273067   54.613333  55
BPM226    equ 27  ;74  225.830078  0.265686   53.137297  53
BPM232    equ 28  ;76  231.933594  0.258695   51.738947  52
BPM238    equ 29  ;78  238.037109  0.252062   50.412308  50
BPM244    equ 30  ;80  244.140625  0.245760   49.152000  49
BPM250    equ 31  ;82  250.244141  0.239766   47.953171  48
BPM256    equ 32  ;84  256.347656  0.234057   46.811429  47
BPM262    equ 33  ;86  262.451172  0.228614   45.722791  46
BPM269    equ 34  ;88  268.554688  0.223418   44.683636  45
BPM275    equ 35  ;90  274.658203  0.218453   43.690667  44
BPM281    equ 36  ;92  280.761719  0.213704   42.740870  43
BPM287    equ 37  ;94  286.865234  0.209157   41.831489  42
BPM293    equ 38  ;96  292.968750  0.204800   40.960000  41
BPM299    equ 39  ;98  299.072266  0.200620   40.124082  40
BPM305    equ 40  ;100 305.175781  0.196608   39.321600  39
*  Time Signature constants as Negative UP Counter values.
Qnote     equ –12 ;–12 clocks per 1/4 note, 1/4 = 1 Beat
TS24      equ 2*Qnote   ;2 beats per Measure
TS34      equ 3*Qnote   ;3 beats per Measure
TS44      equ 4*Qnote   ;4 beats per Measure
TS54      equ 5*Qnote   ;5 beats per Measure
TS64      equ 6*Qnote   ;6 beats per Measure
TS74      equ 7*Qnote   ;7 beats per Measure
*  SIGNED Offsets added to MyPsc to Shorten/Lengthen Envelope times
ENVOK     equ 0   ;No Envelope Length adjust
ENVS1     equ 1   ;make Envelope length Shorter by 1
ENVS2     equ 2   ;make Envelope length Shorter by 2
ENVS3     equ 3   ;make Envelope length Shorter by 3
ENVS4     equ 4   ;make Envelope length Shorter by 4
ENVS5     equ 5   ;make Envelope length Shorter by 5
ENVS6     equ 6   ;make Envelope length Shorter by 6
ENVS7     equ 7   ;make Envelope length Shorter by 7
ENVS8     equ 8   ;make Envelope length Shorter by 8
ENVS10    equ 10  ;make Envelope length Shorter by 10
ENVS12    equ 12  ;make Envelope length Shorter by 12
ENVS14    equ 14  ;make Envelope length Shorter by 14
ENVS16    equ 16  ;make Envelope length Shorter by 16
ENVS18    equ 18  ;make Envelope length Shorter by 18
ENVS20    equ 20  ;make Envelope length Shorter by 20
ENVL1     equ –1  ;make Envelope length Longer by 1
```

```
ENVL2     equ -2  ;make Envelope length Longer by 2
ENVL3     equ -3  ;make Envelope length Longer by 3
ENVL4     equ -4  ;make Envelope length Longer by 4
ENVL5     equ -5  ;make Envelope length Longer by 5
ENVL6     equ -6  ;make Envelope length Longer by 6
ENVL7     equ -7  ;make Envelope length Longer by 7
ENVL8     equ -8  ;make Envelope length Longer by 8
ENVL10    equ -10 ;make Envelope length Longer by 10
ENVL12    equ -12 ;make Envelope length Longer by 12
ENVL14    equ -14 ;make Envelope length Longer by 14
ENVL16    equ -16 ;make Envelope length Longer by 16
ENVL18    equ -18 ;make Envelope length Longer by 18
ENVL20    equ -20 ;make Envelope length Longer by 20

*  Musical Note Time value definitions (FM)
ngrc      equ 1   ;Grace Note, use with n8m etc
n163      equ 2   ;1/16 note triplet
n16    equ 3   ;1/16 note
n83    equ 4   ;1/8  note triplet
n8m    equ 5   ;1/8  note off beat (use for Shuffle feel)
n8     equ 6   ;1/8  note
n8p    equ 7   ;1/8  note on beat  (use for Shuffle feel)
n43    equ 8   ;1/4  note triplet
nd8    equ 9   ;.1/8 note
n4     equ 12  ;1/4  note
n23    equ 16  ;1/2  note triplet
nd4    equ 18  ;.1/4 note
n2     equ 24  ;1/2  note
nd2    equ 36  ;.1/2 note
n1     equ 48  ;1/1  note
nd1    equ 72  ;.1/1 note
n21    equ 96  ;2/1  note
n31    equ 144 ;3/1  note
n41    equ 192 ;4/1  note
n51    equ 240 ;5/1  note
*  End of FMEQU.INC
```

Appendix C

Editing Tools and Data Preparation

This appendix discusses the SDS6000 software editing tool and data preparation for the MSP53C691 using different algorithms.

Topic **Page**

C.1 Editing Tools

TI provides an integrated software tool (SDS6000) which can be used for MELP editing and CELP and MELP encoding.

C.1.1 SDS6000

The SDS6000 is an integrated software tool that accepts sound file inputs (sampled at either 8 kHz, 10 kHz, or 11 KHz) in .WAV format and converts the files into CELP or MELP data format. To launch the SDS6000, encode a WAV file in MELP/CELP and playback the encoded file, users need the following hardware and software:

1) SDS6000 software and hardware (see the developmental tool section in the TI's main web page (www.ti.com) for details.

2) A personal computer running Windows 95/98

3) One 32-Ω speaker (if wanting to connect direct drive) with its own enclosure, or an 8-Ω speaker with its own amplifier and enclosure, an 8-Ω speaker with its own enclosure if wanting to use the onboard LM386, or the four transistor amplifiers of the hardware.

Download the SDS6000 software through the Texas Instruments speech website at http://www.ti.com/sc/speech. E-mail Texas Instruments at speak2me@list.ti.com for problems accessing the software or help with the software.

Hardware tools 3 and 4 can be ordered through an authorized TI distributor.

C.2 Data Preparation

The MSP53C691 slave synthesizer supports several speech synthesis algorithms. The speech data sent to the slave device must match the format defined by TI or generated by a TI's tool (SDS6000 for MELP, CELP, or ADPCM). The data preparation for the MSP53C691 using different algorithms is discussed in the following paragraphs.

C.2.1 ADPCM

A WAV file can be encoded in ADPCM through the SDS6000 software tool. The bit rate for ADPCM is: (sampling rate)*16 bits/4. For example, an 8-kHz WAV file produces 32 kbps of encoded data.

C.2.2 MELP and CELP

The SDS6000 is used to convert an input audio data file into MELP or CELP data formats. The input data file must be in .WAV format. The audio data must be sampled at either 8 kHz, 10 kHz, or 11KHz and must have 16-bit-signed precision.

See the *Tips for Optimal TI Speech* appendix for details about recording and preprocessing a file.

Note:

While processing a music file in CELP, it is advisable to check the Music button before processing. The file is processed at a slightly higher bit rate in this selection so enhancing the quality of the compressed file. Please see the HELP in SDS6000 for details.

The sound files must start and stop at a level close to zero; errors may result otherwise. The volume of the sound files must also be adjusted to a peak-to-peak level around 0.5 V to –0.5 V (assume full scale is 1 V to –1 V). Clipping may result if the sound file is too loud after the encoding. Additionally, if the wave file is sampled at a higher rate (CD quality sound file 44.1-kHz sampling or DAT in 48-kHz), resampling to 8 kHz or 10 kHz is necessary before the conversion. Filtering and renormalizing may be necessary during downsampling to reduce aliasing and noise.

There are several software programs available to perform this resampling function. Two commonly available examples are GoldWave and Cool Edit Pro. GoldWave is a shareware program that provides editing function on sound files in a Windows environment. This software can be downloaded from the following web site: http://www.goldwave.com.

To resample the original wave/sound file, the high-frequency portion must be filtered first to eliminate the aliasing in resampling. All frequencies above one half of the final sampling rate must be removed from the sound file. For example, before converting the sampling rate to 8 kHz, the data must be filtered to remove all frequency components above 4 kHz. Some sound editing tools

have poor low-pass filters. A sound editing tool with a good low-pass filter is recommended.

After preprocessing the input WAV file as described above and saving it in a WAV format through a sound editing tool, the file is ready to be processed through the SDS6000 for encoding in MELP, CELP, or ADPCM. See the software HELP menu for detailed operation of the SDS6000. For more information about the tool, e-mail TI at speak2me@list.ti.com. The following is an overview of the hardware setup and other necessary steps required to process a file through the SDS6000 in MELP, CELP, or ADPCM. Check the SDS6000 setup instructions in the tools development section of the main TI web page (www.ti.com).

Hardware Setup

1) Connect the parallel port cable with the parallel port of the pc and with the hardware.

2) Connect the power supply to the hardware.

Software

1) Unzip and install the SDS6000 software.

2) Launch the SDS6000 software by double-clicking on the icon.

3) Click Accept when the license agreement screen pops up. The SDS6000 software will try to communicate with the hardware.

4) When the software is running, select Open a New Project. Select the files you want to compress, then click the button to select the compression technique you want to use (MELP, CELP, or ADPCM).

5) The status of the encoding process is displayed in the messages window. You can play the encoded file through your connected audio circuit by right clicking the mouse button on the file and selecting Play on Chip.

After successful completion of the encoding and decoding processes, the SDS6000 makes two files: an ASCII representation of the encoded file with BYT extension, and a WAV file representing the encoded file decoded in WAV format.

The ASCII file or its binary representation is used as the compressed speech data for the MSP53C691. The ASCII file can be used in the master microcontroller assembly code. The assembler of the master microcontroller must convert the .BYT file to binary after assembling the master code. The MSP53C691 expects the speech data from the master in binary format.

Troubleshooting TIPS for SDS6000

If the SDS6000 software can communicate with the hardware board, the software starts and the SDS600 Speech Development window displays. If the window does not appear, verify the following:

1) The parallel port in your PC is set to either bidirectional, ECP, or EPP mode in the BIOS.

2) All the cables are properly connected.

3) The board is powered up.

4) There are no other devices connected to the parallel port(s) of the PC.

C.2.3 FM

Music can be coded manually or can be converted from MIDI files. For manual coding, refer to Appendix B for the data format of FM synthesis. If the song is composed in MIDI format (.mid), it can be converted to FM by the DOS executable routine MD2FM.EXE. Several limitations are imposed on the MIDI files processed by the MD2FM program.

For details about the process of converting MIDI files into FM files, refer to the *Data Preparation of FM Synthesis* section in Appendix B.

Appendix D

Pitch and Speed Shifting for 6xx MELP

This appendix discusses MELP speech pitch and speed shifting.

Topic **Page**

D.1 Pitch Shifting

Pitch shifting gives an interesting texture to the voice encoded in MELP. It can make a voice sound lower or higher in frequency depending upon the parameter supplied with the pitch-shifting command.

Pitch shifting adds an integer value to the encoded pitch parameter (pitch table offset) of a frame. The pitch parameter has upper and lower limits based on the size of the pitch decoding table. The pitch shift bias integer (a parameter of the pitch shift command) is added to the pitch parameter (read in from the bitstream of the MELP speech file). For example, if the pitch for the first frame is 10 and the pitch shift bias is 2, the pitch is decoded using a table offset of 12 (lower voice than originally encoded). Similarly, if the pitch for the first frame is 10 and the pitch shift bias is –2, the pitch is decoded using a table offset of 8 (higher voice than originally encoded).

If the pitch shift bias modifies the pitch table offset to an out of bounds pitch index, then the pitch is shifted only as far as the boundaries allow. Thus, if the pitch is 10 and the pitch shift bias is –11 (10 + –11 = –1), the pitch is decoded using a table offset of 0. The same applies to the upper bound (length of the pitch decoding table).

Since pitch shifting affects the pitch every time a frame is decoded, it can be performed once every frame. For uncompressed speech, new changes to the pitch can occur every 15 ms. For compressed speech, new pitch changes can occur each time an *encoded* frame is decoded. Thus, if only half the speech frames are encoded, the pitch shift bias parameter can be changed every 30 ms (since each frame is 15 ms long).

The pitch shifting in MELP can be achieved by sending the pitch shift command followed by the four nibble parameter. The command for pitch shift is 0xA followed by the parameter 0x2. The value of the parameter for the amount of pitch shift is then transferred to the MSP53C691 in four nibbles (most significant nibble first).

The following table summarizes how to control the pitch by sending the four nibble parameter of the pitch shift command.

Table D–1. Pitch Control

Parameter	Result
Zero value	No pitch shifting
Positive value	Voice sounds lower in frequency (deeper, towards male)
Negative value	Voice sounds higher in frequency (higher, towards female)

To make speech sound higher, make the parameter a negative number using the command:

```
0xFFF8  ; –8 twos complement
```

sent to the MSP53C691 as F, F, F, and 8.

To make speech sound lower, make the parameter a positive number, for example,

0x0009

sent to the MSP53C691 as 0, 0, 0 and 9.

D.2 Speed Shifting

The speed of MELP speech can be increased or decreased by sending the speed shift command followed by the four-nibble parameter specifying the amount of speed shift. The number of subframes per frame can be adjusted to increase or decrease the speed of the speech. For example, if the algorithm calculates 10 subframes per frame, it is possible to lengthen the speech by repeating subframes. Similarly, one can speed up the speech by skipping subframes. This is the premise for speed shifting on the 6xx MELP algorithm.

The resolution of the increase or decrease in speed is determined by the number of skipped or repeated subframes. The 6xx MELP algorithm only allows lengthening the speech by two (repeat every other subframe) or to reducing it by half (skip every other subframe). In other words, the fastest increase in speed is twice as fast as the normal speech and the slowest decrease in speed is twice as slow as normal. Use the following formulas to calculate the resolution of speed shifting in terms of percentages:

Again, the transfer function looks much different than our standard form in Equation 1. Make the following substitutions:

$$\frac{2}{\text{subframeBetweenSkips}} \times 100 = \text{percentfaster}$$

$$\frac{1}{\text{subframeBetweenRepeats}} \times 100 = \text{percentslower}$$

Thus, the maximum percentfaster is 100%, and the maximum percentlower is 50%.

Here are the first few steps of shifting resolution that can possibly modify variable *subframesBetweenSkips:*

2 * 1 / 2 = 50% faster (twice as fast)	100% faster (= twice as fast)
2 * 1 / 3 = 33.3% faster	66% faster
2 * 1 / 4 = 25% faster	50% faster
2 * 1 / 5 = 20% faster	40% faster
2 * 1 / 6 = 16.7% faster	33% faster

and so on.

Variable: *subframesBetweenRepeats can be modified by:*

1 / 2 = 50% slower (twice as slow)
1 / 3 = 33.3% slower
1 / 4 = 25% slower
1 / 5 = 20% slower
1 / 6 = 16.7% slower

and so on …

See the following table for a summary on speed shifting.

Table D–2. Speed Shifting Summary

Speed shift command	First parameter (1 nibble)	Second parameter (4 nibbles)	Result
0xA	0x1	> 1	Speed increases
0xA	0x0	0	Normal speed
0xA	0x0	< –1	Speed decreases

The value of the second parameter can range from 0x2 to 0x7FFF. While increasing the speed, the parameter 0x2 represents the highest speed. Similarly, while decreasing the speed, the parameter 0x2 represents the slowest speed. To detect noticeable difference in the speed, the range of the second parameter must be between 0x2 and 0x20.

To increase the speed by 100%, use the following sequence of commands (including the parameter):

A, 1, 0, 0, 0, and 2.

Note: The fifth bit of the data line is held low while sending the command or the parameters.

Note: The parameters set for a certain speed or pitch shift are destroyed and must be reset to play speech for a different vocoder.

Appendix E

Guidelines for Optimal TI Speech

This appendix discusses the guidelines for obtaining optimal TI speech such as voice talent selection, recording techniques and equipment, and input signal processing.

Topic **Page**

E.1 Voice Talent Selection

Selecting the right voice talent to record the speech for a product is extremely important. The following criteria must be considered when choosing a voice talent: professional versus amateur, gender-related voice characteristics, dialect, and pronunciation.

Professional VS Amateur Voice Talent—Professional voice talent (usually union members) is recommended over amateur voice talent for the following reasons: better pitch and amplitude control, endurance, and better recording demeanor (such as avoiding extraneous noise and movement).

Gender-Related Voice Characteristics—Avoid female voice talent with *breathy* voice (such as American actress Marilyn Monroe), especially when using MELP.

Avoid very low-pitched male voices with prominent vocal fry (such as American Exsecretary of State Henry Kissinger and American actor Vincent Price). *Vocal fry* has a low fundamental frequency range (30–90 Hz) and is characterized by *jitter* (irregular timing in glottal cycles). This voice quality is used in Arabic and Swedish speech sounds, and it often occurs in English at the end of an utterance.

Dialect—The voice talent must be a native speaker of the standard dialect of a given language; that is, the dialect spoken by anchor persons reporting the news on radio or television. Examples of standard dialects are the London dialect of British English, the Madrid dialect of Castilian Spanish, the Cali (Colombia) dialect of Latin American Spanish, the Parisian dialect of French, etc.). The dialects do not exhibit features that allow the listener to identify the speaker as belonging to any given geographical region of socioeconomic or ethnic group.

Pronunciation—Avoid voice talent with perceptible speech impediments or distracting qualities such as whistled or lisped [s] or [z].

E.2 Auditions

Audition several voice talents to determine which voice and speaking style processes the best with the selected TI vocoder (an official character voice is an exception to this). If possible, have the product decision maker present.

E.2.1 Audition Material

Try to record material similar to that which will be used in the product. Include phrases containing sounds that may not process well.

E.2.2 Recording Script

If you have to create a recording script, keep the following considerations in mind.

Vocabulary—Start with a list of the desired vocabulary, which is generally provided by the customer.

Context (*Pronunciation*)—When dealing with context pronunciation, the following principles must be considered:

- *Sentence Frame*—The beginning and end of the target vocabulary word must not be altered by the sounds in adjacent words. To accomplish this, it is best to add a neutral sound such as schwa (pronounced as *uh*) on either side of the target word.

- *Position of Target Word*—A word spoken in isolation is approximately twice as long as the same word occurring in a sentence (except in final position, where duration is the same as in isolation).

  Try to minimize the occurrence of the target word in utterance–final position, since the rate of vocal-fold vibration decreases as an utterance is terminated, resulting in partial unvoicing and vocal (glottal) fry, which sounds like a squeaky door opening or closing. The first sound in utterance-initial position, especially a voiced stop ([b,d,g] as in *<u>b</u>all*, *<u>d</u>oll*, and *<u>g</u>all*), may also be distorted. Put a neutral vowel before the first sound in the utterance such as *a <u>ball</u> uh, a <u>doll</u> uh, a <u>gall</u> uh.*

- Pitch Contour (Intonation)—Put the target word in different positions in the sentence or phrase for several versions of the word with different intonation patterns (such as rising, level, falling). For example:
  It's a one again (falling contour)
  Is it a one again? (rising contour)
  It's a one, a two, a three. . .(level or *continuation* contour)

  To cause the speaker to use different intonation patterns, sometimes it is useful to put the target word utterance-final position and then to add extra words onto the end. For example, to get a *continuation* contour (see above) as in *It's Saturday. . .*, add words such as *three o'clock P.M*. so that the utterance becomes, *It's Saturday (1s) three o'clock P.M*.

E.3 Speech Rate and Pausing

Speech Rate—The speech rate must be moderate. As a speaker speaks slower, the pause time (silence) between words increases and sounds are articulated more precisely. The speech rate should not be so slow, though, as to distort features such as pitch (intonation) and energy (perceived as loudness).

Pausing—Insert two-second pauses in the script between the item number (such as 2a) and item *a ball uh*, as well as between the item and the next item number to help you capture data more easily. Insert pauses (marked with //) within phrases to remind the speaker to slow down at specific points.

E.4 Reference Numbers

Use reference numbers to group the contexts of a given vocabulary word:

SET 1: NUMBERS

1a	(2s)	*One* (2s)
1b	(2s)	*You have // one // saved // message*. (2s)
	.	
	.	
	.	
3a	(2s)	*Three* (2s)
3b	(2s)	*You have // three // new // messages.* (2s)

The sound engineer should say the item number (such as 1a) and give some direction before the voice talent says each item. The voice talent should say whatever is placed within quotation marks (“” or «») on the recording script.

E.5 Recording

Sound Booth—Recording in a sound booth (anechoic chamber) is essential for high-quality synthetic speech. The larger the sound booth, the better. Eliminate any sources of ventilation or environmental noise. Prevent reverberation by making sure that the voice talent is not directly facing a hard surface (such as glass window, hard-surfaced wall, or metal script stand). Use a baffle (or substitute) behind the voice talent.

Audio Equipment—Choosing the right microphone and audio equipment enhances the quality of the speech, reduces editing time, and decreases the time-to-market for Texas Instruments vocoders. A rubberized shockmount prevents microphone vibration and a windscreen prevents *explosive plosives* (such as initial [p]).

- ❏ *Microphone*—The following are examples of microphones that have been shown to work well in speech processing with TI vocoders: Samson Audio S12 Hypercardioid Neodymium microphone and the Audio Technica 4033 SM cardioid condenser microphone.

- *Microphone Preamplifier*—Choose a good microphone preamplifier (such as Martech).

- *Other Audio Equipment*—Consider a good analog-to-digital converter (such as Lucid), a PCI card, a mixer for analog monitoring and routing (such as Mackie), and a good playback system (such as Mackie).

Recording Session—Important considerations for the recording session include microphone placement, audio input levels, voice talent direction, and data capture techniques.

- *Microphone Placement*—A free-standing microphone must be approximately eight inches from the voice talent's lips. The microphone must be pointed toward the voice talent's chin. Use a shockmount to prevent microphone vibration. The voice talent must be instructed not to move his/her head during the recording since varying the distance from the microphone results in varying amplitude (energy level) in the signal.

- *Input levels*—Set audio input levels to prevent clipping the signal during recording (anywhere from around –3 dB to –12 dB, depending on the content to be recorded). Do not apply equalization, filtering, etc. at this point, since the aim is to get a very *clean* unclipped signal.

- *Voice Talent Direction*—The voice talent must be instructed to avoid extraneous noise (such as page shuffling, scratching, etc.) while speaking. Ask the voice talent to practice reading the script while levels are being set and other final preparations are being made. Be sure that final consonants (such as [m,n] and [b, d, g]) are articulated distinctly. Answer questions and adjust speaking rate and vocal effort if necessary, correct mistakes and give some initial direction regarding the number of *takes* and how each take must be said (for example, Take 1–Energetic, Take 2–Clear articulation, etc.). Maintain uniform pitch and loudness levels within and across recording sessions.

Data Capture—Record using a sampling rate of at least 44.1 kHz or 48 kHz. Save the recorded speech data as 16-bit, mono, signed integer files of the following type:

.WAV Windows wave format (preferred)

During the Recording Session—Sample different takes to preprocess and analyze with the SDS6000.

E.6 Preprocessing the Input Signal

Preprocessing (that is, *small amounts* of digital multiband processing and analog compression, amplitude normalization, and graphic equalization) can enhance the quality of the speech, reduce editing time, and reduce the time-to-market for Texas Instruments synthesizers.

Amplitude Normalization—Be sure the wave file is fully amplified (100%). *Normalize* (reduce) the amplitude by approximately 15% (~3 dB) with a good wave editor (such as Cool Edit Pro®). It is a good idea to save this file under a different name to indicate that it has been down-sampled and normalized. This distinguishes the normalized file from the original wave file.

Down-Sampling—Using a wave-editing tool with good filtering, such as Cool Edit Pro® or Matlab®, down-sample .WAV files from 48 kHz (preferred) or 44.1 kHz to 8 kHz or 10 kHz, respectively. A higher sampling rate improves the quality but increases the size of the data.

Note:

Some wave-editing tools may introduce noise when down-sampling. The quality of the results depends upon the quality of the *filter*.

Editing the Wave File—Border-editing (i.e., deleting silence at the beginning and end of the signal) must also be performed with a wave-editing tool. The following editing steps can be applied:

- ❑ Remove artifacts (such as transients, etc.) and extraneous noise from the signal.
- ❑ Replace unacceptable speech sounds (such as slurred or distorted sounds) with correct sounds from somewhere else in the file.
- ❑ Selectively boost the amplitude of weak fricative sounds *[f, v, th (voiced and unvoiced), and z]*, if needed. Cut the amplitude of strong sibilants (*s, sh, ch, j*), if needed).

Graphic Equalization—The synthesis process tends to boost the lower frequencies and attenuate the higher frequencies in a signal. Sometimes the effect is a *muffled* or *muddy* quality. Therefore, wave files of such voices may require graphic equalization to increase or decrease the gain for certain frequencies. With the graphic equalizer of a wave editing program, the amplitude of higher frequencies (~2–4 kHz and above) can be boosted while the amplitude of lower frequencies (below ~100 Hz) can be attenuated.

Note:

Use graphic equalization as a last resort, since this process can cause the voice to process more poorly.

E.7 Processing Speech on the SDS6000

Use the following procedure to process speech using the SDS6000:

1) Create a new project in the SDS6000.
2) Add the normalized down-sampled .WAV files to the project.
3) Choose the type of analysis to process the files (such as MELP or CELP).
4) Select the appropriate bit rate or compression ratio.
5) Analyze the file(s).
6) Listen to the file(s) and experiment with different analysis types (MELP or CELP), different bit rates or compression ratios, and different sampling rates (8 kHz or 10 kHz).
7) If the quality is poor for a large portion of the file(s), you have some options:
   - Make selective amplitude adjustments
   - Use a little graphic equalization on the .WAV file
   - Ask the voice talent to adjust his/her voice quality
   - Use a different voice talent
   - Change some aspect(s) of the recording setup (such as microphone, microphone placement, etc.)

E.8 Editing Speech on the SDS6000

E.8.1 General Rules for Speech Editing

Observe the following rules while editing speech using the SDS6000:

- ❏ Ignore the *letters* that spell the words you are editing. Listen to the sounds. In this manual, speech sounds are represented by International Phonetic Alphabet symbols (IPA).
- ❏ Fix the worst (most obvious) problems first. Change values as little as possible (that is, make small changes).
- ❏ Edit speech through the product enclosure in which the speech is to be played.

E.8.2 Editing MELP Speech on the SDS6000

MELP Parameter Sets—MELP analysis creates one analysis file containing three sets of parameters: uncoded, coded, and compressed, with the compression factor defined in the MELP Setup dialog box.

- Uncoded MELP—There is really no *uncoded MELP* synthesizer. When you edit uncoded MELP parameters, the changes are automatically transferred to the coded parameters. When you click on *Uncoded* on the play pad, you hear the simulation of MELP synthesis using the coded parameters.
- Coded MELP—When you edit coded parameters, the changes are automatically passed to the compressed parameters, because the only difference is in the *send bit*. When you edit compressed parameters, you are, in effect, editing coded parameters.
- Compressed MELP—Using the Edit menu's Compress command on any file (uncoded, coded, or compressed) works on the coded parameters and computes the *send [frame] bit* for every frame.

MELP Editing Parameters—MELP synthesized speech requires less editing than LPC speech. However, some problems may still arise, especially with a high degree of compression.

Three parameters can be edited in the text editor for coded MELP:

- Gain—Energy or A0 value refers to the loudness of the signal
- Pitch—F0 or T0 value refers to the frequency of a sound
- Voicing (BPVC)—Band-pass voicing refers to the degree of voicing in a frame. A value of 1 or 2 is close to unvoiced (0), while 3 is moderate voicing, and 4 or 5 are voiced.

Also:

- Send—*Send bit* refers to file compression. If the file is uncompressed (that is, its compression ratio is set to 1.000), the send bit has a value of 0. If a file is compressed (that is, its compression ratio is set to less than 1.000, such as 0.600), then the send bit has a value of 1. This parameter cannot be edited.

The commands in the Modify menu can also be used: Zero Out, Linear Interpolation, % Change, and Compress. The commands Select, Copy, Cut, and Paste are also available for editing MELP. All of these commands work just like they do for LPC editing.

For uncoded MELP, the text editor contains the following parameters:

- Pitch—F0 or T0 value refers to the frequency of a sound
- Gain—Energy or A0 value refers to the loudness of a signal
- BPVC—*Band-pass voicing* refers to the degree of voicing in a frame. A value of 1 or 2 is close to unvoiced, while 3 is moderate voicing, and 4 or 5 are voiced.

- UV_Flag—*Unvoiced flag* can be used to make an unvoiced frame a voiced frame. A value of 0 represents an unvoiced frame, while a value of 1 represents a voiced frame.

- K1–K10—K-parameters correspond to cross-sectional areas of the vocal tract, which extends from the lips to the larynx. K-parameters are related to articulation of speech sounds. For more details on K-parameters, please see the Appendix.

Editing MELP Speech—Most MELP editing problems are related to gain or pitch. These parameters can be edited using the text editor or the graphic editor. Edit pitch and gain just as you would with LPC speech. Please see the LPC Editing section above for details of editing different sound classes. (NOTE: Changing BPVC (voicing) may create static noise in the signal.)

- Send—*Send bit* refers to file compression. If the file is uncompressed (that is, its compression ratio is set to 1.000), the send bit has a value of 0. If a file is compressed (that is, its compression ratio is set to less than 1.000, such as 0.600), then the send bit has a value of 1. This parameter cannot be edited.

The commands in the Modify menu can also be used: Zero Out, Linear Interpolation, % Change, and Compress. The commands Select, Copy, Cut, and Paste are also available for editing MELP. All of these commands work just like they do for LPC editing.

For uncoded MELP, the text editor contains the following parameters:

- Pitch—F0 or T0 value refers to the frequency of a sound

- Gain—Energy or A0 value refers to the loudness of a signal

- BPVC—*Band-pass voicing* refers to the degree of voicing in a frame. A value of 1 or 2 is close to unvoiced, while 3 is moderate voicing, and 4 or 5 are voiced.

- UV_Flag—*Unvoiced flag* can be used to make an unvoiced frame a voiced frame. A value of 0 represents an unvoiced frame, while a value of 1 represents a voiced frame.

- K1–K10—K-parameters correspond to cross-sectional areas of the vocal tract, which extends from the lips to the larynx. K-parameters are related to articulation of speech sounds. For more details on K-parameters, please see the Appendix.

Editing MELP Speech—Most MELP editing problems are related to gain or pitch. These parameters can be edited using the text editor or the graphic editor. Edit pitch and gain just as you would with LPC speech. Please see the LPC Editing section above for details of editing different sound classes. (NOTE: Changing BPVC (voicing) may create static noise in the signal.)

E.9 Hardware Considerations in Product Design

E.9.1 Speaker Selection

The physical properties of the speaker have a substantial influence on the quality of the reproduction of speech. The diaphragm of the speaker should have a large surface area to efficiently produce low frequencies. The mass of the voice coil and diaphragm should be small to efficiently produce high frequencies. Most home speaker systems, for example, use a large woofer to produce sound below a few hundred Hertz, a tweeter to produce sound above 6–8 kHz, and a 4–6 inch diameter midrange speaker to produce sound between these ranges.

The most valuable portion of a speech signal is contained in the frequency range between 100 Hz and 5000 Hz. Therefore, it is possible to get reasonably good speech quality with a small speaker. The 2.25-inch diameter speaker is a common size used with TI speech devices.

Other speaker properties that affect how well the speaker reproduces sound are the stiffness and the material used for the diaphragm, the impedance of the voice coil, the strength of the magnet, and the depth of the speaker cone. A stronger magnet and a deeper cone are normally preferred, but these properties result in a larger speaker. The other properties depend on the complete system.

Each speaker has unique characteristics (for example, its frequency response) and it is important to understand how a speaker sounds in your product. The speaker characteristics are likely to change from speaker to speaker during the manufacturing process. We recommend being proactive in the speaker selection for your product, and not to change the speaker without a careful evaluation of the speech quality.

E.9.2 Speaker Enclosure Design

The speaker enclosure design is just as important as the speaker selection. This is especially true when using a small speaker to reproduce low-frequency sounds.

A speaker produces sound by increasing and decreasing air pressure at a rate equal to the frequency of the sound. The air pressure increases as the diaphragm moves toward the listener and decreases as it moves away from the listener. This assumes that the air displaced as the diaphragm moves forward does not fill the void behind the diaphragm. When the diaphragm is vibrating at higher frequencies, the air does not have time to rush around behind the diaphragm. This is the reason speaker efficiency at the higher frequencies is not as dependent on the speaker enclosure. However, this is not true for lower frequencies.

A speaker enclosure can be used to prevent air flowing from the front to the back of the speaker. The size and shape of the speaker enclosure affects the performance of the speaker. Two types of enclosures are commonly used: closed box and bass-reflex.

The closed box will most likely improve the low frequency response. However, the volume, minimum dimensions, rigidity, and absorption lining of the enclosure dramatically alter the performance.

The bass-reflex enclosure inverts the phase of the backside radiation of the speaker so that it adds to the front radiation. The result could increase the low-frequency efficiency by 5 dB over the closed box. This enclosure has the same complexities as the closed box, plus the effect of the port.

It may not be necessary to have a professionally designed speaker enclosure for your product. However, it is important that you understand how speaker enclosures can affect the reproduction of speech.

E.9.3 Speaker Drive and Audio Amplifier Buffer

The TI speech devices reproduce speech from a stream of numbers. Each number represents the sign and magnitude of the speech signal for a specific instant in time. These numbers are called samples, and the number of samples used to reproduce one second of speech is called the sample rate. The digital-to-analog converter (DAC) takes the number that represents the sign and magnitude of a sample and converts it to a sequence of voltage changes that represents the speech signal. This conversion from digital to analog requires an antialiasing (low-pass) filter to smooth the samples into a continuous representation of the analog signal.

The MSP53C691 has the capability to directly drive a 32-Ω speaker. The speaker is connected between two push-pull outputs. Each of these outputs can toggle between ground and the supply voltage. The DAC circuit controls when the outputs toggle. When the sample is zero, both outputs are in the same state. If the sample is positive, the output that causes positive current flow through the speaker toggles. When the sample becomes negative, the other output (the one that causes negative current flow through the speaker) toggles. The DAC controls the duty cycle or pulse density of these outputs for each sample, so that the average energy applied to the speaker reproduces the analog signal.

When a speaker is driven directly by digital outputs as described above, the sound produced contains the desired sounds plus some additional sounds. These additional sounds are created because the digital-to-analog conversion does not contain an antialiasing filter. The speaker may act like a low-pass filter and eliminate some of these sounds. In many cases, these additional sounds are not objectionable enough to warrant the additional cost of other components.

The major advantage of directly driving the speaker is reduced cost due to the avoidance of additional external components. In some cases, it may be worthwhile to provide an external buffer/amplifier between the TI speech device and the speaker. Potential advantages are increased output volume and increased signal fidelity due to the incorporation of a low-pass antialiasing filter.

The supply voltage and the speaker impedance are the primary factors in how much power is delivered to the speaker. The minimum speaker impedance (normally 32 Ω) and the maximum supply voltage are determined by the limits of the TI speech device. Adding an external amplifier/buffer allows a lower impedance speaker and a higher drive voltage to be used, resulting in increased speaker volume.

A buffer may be a simple circuit with four resistors, one nonpolar capacitor, and four transistors connected in an H-bridge configuration. This type of circuit buffers the DAC outputs so that a lower impedance speaker can be used. The cost and availability of the nonpolar capacitor may be the major disadvantage to this circuit.

The LM386 is another common audio amplifier used with TI speech devices. Implementing a good low-pass filter and analog volume control is easy and cost-effective when using this audio amplifier.

E.9.4 Impact of Batteries on Speech Quality

All batteries have an associated source impedance. This impedance is normally larger on smaller batteries (for example, button cells) than on larger batteries (for example, D-cell batteries). When the speech is being produced, the large currents driving the speaker may cause a significant sag in battery voltage. This modulation of the battery voltage may significantly affect both the output volume and the quality of the speech. In many cases, using a larger battery significantly improves the perceived quality of the product.