



CT8021

H.32x G.723.1/G.728

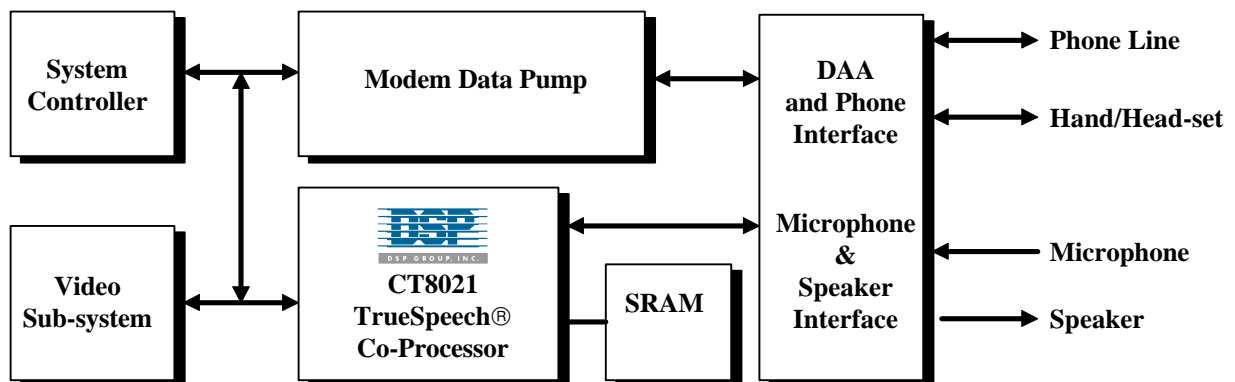
TrueSpeech Co-Processor

Introduction

The CT8021 is a speech co-processor which performs full duplex speech compression and decompression functions. It provides speech compression for H.320, H.323 and H.324 Multimedia Visual Telephony / Video Conferencing products and DSVD Modems. The CT8021 has built-in TrueSpeech® G.723.1 (for H.323 and H.324) as well as G.728 LD-CELP speech compression (for H.320). This combination of ITU speech compression standards within a single device enables the creation of a single multimedia terminal which can operate in all types of Video Conferencing systems including H.320 ISDN-based, H.324 POTS-based, and H.323 LAN/Internet-based. TrueSpeech® G.723.1 provides compressed data rates of 6.3 and 5.3 Kbps and includes G.723.1 Annex A VAD/CNG “silence” compression which can supply an even lower average bit rate. The CT8021 provides an additional two non-ITU TrueSpeech® data rates at 4.8 and 4.1 Kbps. G.728 provides LD-CELP speech compression at 16 Kbps. The CT8021 also supports download of additional speech compression software modules into low cost external memory (e.g. TrueSpeech® 8.5, G.722 and G.729A/B). The CT8021 is designed to operate as a micro-processor peripheral device and co-exist with other devices such as modem-datapump and video compression chipsets. In addition the CT8021 includes built-in Acoustical Echo Cancellation which complements the speech compression functions by providing concurrent hands-free operation. *-continued on pg. 2*

Features

- TrueSpeech® G.723.1 at 6.3, 5.3, 4.8 and 4.1 kbps at 8KHz sampling rate (including G.723.1 Annex A VAD/CNG)
- G.728 16 Kbps LD-CELP
- Download of additional speech compression software modules into external sram for TrueSpeech® 8.5, G.722 & G.729-A/B
- Real-time Full duplex or Half duplex speech compression and decompression
- Acoustic Echo Cancellation concurrent with full-duplex speech compression
- Full Duplex standalone Speakerphone
- Host-to-Host (codec-less) and Host-CODEC modes of operation
- Parallel 8-bit host interface provides simple memory-mapped I/O host connection.
- 1 or 2-channel DMA support (Single Cycle and Burst Modes)
- Flexible A-to-D/D-to-A CODEC interface, low cost PCM μ A-Law and 16-bit Linear
- CODEC interface supports TDM bus connection
- Automatic handling of frame slippage and frame synchronization
- Automatic AGC for message recording
- Speech level detection for silence compression support
- DTMF and Tone Generation
- DTMF and Programmable Tone Detection
- Pass-through modes - 8-bit and 16-bit linear at 8 KHz and 11 KHz (host programmable codec sample rate) and G.711 μ A-Law
- Power Down and Power Save modes



CT8021 System Block Diagram

PRELIMINARY
Version 1.1

The TrueSpeech® G.723.1 algorithm delivers very highly compressed speech without compromising the speech quality. TrueSpeech® G.723.1 at the 6.3 bit-rate has a MOS score of 3.9 for use with the ITU H.324 and H.323 standards. G.728 LD-CELP provides slightly higher speech quality but at the higher bit rate of 16 Kbits/sec as required by the ITU H.320 ISDN-based Video Conferencing Standard.

The CT8021, an Application Specific Digital Signal Processor, is controlled by the system's host processor through a simple host interface command protocol. The host interface supports full-duplex data transfer using DMA as well as host-interrupt and host-polling modes. The CT8021 supports two modes of uncompressed speech input/output. In HOST-CODEC mode, the uncompressed speech input/output is provided by one of the external serial codecs. In HOST-HOST (codec-less) mode, the Host provides the uncompressed speech input/output via the host interface.

The only additional external component ICs needed to implement these functions are two low cost 8Kx8 or 32Kx8 SRAMs, a CODEC and oscillator crystal circuit.

Note: MOS, Mean Opinion Score is a subjective measure of speech quality where a score of 5 means that the speech quality is Excellent, 4 is Good or Toll Quality (as expected in PSTN) and 3 is fair.

Applications

- Simultaneous Voice and Data Modems
- Teleconferencing and Video Conferencing
- CTI - Computer-Telephony Applications
- Digital Telephony Applications
- Voice Enabled Wireless Terminals
- Store & Forward applications for speech
- Desktop Telephony & Speakerphone
- Internet Telephony Applications

CT8021 Functional Block Diagram

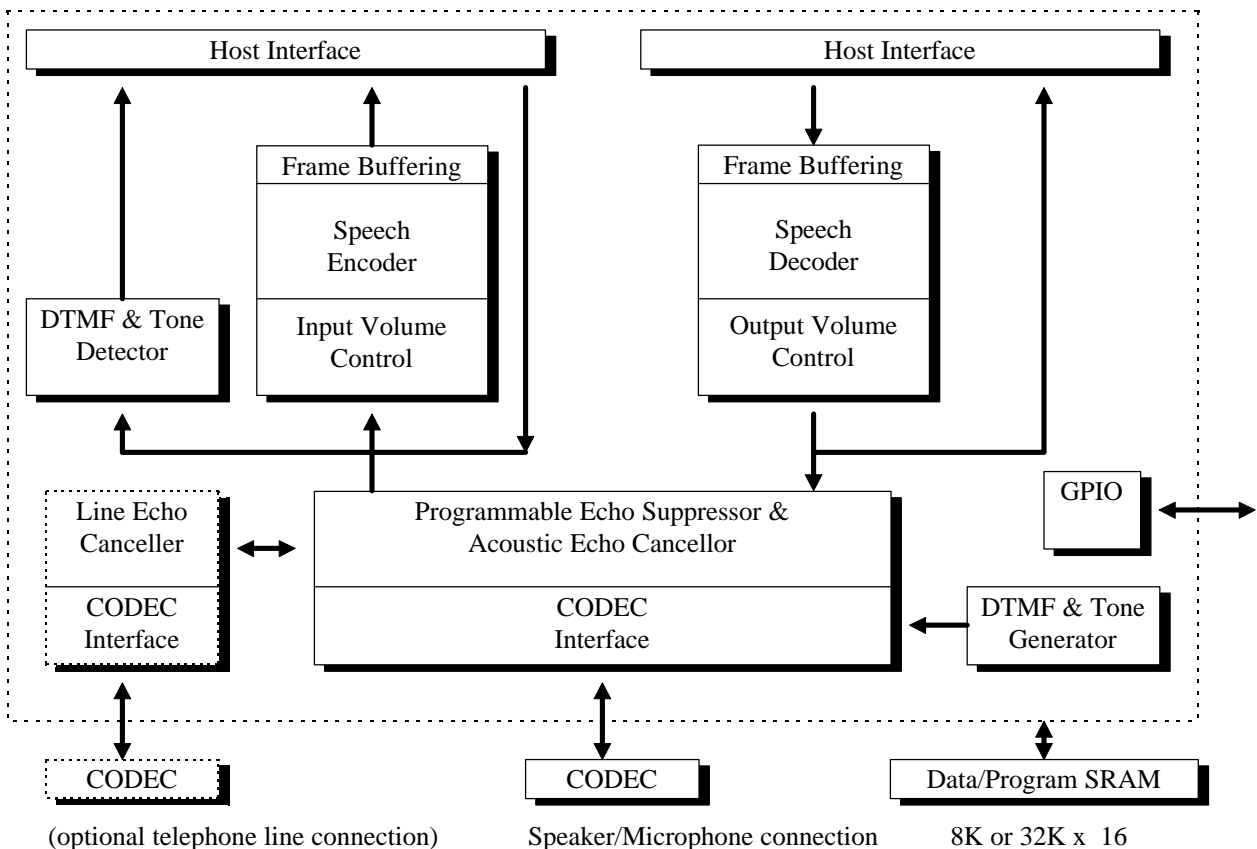


TABLE OF CONTENTS

1. PIN-OUT AND DESCRIPTIONS..... 1-1

2. EXTERNAL COMPONENT CONNECTIONS..... 2-1

2.1 PLL CIRCUIT..... 2-1

2.2 CT8021 EXTERNAL SRAM CONNECTIONS..... 2-2

 2.2.1 CT8021 16-bit external data only SRAM connection..... 2-3

 2.2.2 CT8021 16-bit combined program-data (download) SRAM connection..... 2-4

2.3 CODEC CONNECTION..... 2-5

3. FEATURE OVERVIEW..... 3-1

3.1 INTRODUCTION..... 3-1

3.2 SPEECH MODES..... 3-2

3.3 G.728 LD-CELP..... 3-3

3.4 TRUESPEECH..... 3-3

3.5 TRUESPEECH 6.3 (G.723.1)..... 3-4

3.6 TRUESPEECH 5.3 (G.723.1)..... 3-4

3.7 TRUESPEECH 4.8..... 3-4

3.8 TRUESPEECH 4.1..... 3-4

3.9 G.711 MU-LAW/A-LAW..... 3-4

3.10 TRUESPEECH 8.5 DOWNLOAD-ABLE..... 3-5

3.11 G.729 ANNEX A DOWNLOAD-ABLE..... 3-5

3.12 G.722 DOWNLOAD-ABLE..... 3-5

3.13 AUTOMATIC GAIN CONTROL..... 3-5

3.14 RECORD & PLAYBACK VOLUME..... 3-5

3.15 DSVD..... 3-5

3.16 MICROSOFT WINDOWS SOUND SYSTEM..... 3-6

3.17 SPEECH DATA COMPRESSION AND DECOMPRESSION ACCELERATOR..... 3-6

3.18 DTMF & TONE GENERATION..... 3-6

3.19 TONE DETECTION..... 3-6

3.20 DTMF DETECTION..... 3-7

3.21 FULL DUPLEX SPEAKERPHONE..... 3-7

3.22 ACOUSTIC ECHO CANCELLOR..... 3-7

3.23 8 OR 16-BIT HOST CONTROLLER INTERFACE..... 3-8

3.24 CODEC INTERFACE..... 3-8

3.25 CT8021 CRYSTAL..... 3-8

3.26 POWER SAVE MODES..... 3-9

4. HOST CONTROLLER INTERFACE..... 4-1

4.1 8 OR 16-BIT HOST CONTROLLER INTERFACE..... 4-1

4.2 HOST INTERFACE SIGNALS..... 4-1

4.3 CT8021 HOST CONNECTION WITH EXTERNAL DMA CONTROLLER..... 4-2

4.4 HOST PROGRAMMERS MODEL..... 4-3

4.5 REGISTER DESCRIPTIONS..... 4-4

 4.5.1 Hardware Control Register..... 4-4

 4.5.2 Hardware Status Register..... 4-6

 4.5.3 Software Control Register..... 4-7

 4.5.4 Software Status Register..... 4-7

 4.5.5 Host Receive & Transmit Data Buffer Blocks..... 4-8

4.6 DMA Transfers..... 4-15

 4.6.1 DMA Modes..... 4-15

 4.6.1.1 Full Duplex DMA Mode..... 4-15

 4.6.1.2 Half Duplex DMA Mode..... 4-15

 4.6.2 Burst Mode & Single Cycle Mode..... 4-16

 4.6.3 Flow-Through DMA Transfers..... 4-18

5. CT8021 CODEC INTERFACE..... 5-1

5.1 CODEC OPTIONS..... 5-2

 5.1.1 Master/Slave..... 5-2

 5.1.2 8-bit A-law/Mu-law and 16-bit linear codec..... 5-2

 5.1.3 Short or Long FSYNC..... 5-2

 5.1.4 Programmable SCLK and FSYNC rates..... 5-3

 5.1.5 Stop Codec..... 5-3

6. HOST DATA TRANSFER MODES..... 6-1

6.1 DATA TRANSFER USING THAT DATA BUFFERS.....	6-1
6.1.1 DMA Transfers.....	6-1
6.1.2 Host Processor Transfers.....	6-1
6.2 DATA TRANSFER USING THE SOFTWARE CONTROL AND STATUS REGISTERS.....	6-1
6.2.1 Host Control/Status Register Data Transfer Synchronization Modes.....	6-2
6.2.1.1 Codec Sync Mode.....	6-3
6.2.1.2 Data Sync Mode.....	6-3
6.2.1.3 Poll Sync Mode.....	6-3
6.2.1.4 Host Sync Mode.....	6-4
6.3 DATA TRANSFER OPTIONS SUMMARY.....	6-4
7. PLAY & RECORD DELAY MANAGEMENT.....	7-1
7.1 DELAY AND LATENCY MANAGEMENT.....	7-1
7.2 DATA UNDER-RUN AND OVER-RUN.....	7-1
7.3 BUFFER MONITORING.....	7-1
7.4 FRAME CREATION AND DELETION.....	7-1
7.5 BUFFER FREEZING (PAUSING).....	7-2
7.6 VARIABLE BUFFER DEPTH.....	7-2
7.7 SILENCE GENERATION DURING TRANSMIT (PLAYBACK) DATA UNDER-RUN.....	7-3
7.8 INSERTING SILENCE FRAMES DURING TRUESPEECH PLAYBACK.....	7-3
8. TEST MODES.....	8-1
8.1 TEST MODE 1: COUNT MODE.....	8-1
8.2 TEST MODE 2: DIGITAL MILLIWATT.....	8-1
8.3 TEST MODE 3: INTERNAL LOOPBACK.....	8-1
8.4 CODEC LOOPBACK AND MONITORING.....	8-1
9. CT8021 HOST CONTROL PROTOCOL.....	9-1
9.1 OPERATIONAL MODES.....	9-1
9.2 BASIC PROTOCOL.....	9-2
9.3 RESET & START-UP SEQUENCE.....	9-3
9.4 IDLE.....	9-4
9.5 CODEC CONFIGURATION.....	9-5
9.5.1 Codec Configuration Command.....	9-5
9.5.2 Example codec configurations.....	9-7
9.6 SETTING THE BASE FRAME SIZE.....	9-8
9.7 RECORD.....	9-11
9.7.1 Transfer Mode.....	9-11
9.7.2 Recording Modes.....	9-12
9.7.3 Peak Level Indication.....	9-13
9.7.4 Record Protocol.....	9-14
9.7.5 Select TrueSpeech/G.732.1/G.728 Record Rate.....	9-19
9.7.6 Enable VAD.....	9-20
9.7.7 Dynamic Switching between TrueSpeech/G.723.1 6.3 and 5.3.....	9-20
9.7.8 G.723.1 Frame Type Encoding.....	9-21
9.7.9 Programming the record Peak Threshold.....	9-22
9.7.10 Reading the Record Level Value and Threshold Value.....	9-23
9.8 PLAYBACK.....	9-25
9.8.1 Transfer Mode.....	9-25
9.8.2 Playback Modes.....	9-26
9.8.3 Playback Protocol.....	9-28
9.8.4 Playback Frame Alignment Using Transmit Data Buffer.....	9-34
9.8.5 Select TrueSpeech Playback Rate.....	9-36
9.8.6 Inserting Silence During Compressed Speech Playback.....	9-37
9.8.7 Activating The G.723.1 Frame Erasure Mechanism.....	9-37
9.8.8 G.723.1 Comfort Noise Generation (CNG).....	9-37
9.8.9 Reading the Playback signal Level Value.....	9-39
9.9 STOP RECORD/PLAYBACK.....	9-41
9.10 PLAYBACK & RECORD VOLUME CONTROL.....	9-42
9.10.1 Read Record/Playback Volume Control Command.....	9-42
9.10.2 Set Record/Playback Volume Control.....	9-42
9.10.3 Automatic Gain/Level Control (ACG or ALC).....	9-44
9.10.3.1 PROGRAMMING THE AGC.....	9-46
9.11 PRE-SCALING OF DTMF AND CALL PROGRESS FILTER INPUT.....	9-49
9.12 HOST SYNC MODES.....	9-50
9.13 PLAYBACK & RECORD BUFFER CONTROL.....	9-51

9.13.1	Monitoring Buffer Depth.....	9-51
9.13.2	Speech Frame Create.....	9-51
9.13.3	Speech Frame Delete.....	9-52
9.13.4	Playback & Record Buffer Freeze (Pause).....	9-52
9.13.5	Buffer Depth Limit.....	9-53
9.13.6	Playback Auto-Repeat.....	9-54
9.14	FULL DUPLEX SPEECH MODE.....	9-55
9.15	ACOUSTIC ECHO CANCELLOR IN DSVD MODE.....	9-56
9.15.1	Concurrent AEC and Speech Operation.....	9-57
9.15.2	AEC Tail Length.....	9-58
9.15.3	Controlling the AEC adaptation (training) rate.....	9-59
9.15.3.1	READING THE CT8021 DSP PROCESSOR LOAD.....	9-60
9.15.4	Adding Additional Echo Suppression.....	9-62
9.15.4.1	SET ECHO SUPPRESSER ATTENUATION.....	9-63
9.15.5	Activating the Advanced Echo Canceller features in the CT8021.....	9-64
9.15.5.1	NON-LINEAR PROCESSING (NLP).....	9-65
9.15.5.2	ATTENUATION SCALING (AS) & ATTENUATION BOOST (AB).....	9-66
9.15.5.3	DYNAMIC ACTIVATION OF NLP AND ATTENUATION SCALING (AS).....	9-66
9.15.5.4	EXAMPLE ADVANCED AEC CONFIGURATIONS.....	9-67
9.15.6	AEC TRAIN AND LOCK.....	9-70
9.15.6.1	DETAILED TRAIN & LOCK COMMAND DESCRIPTION.....	9-71
9.15.6.2	AEC LOCKED STATE.....	9-72
9.15.6.3	SETTING THE LOCK THRESHOLD.....	9-73
9.15.6.4	SETTING THE UN-LOCK THRESHOLD.....	9-73
9.15.6.5	FREEZING THE AEC.....	9-74
9.15.7	Controlling the Automatic Loop Adjustment Attenuation.....	9-74
9.15.7.1	PROGRAMMING THE LOOP ADJUSTMENT THRESHOLD.....	9-74
9.15.7.2	PROGRAMMING THE LOOP ADJUSTMENT ATTENUATION.....	9-75
9.15.8	Reading the instantaneous AEC attenuation.....	9-75
9.15.9	Saving and Restoring the AEC and EEC coefficients.....	9-76
9.15.9.1	SAVE AEC COEFFICIENTS.....	9-76
9.15.9.2	RESTORE AEC COEFFICIENTS.....	9-76
9.15.9.3	SAVE EEC COEFFICIENTS.....	9-76
9.15.9.4	RESTORE EEC COEFFICIENTS.....	9-76
9.15.10	Evaluating Echo Canceller Performance.....	9-76
9.16	SPEECH FRAME INTERRUPT.....	9-77
9.16.1	FRAME INTERRUPT VIA THE FR PIN.....	9-77
9.16.2	FRAME INTERRUPT VIA AUX SOFTWARE STATUS REGISTER.....	9-79
9.17	DEVICE SELF-TEST.....	9-80
9.17.1	Check Internal Program ROM Integrity.....	9-80
9.17.2	Test External Data SRAM.....	9-81
9.17.3	External Data SRAM Configuration.....	9-82
9.18	DEVICE IDENTIFICATION.....	9-82
9.18.1	Get Device Identification code.....	9-82
9.18.2	Get Device Version (revision) code.....	9-82
9.19	TONE GENERATION IN IDLE PLAYBACK & RECORD MODES.....	9-83
9.19.1	Stop Tone Generation.....	9-84
9.19.2	Synchronous Mode.....	9-84
9.19.3	Asynchronous mode.....	9-85
9.19.4	Tone Generation in Asynchronous mode.....	9-86
9.19.5	Tone Level Table.....	9-87
9.19.6	New Tone Initialization.....	9-88
9.19.7	Default (Power-On) Tone Table Contents.....	9-89
9.20	CT8021 LINE MONITOR COMMANDS.....	9-90
9.20.1	Line Monitor Command.....	9-90
9.20.2	Synchronous and Asynchronous Monitor Mode.....	9-91
9.20.3	Enabling and Disabling the DTMF Detector.....	9-91
9.20.4	Controlling the Call Progress Tone Filters F0-F3.....	9-92
9.20.5	Reading the Filter Energy Output.....	9-93
9.20.6	Selecting the Filer Characteristics.....	9-94
9.20.7	Changing the Filter Parameter Set Mapping Table.....	9-95
9.20.8	Filter Detection Algorithm.....	9-96
9.20.9	Reading the Filter History Register.....	9-97
9.20.10	Reading the Frame Counter.....	9-97
9.20.11	Programming the CT8021 Filters.....	9-98
9.20.12	Generating Interrupts from Line Monitor Events.....	9-101
9.20.13	Operating the Line Monitor detectors on the playback audio stream.....	9-102
9.21	STANDALONE SPEAKERPHONE.....	9-103
9.21.1	Enter Standalone Speakerphone Mode Command.....	9-103

9.21.2 <i>Get Speakerphone Status Command</i>	9-104
9.21.3 <i>Set Speakerphone Parameters Command</i>	9-105
9.21.4 <i>speakerphone Configuration Command</i>	9-106
9.21.5 <i>Generate Tone (in speakerphone mode)</i>	9-107
9.21.6 <i>Get Electrical Echo Canceller Quality Factor Command</i>	9-107
9.22 HOST-TO-HOST DATA COMPRESSION AND DECOMPRESSION.....	9-109
9.22.1 <i>Host-to-Host compression</i>	9-110
9.22.1.1 RECORDING (COMPRESSION) MODES.....	9-110
9.22.2 <i>Host-to-Host Decompression</i>	9-112
9.22.2.1 PLAYBACK (DECOMPRESSION) MODES.....	9-112
9.22.3 <i>Full Duplex Host-to-Host Compression-Decompression</i>	9-114
9.23 TEST MODES.....	9-115
9.23.1 <i>Test Mode 1: count mode</i>	9-115
9.23.2 <i>Test Mode 2: digital milliwatt</i>	9-115
9.23.3 <i>Test Mode 3: loopback</i>	9-115
9.23.4 <i>Exit Test Mode</i>	9-115
9.23.5 <i>Codec Loopback</i>	9-116
9.23.6 <i>Codec (Audio) Monitoring</i>	9-117
9.23.7 <i>Speech Algorithm Testing</i>	9-118
9.24 POWER SAVE MODES.....	9-119
9.24.1 <i>CT8021 Stop Mode</i>	9-119
9.24.2 <i>Stop Codec Mode</i>	9-119
9.24.3 <i>Re-start Codec</i>	9-119
9.24.4 <i>CT8021 Slow Down (Power Save) Modes</i>	9-120
9.24.5 <i>Disable CLKOUT</i>	9-120
9.25 GENERAL PURPOSE INPUT OUTPUT PINS (GPIO).....	9-121
9.25.1 <i>Configure GIPO system use</i>	9-121
9.25.2 <i>Configure GPIO input/output direction</i>	9-121
9.25.3 <i>Write GPIO pins</i>	9-122
9.25.4 <i>Read GPIO pins</i>	9-122
10. CT8021 HOST INTERFACE TIMING.....	10-1
10.1 HOST WRITE TO SOFTWARE CONTROL REGISTER M.S. BYTE.....	10-1
10.2 HOST READ FROM SOFTWARE STATUS REGISTER M.S. BYTE.....	10-3
10.3 HOST WRITE TO HOST TRANSMIT DATA BUFFER ACCESS PORT.....	10-5
10.4 HOST READ FROM HOST RECEIVE DATA BUFFER ACCESS PORT.....	10-7
10.5 DMA WRITE TO HOST TRANSMIT DATA BUFFER ACCESS PORT (BURST MODE).....	10-9
10.6 DMA WRITE TO HOST TRANSMIT DATA BUFFER ACCESS PORT (SINGLE CYCLE MODE).....	10-10
10.7 DMA READ FROM HOST RECEIVE DATA BUFFER ACCESS PORT (BURST MODE).....	10-11
10.8 DMA READ FROM HOST RECEIVE DATA BUFFER ACCESS PORT (SINGLE CYCLE MODE).....	10-12
11. CT8021 CODEC INTERFACE TIMING AND AC SPECIFICATION.....	11-1
11.1 SHORT FRAME SYNC.....	11-1
11.2 LONG FRAME SYNC.....	11-2
12. ELECTRICAL CHARACTERISTICS.....	12-1
13. MECHANICAL DATA - CT8021.....	13-1

APPENDIX

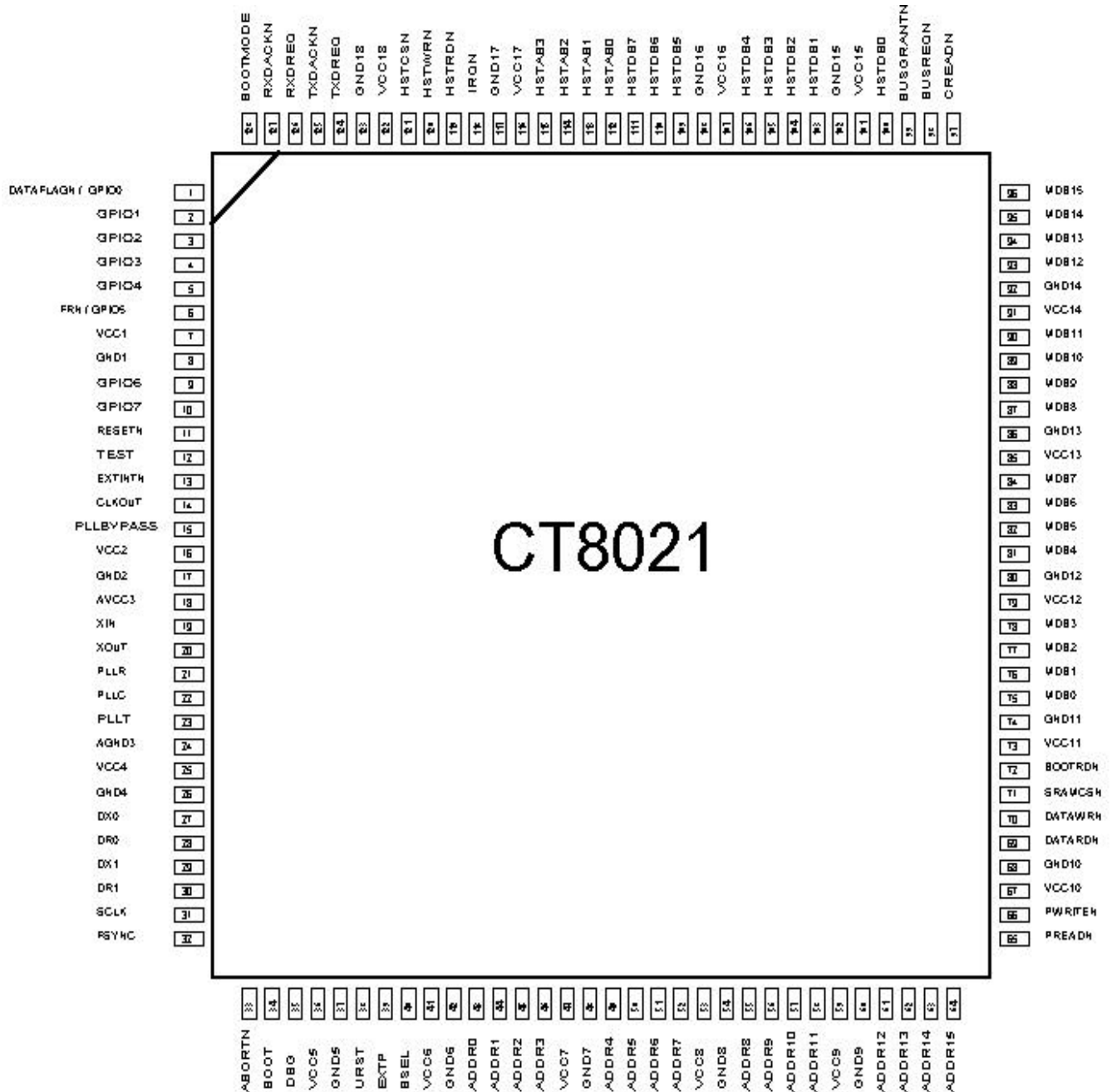
APPENDIX A: SPEAKERPHONE THEORY OF OPERATION.....	A-1
INTRODUCTION.....	A-2
ANALOG CIRCUITRY.....	A-2
ACOUSTIC SYSTEM.....	A-2
CT8021 DIGITAL CANCELLER.....	A-3
<i>Half-Duplex Mode</i>	A-3
<i>Full-Duplex Mode</i>	A-3
VOLUME CONTROL.....	A-4
DIAL TONE DETECTION.....	A-4
PERFORMANCE.....	A-4
EXAMPLE CONTROL ALGORITHM.....	A-5
APPENDIX B: AEC PERFORMANCE IN DSVB APPLICATIONS.....	B-1
APPENDIX C: WAVE FILE FORMAT (.WAV).....	C-1

APPENDIX D: CT8021 EVALUATION BOARD.....	D-1
CT8021 EVB CONNECTOR PIN-OUT.....	D-2
CT8021 EVB SCHEMATIC.....	D-3
APPENDIX E: G.723.1 FRAME STRUCTURE.....	E-1
G.723.1 VERSION HISTORY.....	E-3
G.723.1 CONTROL BITS.....	E-4
ORDERING INFORMATION.....	LAST PAGE

LIST OF FIGURES

CT8021 System Block Diagram.....	1
CT8021 Functional Block Diagram.....	2
CT8021 Pin-Out.....	1-1
PLL Circuit.....	2-1
CT8021 8-bit external data SRAM connection.....	2-3
CT8021 16-bit combined program-data (download) SRAM connection.....	2-4
CT8021 Codec Connection.....	2-5
CT8021 Host connection with external DMA Controller	4-2
Host Receive Data Buffer Block.....	4-9
Host Transmit Data Buffer Block.....	4-12
Single Cycle Mode.....	4-16
Burst Mode.....	4-17
Flow-Through DMA.....	4-18
CT8021 Codec Interface.....	5-1
AGC Attack Hold & Decay.....	9-45
CT8021 Internal Processing Activity.....	9-55
AEC and Speech Compression.....	9-57
Filter Mapping.....	9-94
Host-to-Host Operation.....	9-109

1. PIN-OUT AND PIN DESCRIPTIONS



128 pin PQFP/TQFP

This section contains a list of the CT8021 pins:

Note that the CT8021 is a CMOS device. It is important to make sure that all input pins are connected and have a valid logic level present at all times. Where noted, certain input pins may require external pull-up or pull-down resistors.

Signal naming convention: Active low signals are indicated by a trailing “/” or “N” in the signal name. For example SRAMCS/ or SRAMCSN.

Pin names	Pin Numbers	type	function
HSTDB0 HSTDB1 HSTDB2 HSTDB3 HSTDB4 HSTDB5 HSTDB6 HSTDB7	100 103 104 105 106 109 110 111	I/O/Z	Host Data Bus. HSTDB0 is the least significant data bit.
HSTAB0	112	I	HSTAB0 is the least significant Host Address bit, it selects between the high and low byte of the host interface register selected by HSTAB3-1.
HSTAB1 HSTAB2 HSTAB3	113 114 115	I	Host Address Bus. These address bits are used to select host interface register addressed by host (in conjunction with HSTCS/).
HSTRD/ (HSTRDN)	119	I	Host Read signal. Active Low, output enables HSTDB pins, allowing host to read from the selected host interface register. The interface register is selected via a decode of HSTAB1-3 if HSTCS/ is active, or the Host Receive Data Buffer access port is selected directly if RXDACK/ is asserted.
HSTWR/ (HSTWRN)	120	I	Host Write signal. Active Low, clocks data from HSTDB pins into selected host interface register. The interface register is selected via a decode of HSTAB1-3 if HSTCS/ is active, or the Host Transmit Data Buffer access port is selected directly if TXDACK/ is asserted.
HSTCS/ (HSTCSN)	121	I	Host Interface Chip Select. Active Low. This signal gates the HSTWR/ and HSTRD/ and HSTAB3-0 address decode during a host processor access cycle. The HSTAB3-0 signals should be stable and valid when HSTCS/ is asserted. This signal must not be asserted during a DMA cycle on the host port. HSTCS/ must be high when either TXDACK/ or RXDACK/ are asserted (low).
TXDREQ	124	O	Active High signal. DMA transmit request. This signal is asserted to indicate that the device is ready to accept transmit data. Data can be transferred either by DMA or a host processor access cycle. This signal can be enabled or disabled by the host via the Hardware Control Register. By default, this signal is disabled.

TXDACK/ (TXDACKN)	125	I	Active Low signal. DMA Transmit Acknowledge. This signal is asserted by an external DMA controller on the host port, together with HSTWR/, to clock a byte from the HSTDB databus pins into the host interface Transmit Data Buffer Access port. This signal provides direct access to the Host Transmit Data Buffer Access Port, without involving HSTAB0-3 or HSTCS/. If this signal is not used, it should be connected to VCC via a 10K ohm pull-up resistor. This signal must not be asserted when either RXDACK/ or HSTCS/ are active.
RXDREQ	126	O	Active High signal. DMA receive request. This signal is asserted to indicate that the device is ready to provide receive data. Data can be transferred either by DMA or a host processor access cycle. This signal can be enabled or disabled by the host via the Hardware Control Register. By default, this signal is disabled.
RXDACK/ (RXDACKN)	127	I	Active Low signal. DMA Receive Acknowledge. This signal is asserted by an external DMA controller on the host port, together with HSTRD/, to accept a receive data byte via the HSTDB databus pins from the host interface Receive Data Buffer Access port. This signal provides direct access to the Host Receive Data Buffer Access Port, without involving HSTAB0-3 or HSTCS/. If this signal is not used, it should be connected to VCC via a 10K ohm pull-up resistor. This signal must not be asserted when either TXDACK/ or HSTCS/ are active.
IRQ/ (IRQN)	118	OC	Interrupt Request. Open collector output (requires external pull-up resistor with min value 1K ohm). This signal is asserted to indicate an interrupt request to the host controller.
MDB0 MDB1 MDB2 MDB3 MDB4 MDB5 MDB6 MDB7 MDB8 MDB9 MDB10 MDB11 MDB12 MDB13 MDB14 MDB15	75 76 77 78 81 82 83 84 87 88 89 90 93 94 95 96	I/O/Z	External Memory Data Bus, used to interface to external memory. The CT8021 requires an external memory configured as 8K or 32K x 16. The 32K x16 configuration is required for support of downloadable external software modules.
ADDR0 ADDR1 ADDR2 ADDR3	43 44 45 46	O/Z	External Memory Address Bus, used to interface to external memory. Note that the CT8021 is a 16-bit device and the address lines indicate access to 16-bit data words.

ADDR4	49		
ADDR5	50		
ADDR6	51		
ADDR7	52		
ADDR8	55		
ADDR9	56		
ADDR10	57		
ADDR11	58		
ADDR12	61		
ADDR13	62		
ADDR14	63		
ADDR15	64		
BSEL	40	O/Z	Byte Select. Not used in CT8021 designs. Used in CT8020 designs to access external byte wide memory.
DATARD/ (DATARDN)	69	O/Z	External Data Memory Read. Active Low, used to indicate a data read cycle from external data memory
DATAWR/ (DATAWRN)	70	O/Z	External Data Memory Write. Active Low, used to indicate a data write cycle to external data memory.
PREAD/	65	O/Z	External program memory read.
PWRITE/	66	O/Z	External program memory write.
CREAD/ (CREADN)	97	O/Z	Combined external program read and data read. This is equivalent to logical AND combination of PREAD/ and DATARD/. This pin is asserted (active low) whenever an external data or program read cycle is executed. Compatability Note: this pin is not provided by the CT8020. Use this pin in place of DATARD/ to create a single external combined program/data address space to support external downloadable software modules with the CT8021.
BOOTRD/	72	O/Z	Reserved - NC (do not connect)
SRAMCS/ (SRAMCSN)	71	O/Z	external data SRAM chip select. Active low. This signal is asserted during external data access only. This signal is NOT asserted during external program memory read or write accesses.
RESET/ (RESETN)	11	I	Reset signal. Active Low, the pin is driven low to reset the device. Note - this pin is not a TTL input. VIH (max) = 4.5 volts . VIL (min) = 1.3 volts. The reset pulse should be a minimum of 10 CLKOUT periods in width (after Vcc has stabilized and a clock is present at XOUT)

GPIO0, DATAFLAG\	1	I/O	General purpose Input/Output pins. For CT8015 compatibility: GPIO 0 may be assigned as DATAFLAG/ (output) GPIO 5 may be assigned as Frame Interrupt, FR/ (output)
GPIO1	2		
GPIO2	3		
GPIO3	4		
GPIO4	5		After and during reset, all the GPIO pins are all configured as inputs. (compatibility note: the CT8020 configures GPIO0 and 5 as outputs by default)
GPIO5, FR\	6		
GPIO6	9		
GPIO7	10		Connect 47K ohm pull-down resistors to GND on each GPIO pin to ensure that a valid input signal level is present at all times.
SCLK	31	I/O	Shift Clock for codec interface. During and after reset, this pin is configured as an input. Connect 47K ohm pull-down resistors to GND to this pin to ensure that a valid input signal level is present at all times. If this pin is permanently driven by an external signal, no pull-down is required. The SCLK signal is typically expected to operate at 2.048 MHz. When used as an output (master mode) the SCLK rate is programmable by the host.
FSYNC	32	I/O	Frame Sync clock for codec interface. During and after reset, this pin is configured as an input. Connect 47K ohm pull-down resistors to GND to this pin to ensure that a valid input signal level is present at all times. If this pin is permanently driven by an external signal, no pull-down is required. The FSYNC signal is typically expected to operate at 8 KHz to provide the 8KHz sample clock required by the external serial codec. When used as an output (master mode), the FSYNC rate is programmable by the host.
DX0	27	O/Z	Serial Transmit Data Output for codec 0. This output is always high impedance when not transmitting data. This codec pin is used for telephone line output in Standalone Speakerphone mode.
DR0	28	I	Serial Receive Data Input for codec 0. Connect 47K ohm pull-down resistors to GND to this pin to ensure that a valid input signal level is present at all times. Note that codec chip output pins typically tri-state when not actually transmitting data. This codec pin is used for telephone line input in Standalone Speakerphone mode.
DX1	29	O/Z	Serial Transmit Data Output for codec 1. This output is always high impedance when not transmitting data. This is the default codec output used for playback.
DR1	30	I	Serial Receive Data Input for codec 1 Connect 47K ohm pull-down resistors to GND to this pin to ensure that a valid input signal level is present at all times. Note that codec chip output pins typically tri-state when not actually transmitting data. This is the default codec input used for record.

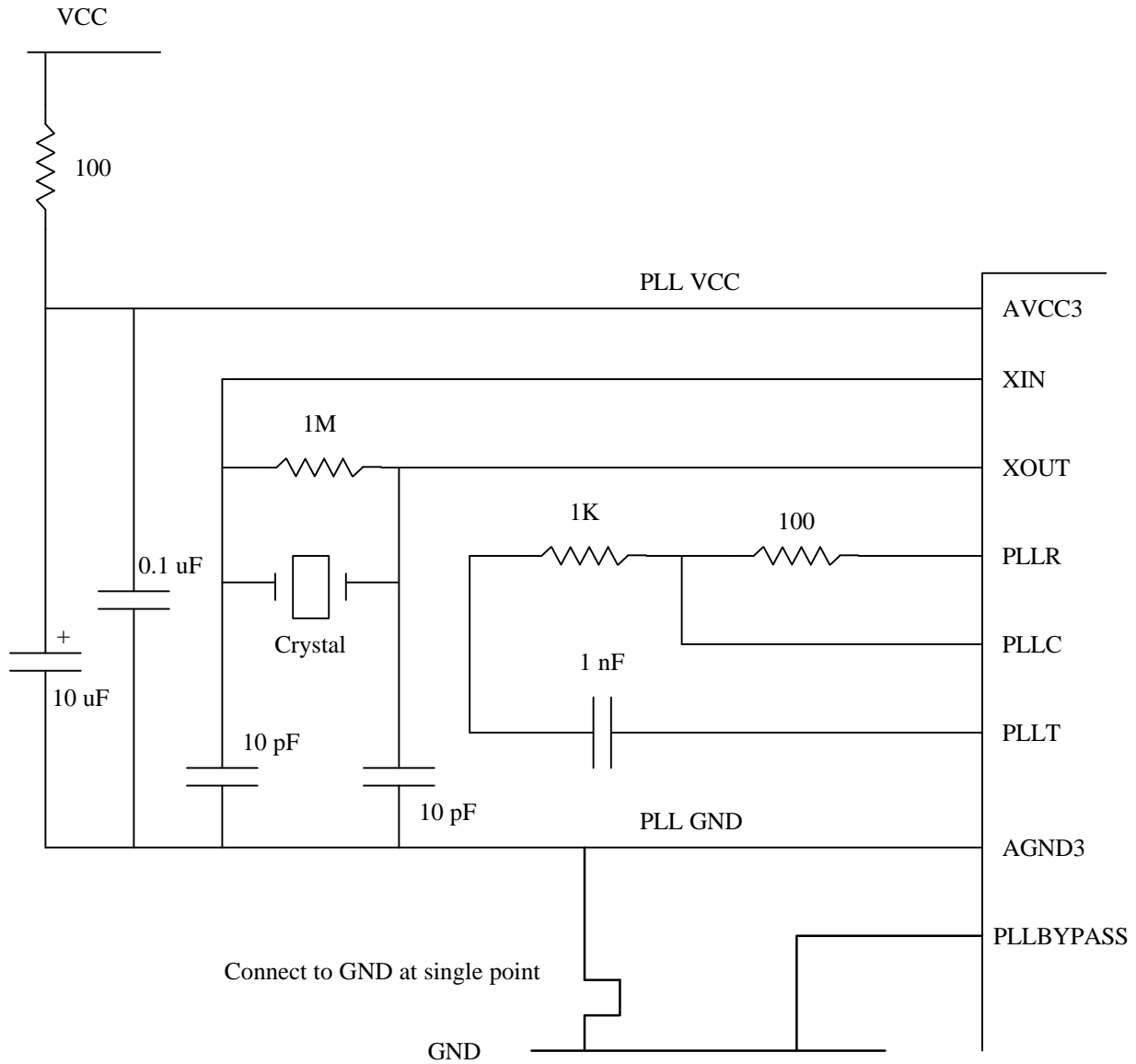
XIN	19	I	Crystal Input or external oscillator input.
XOUT	20	O	Crystal Output
CLKOUT	14	O	Clock Out = MAINCLOCK/(CLK_RATE+1). This is the internal CT8021 DSP core clock. The output may be disabled to reduce power consumption. The MAINCLOCK frequency in PLL mode, is the external crystal frequency * 11. In PLL by-pass mode, MAINCLOCK is the external oscillator frequency divided by 2. CLK_RATE is the DSP core clock division factor that may be programmed by the host controller using the slow down mode command. Note that significant short term clock jitter may be present on the CLKOUT signal when the PLL is enabled.
TEST	12	I	Reserved - connect to GND.
PLLR	21	Analog	PLL support circuitry pins. Connect to external PLL filter circuit.
PLLC	22	Analog	
PLLT	23	Analog	
AVCC3	18	I	PLL VCC connection
AGND3	24	I	PLL GND connection
PLLBYPASS	15	I	Disables internal PLL when high and allows direct use of an external (90.112 MHz) clock applied to the XIN pin. Connect to GND when operating using PLL and external 4.096 MHz crystal.
BUSREQ/	98	I	Reserved input, active low. Connect this pin to VCC via a 10K ohm pull-up resistor.
BUSGRANT/	99	O	Reserved - NC (do not connect)
EXTP	39	I	Reserved - connect to GND
BOOTMODE	128	I	Reserved - connect to GND
DBG	35	I	Reserved - connect to GND
ABORT/	33	I	Reserved, active low - connect to VCC via 10 Kohm pull-up resistor.
BOOT	34	I	Reserved - connect to GND
URST	38	I	Reserved - connect to GND
EXTINT/	13	I	Reserved, active low - connect to VCC via 10 Kohm pull-up resistor.
GND1	8	I	Ground pins
GND2	17		
GND4	26		
GND5	37		
GND6	42		
GND7	48		
GND8	54		
GND9	60		
GND10	68		
GND11	74		
GND12	80		
GND13	86		

GND14	92		
GND15	102		
GND16	108		
GND17	117		
GND18	123		
VCC1	7	I	Power Supply pins
VCC2	16		
VCC4	25		
VCC5	36		
VCC6	41		
VCC7	47		
VCC8	53		
VCC9	59		
VCC10	67		
VCC11	73		
VCC12	79		
VCC13	85		
VCC14	91		
VCC15	101		
VCC16	107		
VCC17	116		
VCC18	122		

Note, pins marked “Reserved - NC (no connect)” should be left un-connected. These Reserved pins are outputs, connecting them to either GND or VCC may cause damage to the device. Other pins are marked either “Reserved - connect to GND” or “Reserved - connect to VCC via 10 K ohm pull-up resistor”. These Reserved pins are inputs which require a defined input signal level. They should not be left to “float”. Pins marked as Reserved are not intended for active use by the user. However these pins may be used by other devices in the CT8000 series family.

2. External Component Connections

2.1 PLL Circuit



Crystal frequency is nominal 4.096 MHz. Crystal may be replaced by external 4.096 MHz clock source.

Figure 2-1 PLL Circuit

2.2 CT8021 external SRAM connection

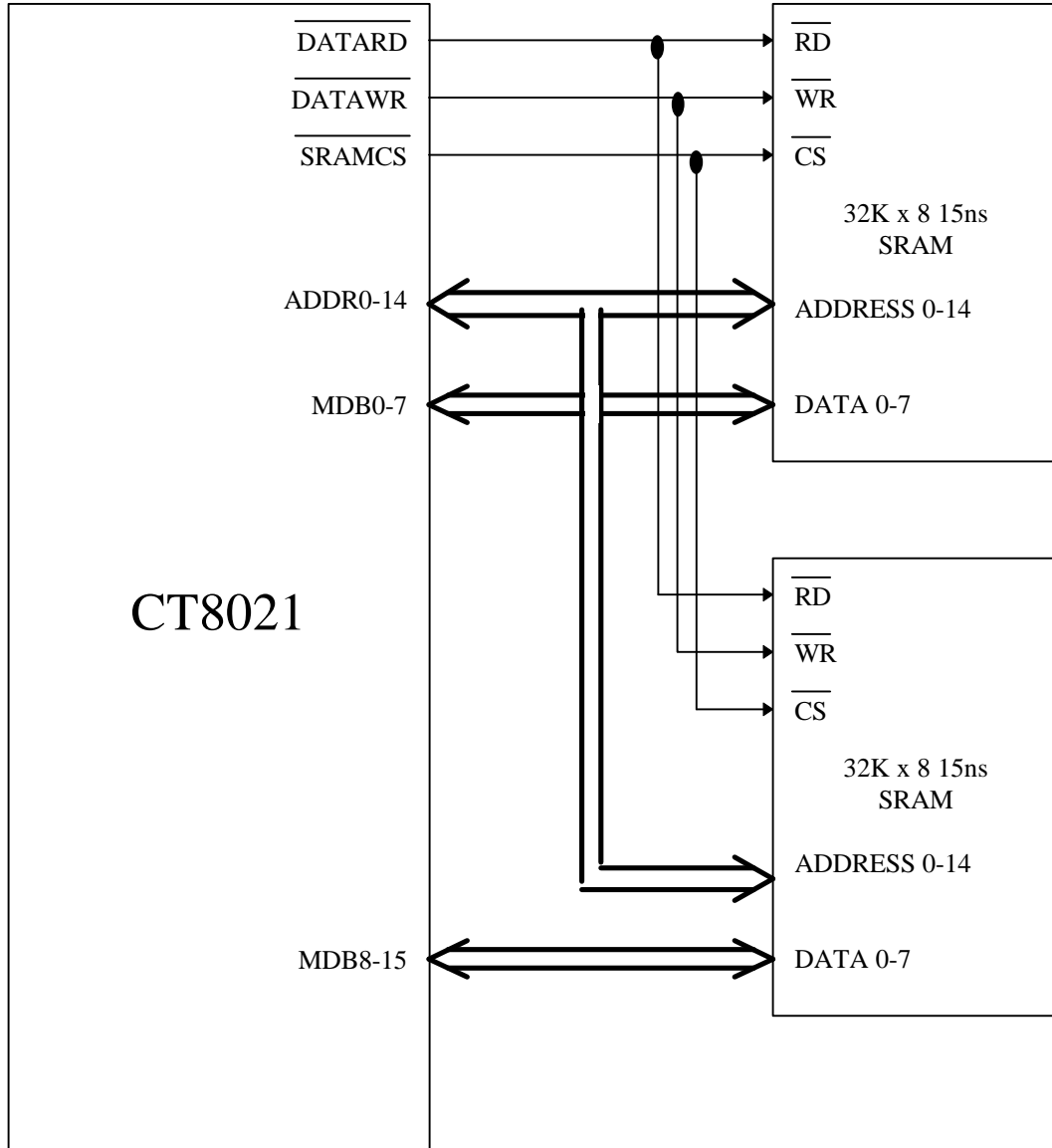
All CT8021 configurations require the use of a minimum 8K x 16 external sram

The CT8021 requires an external memory of 8K x 16 for a “data only” configuration or 32K x 16 for a “combined program and data” configuration that will support download of external software modules. The first 8K x 16 portion of this memory is used to provide storage for data buffers and other data objects required for normal operation of the device. In the basic “data only” configuration, the memory is used only for data storage, with execution of code occurring from the CT8021’s built-in internal ROM memory. This built-in ROM contains all the code required for basic operation of the CT8021, including the G.723.1 and G.728 speech compression code.

The CT8021 includes the capability to expand the functionality of the device by the addition of external software modules (for example a software coder module for TrueSpeech® 8.5). These software modules require external memory. In the 32K x 16 configuration, only the first 8K words are used as data space with the remainder of the space available for downloaded code.

Please refer to the diagrams on the following pages.

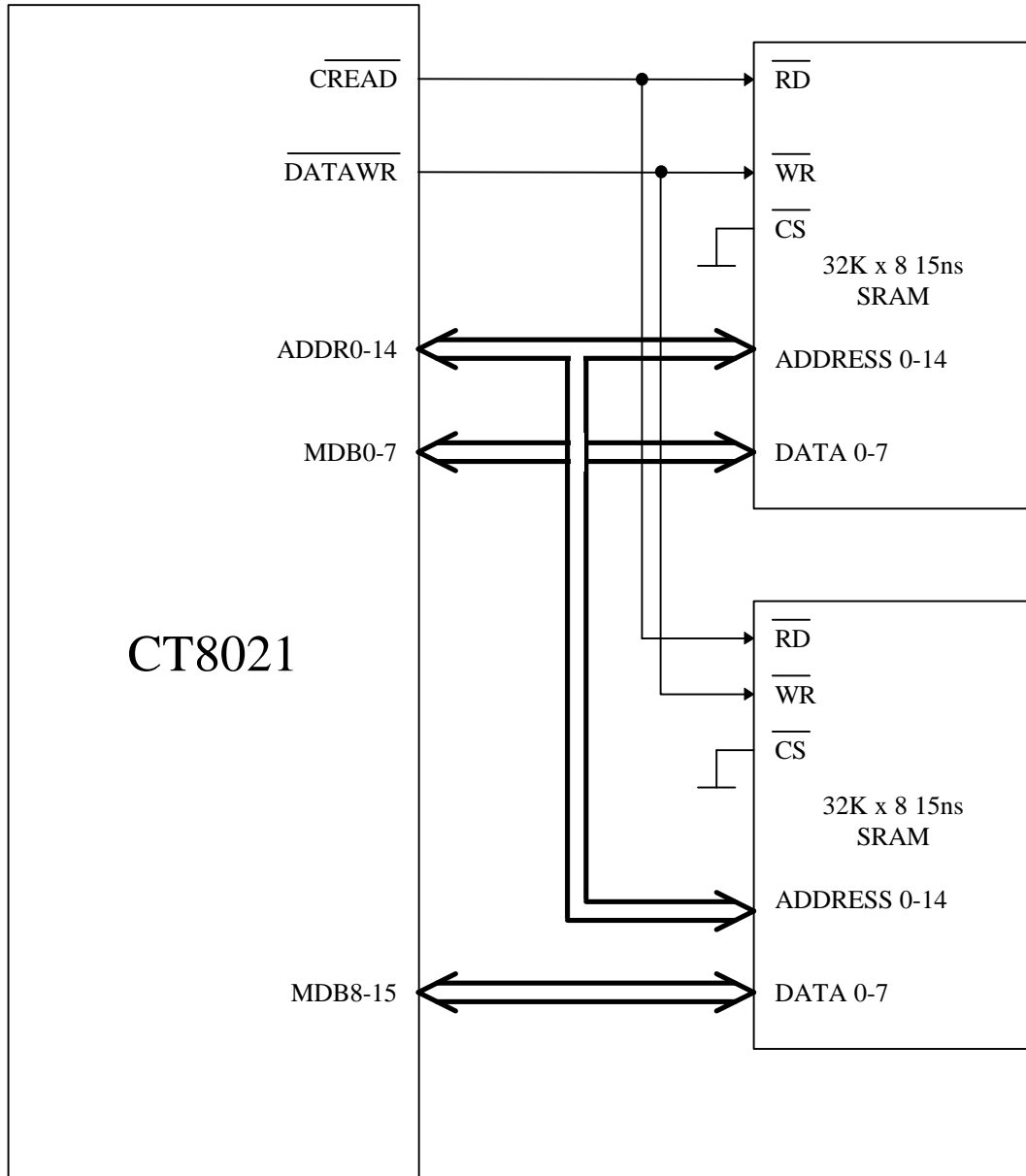
2.2.1 CT8021 16-bit external data only SRAM connection



This is the basic external memory configuration showing the CT8021 connected to 2 external data sram chips. This configuration does not support download of external software coder modules. Note that the BSEL signal is not used in 16 bit wide memory configurations. The connection of the chip select signal SRAMCS/ is optional. The SRAMCS/ signal may be left un-connected with the CS input of the srams connected to GND.

Figure 2-2 External data only SRAM connection

2.2.2 CT8021 16-bit combined program-data (download) SRAM connection



This is the extended external memory configuration showing the CT8021 connected to 2 external combined program-data sram chips. This configuration supports download of external software coder modules into the combined program-data memory. Use this configuration only if you intend to use the CT8021 with external software coder modules such as TrueSpeech 8.5. Note that the BSEL signal is not used in 16 bit wide memory configurations.

Figure 2-3 External combined program-data SRAM connection

2.3 Codec Connection

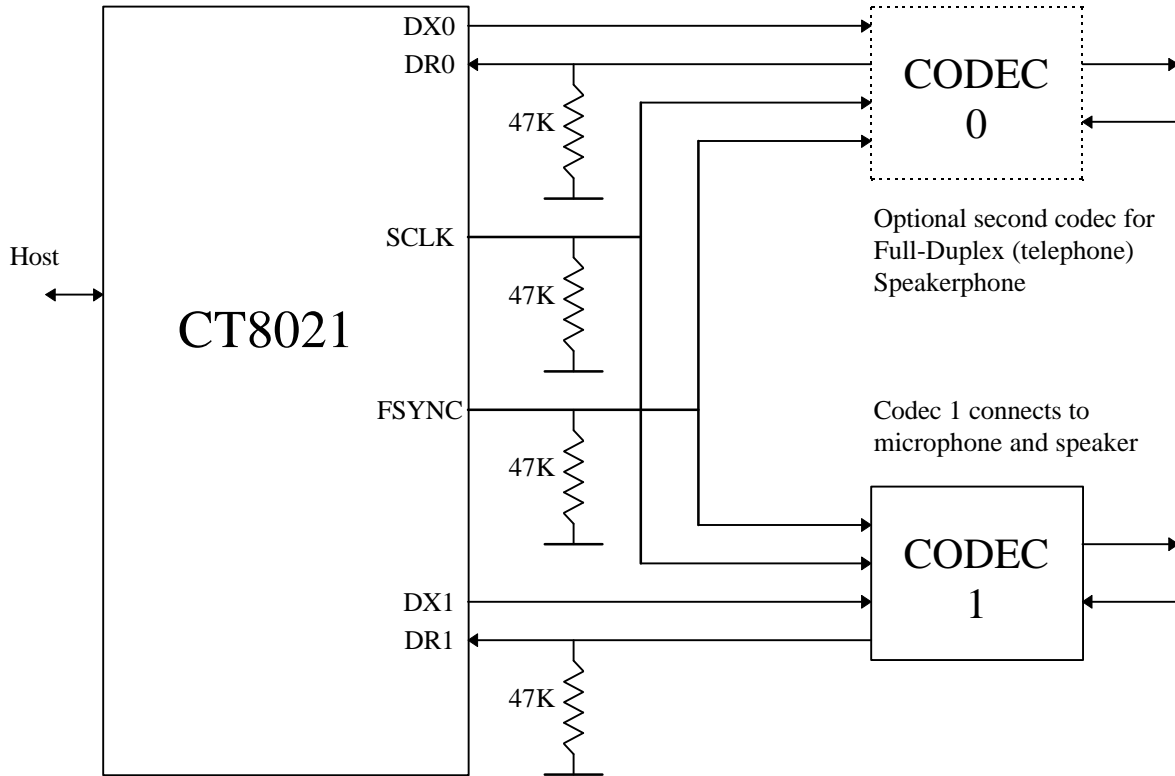


Figure 2-4 CODEC connection

In Master mode, the CT8021 generates the FSYNC and SCLK signals.

In Slave mode, the FSYNC and SCLK signals are generated externally. In slave mode, the CT8021 FSYNC and SCLK pins are inputs.

Note that during and after reset, the SCLK and FSYNC pins are configured as inputs. As such, they require external pull-down resistors to ensure that a safe and defined logic level is present.

CODEC 0 is required only if “Standalone” Full Duplex Speakerphone operation is desired (e.g for use as an analog phone line speakerphone). In this configuration, CODEC 0 connects to the audio input/output of an analog telephone line. CODEC 0 is not required for DSVD/Video Conferencing Speakerphone operation.

For codec-less operation, both CODEC 0 and CODEC 1 may be omitted. In this case, the FSYNC, SCLK, DR0 and DR1 inputs should have pull-down resistors connected to ground to ensure a valid input signal level.

3. FEATURE OVERVIEW

3.1 Introduction

The CT8021 is a full duplex TrueSpeech G.723.1 and G.728 real-time speech compression Encoder/Decoder with built-in concurrent echo cancellation. It is designed to operate in conjunction with a host processor and one or two external A-to-D/D-to-A serial CODECs. The host processor is responsible for managing the bi-directional stream of compressed speech data provided by the CT8021. The CT8021 accepts un-compressed speech in 8-bit A-law/mu-law or 16-bit Linear formats at 64/128 Kbits/sec from the receive channel of the external serial CODEC and compresses the speech to according to the format selected by the host. The compressed speech is passed to the host processor in frames every 30 ms (optionally 30,20,10,5 or 2.5 ms in the case of G.728 LD-CELP). At the same time that the CT8021 is compressing speech, it can concurrently accept compressed speech from the host for decompression, outputting the uncompressed speech to the transmit channel of the external codec. The CT8021 supports independent and asynchronous operation of the host receive and transmit (compressed) data streams. The transmit and receive interfaces to the external A-to-D/D-to-A codec operate synchronously. The device contains variable depth buffers for each data direction and also provides support of frame deletion and insertion to accommodate sample clock differences between the source and destination devices. The CT8021 can also operate in a codec-less mode as a speech compression/decompression accelerator device.

The CT8021 also provides auxillary telephony functions such as DTMF generation and detection, and Call Progress Tone generation and detection (e.g dial tone).

3.2 Speech Modes

The CT8021 playback (transmit) and record (receive) channels can operate in several different speech modes : TrueSpeech G.723.1 6.3/5.3 and G.728 LD-CELP, TrueSpeech 8.5 (see note 1), TrueSpeech 4.8 and 4.1, G.729 Annex A+B (see note 1), G.722 (see note 1), G.711 A-law/Mu-law, 8-bit and 16-bit linear (uncompressed). The mode of operation of the playback and record channel can be selected independently of each other (except in the case of G.728).

Speech Coder	Used In
G.723.1 6.3 & 5.3 kbps	H.324 for video conferencing over dial-up V.34 modems H.323 multi-media conferencing over LANs and TCP/IP type "packet-data" networks (e.g. Internet)
G.728 16 kbps	H.320 for video conferencing over ISDN
TrueSpeech 8.5 (1) 8.5 kbps	DSP Group speech coder built-in to Microsoft Windows 95. Also used in some DSVD modems and Internet Telephones
TrueSpeech 4.8 & 4.1 4.8/4.1 kbps	DSP Group proprietary extension of the 6.3/5.3 coder for use in applications requiring lower bit rates (e.g. proprietary non-standard extensions to H.324)
G.729A+B (1) 8 kbps	optional speech coder used in H.323
G.711 Mu-Law/A-Law 64 kbps	Standard speech compression used in digital telephony systems (e.g T1/E1)
G.722 64 Kbps (1)	7 KHz Wide bandwidth ADPCM audio coder used in H.320 (ISDN)
16-bit or 8-bit linear uncompressed data 128 or 64 kbps	Uncompressed audio (e.g. used in Microsoft WAVE files)

Notes:

1. The CT8021 supports the TrueSpeech 8.5, G.729A+B and G.722 speech coders as external downloadable expansion software modules. These are not built-in to the CT8021 internal program rom. Use of downloadable expansion software modules requires that the CT8021 be used in conjunction with the appropriate application circuit described in section 2 of this datasheet. All other speech compression functions (G.723.1, G.728, TrueSpeech 4.8 and 4.1, G.711 and the 8-bit & 16-bit uncompressed modes) are built-in to the CT8021.

3.3 G.728 LD-CELP

The ITU G.728 16 Kbps LD-CELP (Low Delay Code-Excited Linear Prediction) algorithm is designed to operate on 8KHz 16-bit linear input samples (128 Kbps) and produces output at the rate of 16 kbps. When used with an external G.711 A-law/Mu-law codec, the CT8021 provides intermediate conversion from G.711 A-law/Mu-law to 16-bit linear format prior to the operation of the G.728 function. In the CT8021 implementation, data to be compressed can be operated upon in blocks of 240,160, 80,40 or 20 samples corresponding to 30,20,10,5 or 2.5 ms of speech data. Due to the high processing power (mips) requirements of the G.728 algorithm, there are restrictions on the additional CT8021 features that can be operated concurrently with full-duplex G.728 compression/decompression. The available features are also affected by the block size used.

Uncompressed Block Size (samples)	Period	Compressed Block Size (16 bit words)	Full Duplex G.728	Concurrent AEC only	Concurrent AEC plus DTMF and Call Progress Tone Detection
240	30ms	48	Yes	Yes	Yes
160	20ms	32	Yes	Yes	No
80	10ms	16	Yes	Yes	No
40	5 ms	8	Yes	No	No
20	2.5 ms	4	Yes	No	No

The compressed G.728 speech data is packed with 10 significant bits per 16-bit data word. The upper 6 bits of each compressed data word are not used (set to zero).

3.4 TrueSpeech

TrueSpeech is a speech compression technology which reduces or compresses the amount of data used to encode a speech waveform. TrueSpeech compression reduces speech sampled (digitized) at 8,000 samples per second, in 16-bit samples (128,000 bits/sec) to 6.3, 5.3, 4.8 or 4.1 Kbits/sec with minimal degradation in speech quality. When used with an external G.711 A-law/Mu-law codec, the CT8021 provides intermediate conversion from G.711 A-law/Mu-law to 16-bit linear format prior to the operation of the TrueSpeech function

In the CT8021, uncompressed speech is obtained from an external codec operating at 8,000 samples per second. If an external G.711 A-law or Mu-law codec is used, the speech is sampled by the external codec with 14-bits of effective resolution. The G.711 codec converts the 14-bits (linear) samples to 8-bit A-law/Mu-law for transmission to the CT8021 (at 64,000 bits/sec). The CT8021 receives the speech samples from the codec and converts them back to 14-bit linear form. The 14-bit data is converted to 16 bits by appending two zero bits in the least significant bit positions. The 16-bit data is collected into blocks of 240 samples (480 byte) every 30 ms and TrueSpeech compressed for transfer to the host. If an external linear 16-bit codec is used, the speech data is sampled with 16-bit resolution and passed directly from the CODEC to the CT8021 at 128,000 bits/sec.

The CT8021 supports simultaneous compression and decompression of TrueSpeech data in real-time.

3.5 TrueSpeech 6.3 (G.723.1)

The CT8021 implements the final (version 5.1) ITU-T formal release of the G.723.1 speech coder standard, including G.723.1 Annex A VAD/CNG.

In TrueSpeech 6.3 (G.723.1), data is compressed into 12-word blocks. This provides 192 bits per block. However, only 189 bits are used for actual speech data, giving a raw data rate of 6,300 bits/sec. When used in compliance with G.723.1 two of the spare bits are assigned for control information (the 2 l.s. bits of the first word of each compressed speech block). The CT8021 G.723.1 implementation includes the optional G.723.1 Annex A VAD/CNG “silence” compression feature (Voice Activity Detection / Comfort Noise Generation). When this feature is enabled, the CT8021 can generate and decode the 4-byte and 1-byte Annex A silence frames. The CT8021 supports dynamic switching between the G.723.1 6.3 and 5.3 rates for both the encoder and decoder. In addition, the G.723.1 6.3 and 5.3 decoders support frame erasure (for corrupt or missing frames) triggered using specially marked data frames.

3.6 TrueSpeech 5.3 (G.723.1)

In TrueSpeech 5.3, data is compressed into 10-word blocks. This provides 160 bits per block. However, only 158 bits per block are used for actual speech data, giving a raw data rate of 5,266.67 bits/sec. When used in compliance with G.723.1, the two “spare” bits are used for control information. The 5.3 rate can be operated in conjunction with the VAD/CNG silence compression feature.

3.7 TrueSpeech 4.8

In TrueSpeech 4.8, data is compressed into 9-word blocks. This provides 144 bits per block. However only 142 bits per block are used for actual speech data, giving a raw data rate of 4733.33 bits/sec. The 4.8 rate can be operated in conjunction with the VAD/CNG silence compression feature.

3.8 TrueSpeech 4.1

In TrueSpeech 4.1, data is compressed into 8-word blocks. This provides 128 bits per block. However only 124 bits per block are used for actual speech data, giving a raw data rate of 4133.33 bits/sec. The 4.1 rate can be operated in conjunction with the VAD/CNG silence compression feature.

3.9 G.711 Mu-law/A-law

The CT8021 is designed to operate with one or two low cost external G.711 Mu-law/A-law serial codecs to provide A-to-D and D-to-A conversion (use of 2 codecs is required only for standalone analog speakerphone operation). The CT8021 can provide bit-transparent “pass through” access to the serial codec data stream. In addition the CT8021 can also provide volume control (scaling) operating on the G.711 data. G.711 mode is also supported when the CT8021 is used with an external 16-bit linear codec, where the CT8021 provides 16-bit linear to G.711 Mu-law/A-law conversion.

3.10 TrueSpeech 8.5 Download-able

DSP Group offers an additional TrueSpeech data rate at 8.5 Kbits/sec. This mode is not included as a “built-in” feature of the CT8021. However, the CT8021 application circuit can be configured to support the download of external software coder “expansion” modules from the host controller. In TrueSpeech 8.5 mode, speech data is compressed/decompressed into 16-word (32 byte) blocks, which are transferred to or from the host every 30 ms. This corresponds to a data rate of 8,533.33 bits/sec. The CT8021 TrueSpeech 8.5 implementation is compatible with the TrueSpeech 8.5 format used in the CT8005 TrueSpeech Messaging Co-Processor, CT8015 TrueSpeech DSVD Co-Processor, CT8020 TrueSpeech G.723.1 Co-Processor and Microsoft Windows Sound System 2.0 and Windows 95 software TrueSpeech CODECs.

3.11 G.729 Annex A Download-able

DSP Group offers an external downloadable expansion software module which can enable the CT8021 to support the G.729 Annex A speech coder which operates at 8 Kbits/sec. The DSP Group implementation of G.729A also includes the G.729 Annex B VAD/CNG “silence” compression feature. G.729A is a reduced complexity version of the earlier G.729 speech coder. The two coders (G.729 and G.729A) are fully inter-operable.

3.12 G.722 Download-able

DSP Group offers an external downloadable expansion software module which can enable the CT8021 to support the G.722 7KHz wideband ADPCM operating at 64 kbits/sec.

3.13 Automatic Gain Control

In speech record mode (compressed or uncompressed), the CT8021 record process can optionally include an Automatic Gain Control (AGC) function. When this feature is enabled, the CT8021 can automatically adjust the record signal level by internal digital scaling of the input speech signal. The control parameters for the AGC, which include max and min gain (amplification), attack and decay times (response time) are programmable by the host. Note that for optimal AEC performance in DSVD mode (videoconferencing), use of the AGC feature concurrently with the Acoustic Echo Canceller is not recommended. However, where the user considers simultaneous use of both AEC and AGC to be necessary, the CT8021 does include a mechanism to reduce the echo amplification effect caused when the AGC gain increases.

3.14 Record and Playback Volume

The CT8021 supports host programmable record and playback volume controls. This enables internal digital scaling of the input and output speech signal levels. The host can elect to use fixed level settings during record in situations where AGC operation is not desired (for example - high noise environments). Also AGC operation is not appropriate in full duplex speech mode when the Acoustical Echo Canceller (AEC) is active, since the AGC would interfere with the ability of the AEC to operate correctly. In particular, the AGC would operate to amplify the canceled (attenuated) echo signal. Volume control is available in all record and playback modes including G.711Mu-law/A-law. In the case of G.711, use of the default 1.0 volume scaling provides bit-transparent pass through of data from the host to the external G.711 serial codec chip.

3.15 DSVD

Digital Simultaneous Voice and Data is a protocol used to enable collaborative computing where voice conversation and full duplex data transfers occur simultaneously over a single telephone line via a high speed modem. Since the data transfer rate achievable over a telephone line is limited, speech compression is required to make it possible to send speech simultaneously with data. Applications for DSVD include application sharing, shared whiteboard and interactive games.

3.16 Microsoft Windows[®] Sound System

In addition to supporting playback and recording of compressed TrueSpeech data, the CT8021 can also playback and record un-compressed data in 16-bit linear, 8-bit linear, 8-bit mu-law and 8-bit A-law formats. The 8-bit linear format supports data in both signed and unsigned form (with or without most significant data bit inversion). This provides true Windows Sound System (WSS) 8 bit .wav file compatibility. The CT8021 supports 8- and 16-bit 8 KHz, and 11 KHz Microsoft WAVE file mono formats making it an ideal low cost device for speech enabling personal computers and personal digital assistants (PDAs). The Microsoft WAVE file format also includes support for storing of TrueSpeech 8.5 speech data, making it possible to directly exchange TrueSpeech 8.5 WAVE recordings with WSS compliant PC's. A "software only" TrueSpeech 8.5 encoder/decoder is a standard part of the Sound Recorder application included with Windows[®] 95. Note that sample rates other than 8KHz require that the CT8020 is operated with an external A-to-D codec capable of supporting the desired sample rate.

3.17 Data Compression/De-Compression Accelerator

The CT8021 can be used a data compression accelerator. In addition to supporting real-time recording and playback via external CODECs, the CT8021 can operate in a codec-less environment where it compresses 16-bit linear speech data from the host, returning the compressed data to the host. Simultaneous compression and decompression is supported. This enables the CT8021 to be used in a sound-card type environment where the speech input and output channels are independent of the CT8021.

3.18 DTMF / Tone Generator

The DTMF/Tone generator can generate any single or dual frequency tone in the range 300Hz to 3 KHz. The CT8021 has preset parameters for generating all 16 DTMF digits (0-9, *, #, and the "hidden" digits A, B, C, D) as well as standard USA dial tone, busy tone, and ringing (ringback) tone. The preset parameters can be re-programmed by the host controller to generate any single or dual frequency tone combination. The relative level (twist) of dual frequency tones can also be selected. The tone generator can be operated concurrently with the record/playback speech operation.

3.19 Tone Detection

The CT8021 contains a set of four independent programmable call progress tone filters which can be used to detect call progress tones such as dial tone, busy tone, ringing (or ringback) tone and fax calling tone (CNG). The tone detector can be configured to operate on either the incoming (record) data stream or outgoing (playback) data stream.

3.20 DTMF Detection

The CT8021 includes a DTMF detector which can detect all 16 DTMF digits including the digits ABCD. The tone detector can be configured to operate on either the incoming (record) data stream or outgoing (playback) data stream.

3.21 Full Duplex Speakerphone

The CT8021 can operate in either **Standalone** Speakerphone or in **DSVD** Speakerphone mode (for use in a videoconferencing system). In DSVD Speakerphone mode, the CT8021 provides Acoustical Echo Cancellation between the speaker and microphone connection concurrent with speech compression operation. In Standalone Full duplex Speakerphone mode, Acoustical Echo Cancellation as well as near-end (Electrical) echo cancellation to the telephone line connection are provided. Standalone Speakerphone mode provides support for creating an analog phone line based speakerphone.

Unlike many Speakerphone implementations, the CT8021 implements a **Full Duplex Speakerphone**. In most speakerphones, only one party can talk at once, the other party is always muted, sometimes by as much as 50 dB or more. This is **Half Duplex Speakerphone** operation. In half-duplex operation a Speakerphone “listens” to both the microphone and the telephone line inputs and tries to detect which direction is talking. It allows input from the talking direction and mutes input from the other direction. If this were not done, acoustical feedback or “howling” would take place. The major problem with a half-duplex Speakerphone is that a user experiences “break up” in the conversation when the Speakerphone incorrectly identifies the talking party (due to noise) or both parties attempt to talk at the same time.

The CT8021 implements a Full Duplex Speakerphone. This implementation allows both parties to speak at once and be heard by the party at the opposite end of the call. It does this using DSP technology called acoustical echo cancellation. Acoustical echoing occurs when the Speakerphone microphone picks up the audio signal from the Speakerphone speaker, “echoing” the output (speaker) signal back along the telephone line. The signal sent down the telephone line is also “echoed” by the telephone line and re-output to the speaker. This creates a “feedback” loop : the signal from the phone line is output from the speaker, picked up by the microphone, sent to the phone line, echoed by the phone line and re-output by the speaker. This causes “howling” or “howl round”, a loud annoying high pitched noise. In the CT8021, “howling” is prevented by the use of echo cancellation. This is an adaptive filtering process where the CT8021 compensates for the speaker-to-microphone echo and subtracts the outgoing speaker signal from the incoming microphone signal reducing or eliminating the acoustic echo.

See Appendix A for more details on speaker phone operation

3.22 Acoustical Echo Cancellation

The CT8021 includes an Acoustical Echo Canceller (AEC) as part of its Full Duplex Speaker phone feature. In addition to its use in standalone speakerphone mode, the AEC can be operated concurrently with the CT8021 full duplex speech mode (DSVD mode). This enables the CT8021 to provide acoustic echo cancellation between the speaker and microphone while simultaneously providing full duplex speech compression/decompression. This makes possible “hands free” DSVD (DSVD or Video Conferencing Speakerphone) applications avoiding use of a headset or handset for speech input/output. The AEC implementation included in the CT8021 contains features that allow the echo cancellor to be used to cancel non-acoustical echoes, for example the electrical signal echo from a 2-to-4 wire telephone hybrid. This option supports use of the CT8021 in a digital speech telephony gateway type application.

3.23 8 / 16 bit host controller interface

The operation of the CT8021 is controlled via 8 16-bit registers which provide a control/status/data interface. This interface is accessed physically as 16 8-bit wide I/O locations mapped into the host controller address space. This allows the CT8021 to be used with inexpensive 8-bit micro-controllers or more powerful 16/32 bit microprocessors. The host controls the CT8021 by writing control words to the CT8021 registers and it obtains status information by reading the CT8021 status registers. Speech data is passed through the receive/transmit buffer registers when recording or playing speech data to or from the host. Receive and Transmit data buffers are 16 words deep and Host controllable. In addition, CT8021 provides either single cycle or continuous (burst) DMA modes for both half duplex and full duplex speech transfers.

3.24 CODEC Interface

The CT8021 supports direct connection to one or two external 8-bit G.711 A-law/mu-law or 16-bit Linear CODECs for input and output of audio signals. If two codecs are used, both codecs must be of the same type. The CT8021 can act either as codec clock master, or be driven by an external codec clock. In master mode the codec clock signals SCLK and FSYNC are generated by programmable integer division from the internal CT8021 clock.

3.25 CT8021 Crystal

The CT8021 includes a clock frequency multiplying PLL (phase-locked-loop) to avoid requiring a high frequency external clock source for generation of the internal 45.056 MHz CT8021 clock. The PLL allows generation of the main internal 45.056 MHz DSP core clock from an external 4.096 MHz primary clock or crystal using a x11 multiplication factor. The PLL circuit also includes a by-pass control pin to allow the direct use of an external 2X (90.102 MHz) clock source applied to the XIN pin.

If the PLL is enabled, the CT8021 primary 45.056 MHz clock can be provided either by attaching a 4.096 MHz crystal at XIN/XOUT, or by providing a 4.096 MHz clock input at XIN from an external oscillator. Operation of the PLL is independent of the nature of the primary clock source. If the PLL is disabled, then clock must be provided by an external 90.102 MHz clock applied at XIN. The XIN-XOUT internal oscillator is not capable of supporting operation with an 90.102 MHz crystal.

Operation of the CT8021 at precisely 45.056 mips is needed only if the CT8021 is required to generate codec SCLK at 2.048 MHz and FSYNC at 8.000 KHz ($45.056 \text{ MHz} = 22 * 2.048 \text{ MHz}$). Selection of the operating frequency of the CT8021 is affected by the desired modes of operation and the desired sample rates. 45.056 MHz is the maximum operating frequency of the CT8021 and provides sufficient mips for operation of the G.728 speech coder plus the AEC. If concurrent operation of the AEC is not required, the operating frequency may be reduced in order to reduce power consumption.

3.26 Power Save Modes

The CT8021 supports internal clock slow down modes where the clock rate to the internal DSP core can be reduced. Operation in a slow down mode affects only the clocks to the CT8021 DSP core. The operation of the codec and host interfaces is not affected by slowing down the DSP core. The CT8021 also has a minimum power mode, where all internal clocks are stopped. The internal clock scaling feature of the CT8021 enables the host to adjust the available mips (and power consumption) to provide just sufficient processing power for the mode of operation used. For example operation of the TrueSpeech 8.5 speech compression functions require significantly less mips than the G.728 function.

4. HOST CONTROLLER INTERFACE

4.1 8 or 16 bit host controller interface

The operation of the CT8021 is controlled via 8 16-bit registers which provide a control/status/data interface. This interface is accessed physically as 16 8-bit wide I/O locations mapped into the host controller address space. This allows the CT8021 to be used with inexpensive 8 bit micro-controllers or more powerful 16/32 bit microprocessors. The host controls the CT8021 by writing control words to the CT8021 registers and it obtains status information by reading the CT8021 status registers. Speech data is passed through the receive/transmit buffer registers when recording or playing speech data to or from the host. Receive and Transmit data buffers are 16 words deep and Host controllable. In addition, CT8021 provides either single cycle or continuous (burst) DMA modes for both half duplex and full duplex speech transfers.

4.2 Host Interface Signals

HSTRDN	Active low: asserted by the host to read from the CT8021
HSTWRN	Active low: asserted by the host to write to the CT8021
HSTCSN	Active low: Chip Select, asserted by the host to enable the HSTRDN, HSTWRN signals
HSTAB3-0	Host Address lines, select the host interface register read or written
HSTDB7-0	8 bit host data bus to transfer data to/from the CT8021
IRQN	Active low: Dedicated host interrupt request (replaces CT8015 ACK signal)
TXDREQ	TX DMA request (associated with data write buffer)
TXDACKN	Active low: TX DMA acknowledge
RXDREQ	RX DMA request (associated with data read buffer)
RXDACKN	Active low: RX DMA acknowledge

4.3 CT8021 Host Connection with external DMA controller

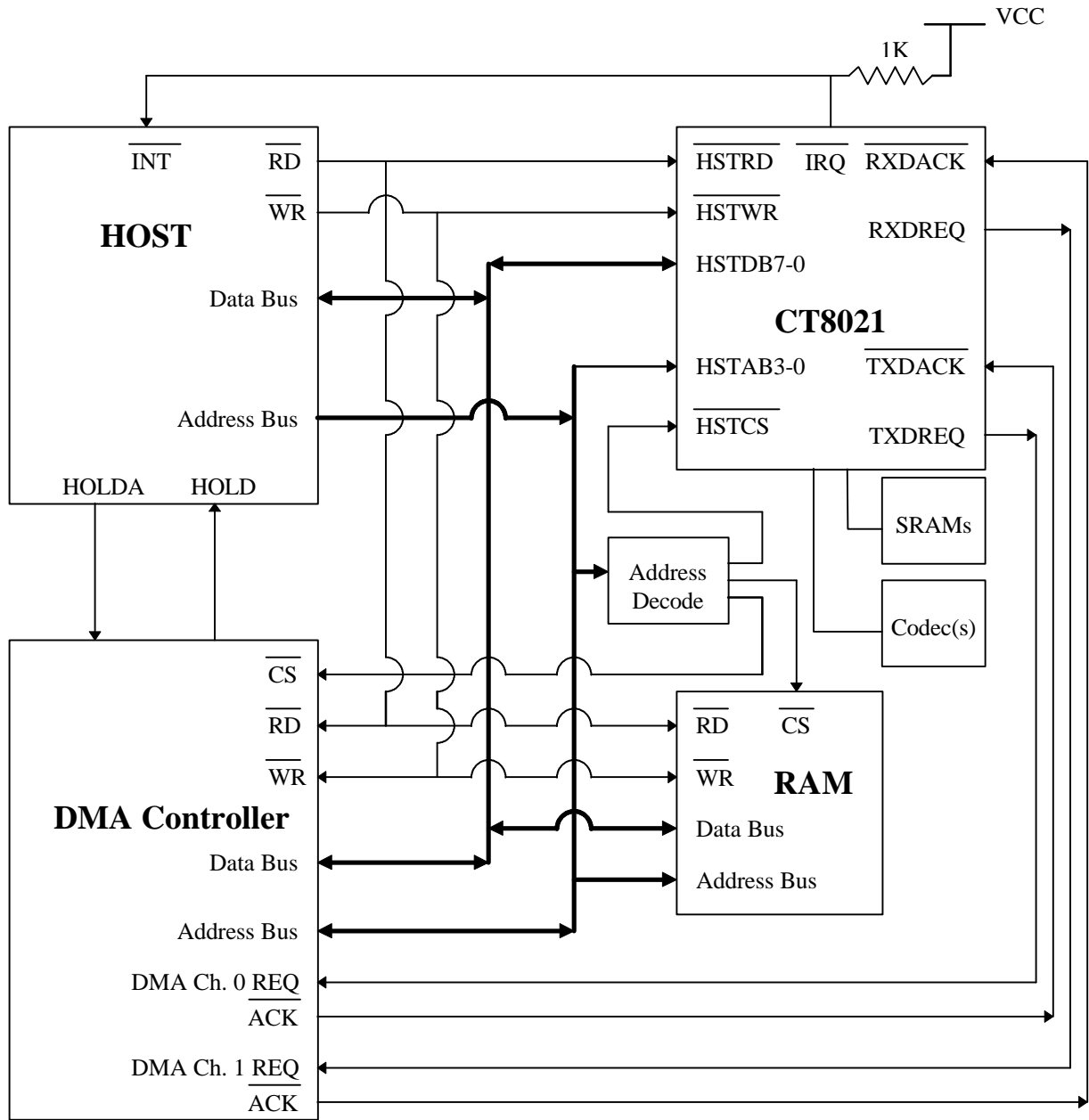


Figure 4-1 CT8021 connection to Host and external DMA controller

4.4 Host Programmers Model

Host Byte Address HSTAB3-0	Host Word Address HSTAB3-1	Function	Access
0-1	0	Aux Software Status Register (Reserved)	Read Only
2-3	1	Software Status Register	Read Only
4-5	2	Aux Software Control Register (Reserved)	Read/Write
6-7	3	Software Control Register	Read/Write
8-9	4	Hardware Status Register	Read Only
A-B	5	Hardware Control Register	Read/Write
C-D	6	Host Transmit (Write) Data Buffer Access Port (buffer input)	Write Only
E-F	7	Host Receive (Read) Data Buffer Access Port (buffer output)	Read Only

Note that the CT8021 registers are all internally 16-bits wide. Physical access to these registers is made using two 8-bit access cycles via the Host Interface. The lower byte of each register is accessed when address line HSTAB0 = 0. The upper byte of each register is accessed when address line HSTAB0 = 1. The registers should always be accessed low byte first followed by high byte. Accesses to the upper byte of the Software Status, Software Control, and Data Buffer Access Port Registers trigger certain internal events within the CT8021. For example reading the upper byte of the Software Status Register clears the Status Ready bit in the Hardware Status Register.

4.5 Register Descriptions

4.5.1 Hardware Control Register

This register has read/write access from the host.

0	0	0	0	0	Host IRQN Master Enable	CONTINUE	DMA Direction
(15)	(14)	(13)	(12)	(11)	(10)	(9)	(8)

TX DMA Burst Mode	RX DMA Burst Mode	TX DMA Enable	RX DMA Enable	Aux Status Update IE	Status Update IE	TX Ready IE	RX Ready IE
(7)	(6)	(5)	(4)	(3)	(2)	(1)	(0)

TX DMA Burst Mode When this bit is set, the TX DMA interface operates in Burst Mode. In Burst Mode, the TXDREQ signal is continuously asserted if the TX Ready bit for the Host Transmit buffer is set.
 When this bit is clear, the TX DMA interface operates in Single Cycle Mode. In Single Cycle Mode, the TXDREQ signal is de-asserted every time the TXDACKN and HSTWRN signals are asserted, then re-asserted if the TX Ready bit is still set, after a delay (buffer not full).
 See section on DMA Transfers.

RX DMA Burst Mode When this bit is set, the RX DMA interface operates in Burst Mode. In Burst Mode, the RXDREQ signal is continuously asserted if the RX Ready bit for the Host Receive buffer is set.
 When this bit is clear, the RX DMA interface operates in Single Cycle Mode. In Single Cycle Mode, the RXDREQ signal is de-asserted every time the RXDACKN and HSTRDN signals are asserted, then re-asserted if the RX Ready bit for the Host Receive Buffer is still set, after a delay (buffer not empty).
 See section on DMA Transfers.

DMA Direction: This field allows the host to use the DMA features in a full duplex or half duplex configuration. In a half-duplex configuration, the host needs to interface a single DMA channel to the CT8021, and needs to be able to transfer data in either direction with this single channel. This is accommodated within the CT8021 by permitting the functional exchange of the external (physical) DMA interface pins.
 See section on DMA Transfers.

0 Normal Direction. The TXDREQ, RXDREQ, TXDACKN and RXDACKN operate in the manner described in the CT8021 pin out. After reset, this bit is set to zero.

1	Reverse Direction. The physical pin functions of TXDREQ and RXDREQ, and also TXDACKN and RXDACKN are exchanged, i.e. the TXDREQ pin assumes the function and behavior of the RXDREQ pin and vice-versa. See the section on Half-Duplex DMA mode (page 28) for more information on this feature.
TX DMA Enable :	Setting this bit to 1 enables the TXDREQ and TXDACKN signals. If this bit is set to zero, the TXDACKN signal is ignored and the TXDREQ signal is not asserted. This control operates on the logical TX DMA signals, not the physical pins whose functions can be exchanged between RX and TX. After reset, this bit is zero.
RX DMA Enable :	Setting this bit to 1 enables the RXDREQ and RXDACKN signals. If this bit is set to zero, the RXDACKN signal is ignored and the RXDREQ signal is not asserted. This control operates on the logical RX DMA signals, not the physical pins whose functions can be exchanged between RX and TX. After reset, this bit is zero.
HOST IRQN Master Enable	Setting this bit to 1 enables the IRQN output signal. If this bit is not set the IRQN signal is not asserted. After reset, this bit is zero.
Aux Status Update IE	If this bit is set, the Host IRQN signal is asserted whenever the Aux Status Ready bit in the Hardware Status Register is set. After reset, this bit is zero.
Status Update IE	If this bit is set, the Host IRQN signal is asserted whenever the Status Ready bit in the Hardware Status Register is set. After reset, this bit is zero.
TX Ready IE	If this bit is set, the Host IRQN signal is asserted whenever the TX Ready bit in the Hardware Status Register is set. After reset, this bit is zero.
RX Ready IE	If this bit is set, the Host IRQN signal is asserted whenever the RX Ready bit in the Hardware Status Register is set. After reset, this bit is zero.
CONTINUE	The host writes a 1 to this bit to clear the COHOST SHUTDOWN bit in the Hardware Status Register. This returns the host and codec logic to normal operation from sleep mode. This bit always reads as a 0. When COHOST SHUTDOWN is in operation, only the continue bit in the Hardware Control Register is accessible. All other bits are undefined for read and write operations. COHOST SHUTDOWN is a power save state where the CODEC and HOST sections of the CT8021 are disabled in order to reduce power consumption. In this state it is still possible for the CT8021 DSP core to remain active.

4.5.2 Hardware Status Register

This register is read only accessible by the Host.

0	0	0	0	0	COHOST SHUT DOWN	TX DMA	RX DMA
(15)	(14)	(13)	(12)	(11)	(10)	(9)	(8)

IRQN Asserted	SYSTEM RESTART	TX Ready	RX Ready	Aux Status Ready	Status Ready	Aux Control Ready	Control Ready
(7)	(6)	(5)	(4)	(3)	(2)	(1)	(0)

SYSTEM RESTART: reserved

TX Ready: This bit reflects the state of the TX Ready bit in the DSP’s Host Transmit Buffer control register. This bit is set to indicate that the Host Transmit Data Buffer can be accessed by the host. When the host writes to the high byte of the Host Transmit Data Buffer access port and the Frame Size limit is reached (buffer full), this bit is cleared. After reset, this bit is clear.

When this bit is set and the TX Ready IE bit in the Hardware Control Register is set, the Host IRQN signal is asserted.

TX DMA: This bit is set if the (logical) TXDREQ signal is asserted. The state of this bit is affected by the TX DMA enable bit in the host interface hardware control register. If the TX DMA enable is not set, this bit will always be zero.

RX Ready: This bit reflects the state of the RX Ready bit in the DSP’s Host Receive Buffer control register. This bit is set to indicate that the Host Receive Data Buffer can be accessed by the host. When the host reads from the high byte of the Host Receive Buffer access port, and the Frame Size limit is reached (buffer empty), this bit is cleared. After reset, this bit is clear.

When this bit is set and the RX Ready IE bit in the Hardware Control Register is set, the Host IRQN signal is asserted.

RX DMA: This bit is set if the (logical) RXDREQ signal is asserted. The state of this bit is affected by the RX DMA enable bit in the host interface hardware control register. If the RX DMA enable is not set, this bit will always be zero.

Status Ready: This bit is set when the Oak core writes to the Software Status Register. When the host reads the high byte of the Software Status Register, this bit is cleared. After reset, this bit is cleared.

When this bit is set and the Status Ready IE bit in the Hardware Control Register is set, the Host IRQN signal is asserted.

IRQN Asserted:	This bit reflects the state of the external IRQN pin. If the IRQN signal is asserted, this bit is set. If the Host IRQN Master Enable is not set, this bit will always be zero.
Control Ready:	This bit is set when the Oak core reads from the Software Control Register. When the host writes to the high order byte of the Software Control Register, this bit is cleared. After reset, this bit is cleared to indicate that the CT8021 is not ready to receive commands. Once the CT8021 is ready to begin communication with the host, it performs a dummy read of the Software Control Register to set the Control Ready bit.
Aux Status Ready:	This bit is set when the Oak core writes to the Aux Software Status Register. When the host reads the high byte of the Aux Software Status Register, this bit is cleared. After reset, this bit is cleared. When this bit is set and the Aux Status Ready IE bit in the Hardware Control Register is set, the Host IRQN signal is asserted.
Aux Control Ready:	This bit is set when the Oak core reads from the Aux Software Control Register. When the host writes to the high order byte of the Aux Software Control Register, this bit is cleared. After reset, this bit is cleared to indicate that the CT8021 is not ready to receive commands. Once the CT8021 is ready to begin communication with the host, it performs a dummy read of the Aux Software Control Register to set the Aux Control Ready bit.
COHOST SHUTDOWN	This bit indicates that the codec and host interface logic is in sleep mode (power save). The host interface must be re-awakened by writing to the CONTINUE bit in the Hardware Control Register before writing or reading from any other host register. When COHOST SHUTDOWN is active, all the other bits in the Hardware Status Register are undefined for read operations.

4.5.3 Software Control Register

These is a 16-bit register. It is used to pass commands from the host to the CT8021. The host can write commands into this register and also read back the command after it has been written.

When the host writes to the upper byte of one of this register, it clears the Control Ready bit in the Hardware Status Register.

When the CT8021 internally reads from this register, it sets the Control Ready bit in the Hardware Status Register.

The CT8021 hardware includes a second “Aux Software Control Register”. This register is reserved and is not used by the internal CT8021 firmware.

4.5.4 Software Status Register

This is a 16-bit register. It is used to pass command status/result information from the CT8021 back to the host. The host can only read from this register.

When the CT8021 updates this register, it sets the Status Ready bit in the Hardware Status Register and also asserts IRQN to the host (if enabled).

When the host reads the upper byte of one of this register, it clears the Status Ready Hardware Status Register. The IRQN will also clear at this time, if enabled.

The CT8021 hardware includes a second “Aux Software Status Register”. This register may be used to provide an independent status register for reporting the status of the DTMF and Call Progress Tone detector. This allows the CT8021 to provide an “interrupt on DTMF” feature when used in conjunction with the interrupt enable controls in the Hardware Control Register.

4.5.5 Host Receive and Transmit Data Buffer Blocks

The CT8021 includes **two** 16-word (32 byte) data buffer blocks for use in transferring blocks of data to or from the host. Access to these buffers is controlled by the CT8021. The host accesses these buffers in sequential address order through two 16-bit Host Data Buffer access ports (one for RX and one for TX). The host address sequence is generated by an internal address counter within the CT8021.

Each buffer is shared between the Host and CT8021. At any one time, either the Host or the CT8021 has access to the buffer, at no time can both Host and CT8021 access the buffer simultaneously. The CT8021 controls whether access belongs to the CT8021 or Host via an internal control register. For example, when the host transmits data to the CT8021, the CT8021 first grants the host access to the buffer. The host fills the “empty” buffer. While the host is filling the buffer, the CT8021 can not access it. Once the buffer is full, the CT8021 regains access to the “full” buffer. The CT8021 then copies the data from the buffer. At this point the host can not access the buffer. Once the buffer has been emptied, the CT8021 can then pass it back to the host.

The receive and transmit buffer blocks are conceptually almost identical except for the direction of data transfer associated with host accesses. The Receive buffer is used to transfer data from the CT8021 to the host. The Transmit buffer is used to transfer data from the host to the CT8021.

Host Receive Data Buffer Block

This block is used to transfer data from the CT8021 to the host, using an external DMA controller or a host processor access cycle.

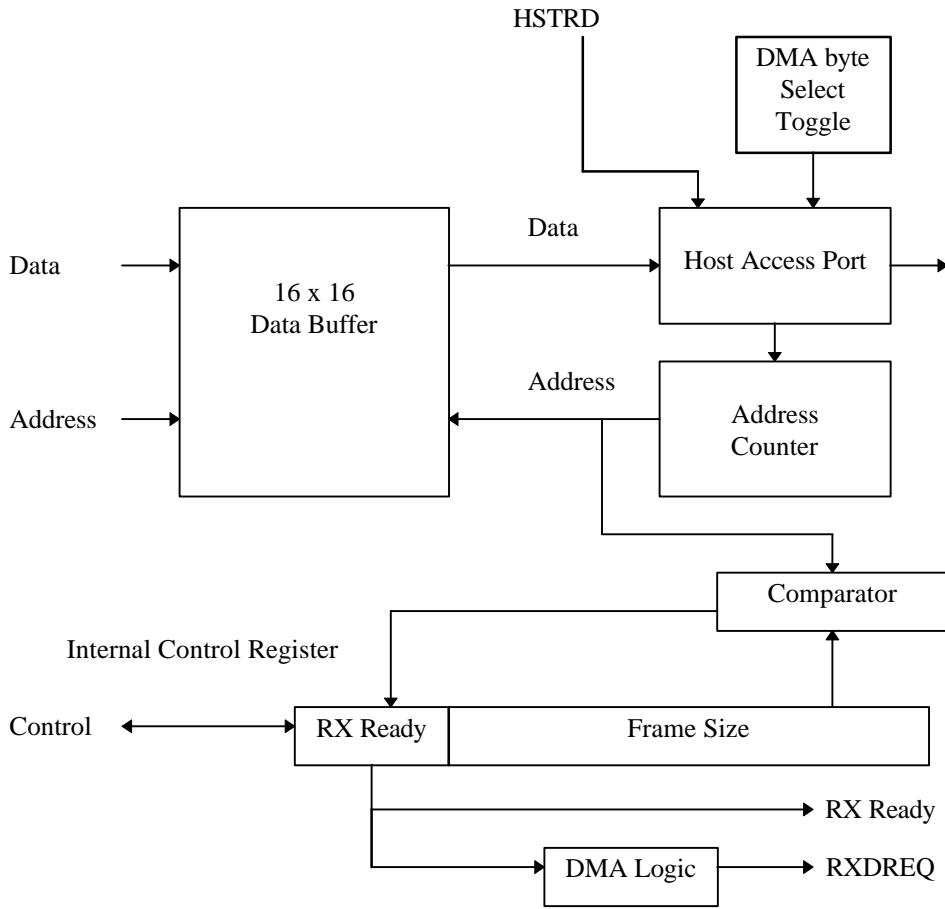


Figure 4-2 Host Receive Data Buffer Block

RECEIVE DATA BUFFER INTERNAL CONTROL REGISTER

0000 0000 (15-8)	RX Ready (7)	00 (6-5)	Frame Size (4-0)
---------------------	-----------------	-------------	---------------------

This is an internal register, and is not accessible by the host.

RX Ready When set to zero, the CT8021 has control of the data buffer and has write only access to any location in the buffer.

When set to one, the host has control of the data buffer and may read its contents via the Host Data Buffer access port.

The state of this bit can be checked by the Host at any time since it is reflected by the RX Ready bit in the Hardware Status register. The RX Ready bit in the Receive Data Buffer Control Register drives the RX Ready bit and also RXDREQ via the DMA logic.

After reset, this bit is zero.

Frame Size The Frame Size field determines the number of words contained in the data buffer. It acts as the reference input to the host buffer address comparator.

ADDRESS COUNTER

The 5-bit address counter provides the sequential buffer access address for access by the host via the Host Data Buffer access port. The address counter is reset to zero when the CT8021 writes a 1 to the RX Ready bit of the internal control register causing control of the buffer to transfer to the Host. Each time the host accesses the upper byte of the Host Data Buffer access port the address counter increments by 1.

The data buffer access port is only physically 8-bits wide. When an access to this port is made by a host processor, the byte select address line HSTAB0 is valid and can be used to select the byte accessed. When access is made by a DMA controller, the address decode which selects the data buffer access port is provided by the RXDACKN signal. During a DMA cycle, the HSTAB3-0 address bits are not valid. Generation of an internal HSTAB0 is thus required to select the byte accessed. This is provided by a simple toggle mechanism which changes state on each DMA access. The DMA byte select toggle is set to zero each time the RX Ready bit changes from a zero to a one (this also clears the main address counter). A separate DMA byte select toggle is required for transmit and receive.

COMPARATOR

The 5-bit comparator is active only when the Host has control (the RX Ready bit is set). The comparator compares the value in the address counter with the value in the Frame Size field. If the values are not equal, the host continues to have access to the data buffer. When the values become equal (after the last host access), the RX Ready bit is reset, transferring control back to the CT8021.

TRANSFERRING DATA FROM CT8021 TO HOST

The CT8021 determines the size of the data frame to be transferred by some protocol with the Host. At the beginning of data transfer, the RX Ready bit will be zero. The CT8021 then writes the appropriate number of words into the data buffer. It then programs the Frame Size and sets the RX Ready bit. This causes the RX Ready bit visible to the Host in the Hardware Status Register to be set. The host discovers that the RX Ready bit is set (by polling or interrupt). The Host then reads Frame Size words from the Host Receive Buffer Access Port, after each word is read, the address counter increments. When the host reads the last word, the address counter is incremented and matches the Frame Size value. This is detected by the comparator which causes the RX Ready bit to be cleared. The CT8021 determines that it has access to the data buffer and loads the next frame into the buffer. This process is repeated to transfer the next frame.

For DMA transfers, the RX Ready bit drives the state of the DMA Request signal. When RX Ready is set, the DMA request is asserted. When the buffer becomes empty, and buffer control automatically returns to the CT8021, the RX Ready bit is cleared and the DMA Request signal is de-asserted.

Host Transmit Data Buffer Block

This block is used to transfer data from the host to the CT8021, using an external DMA controller or a host processor access cycle.

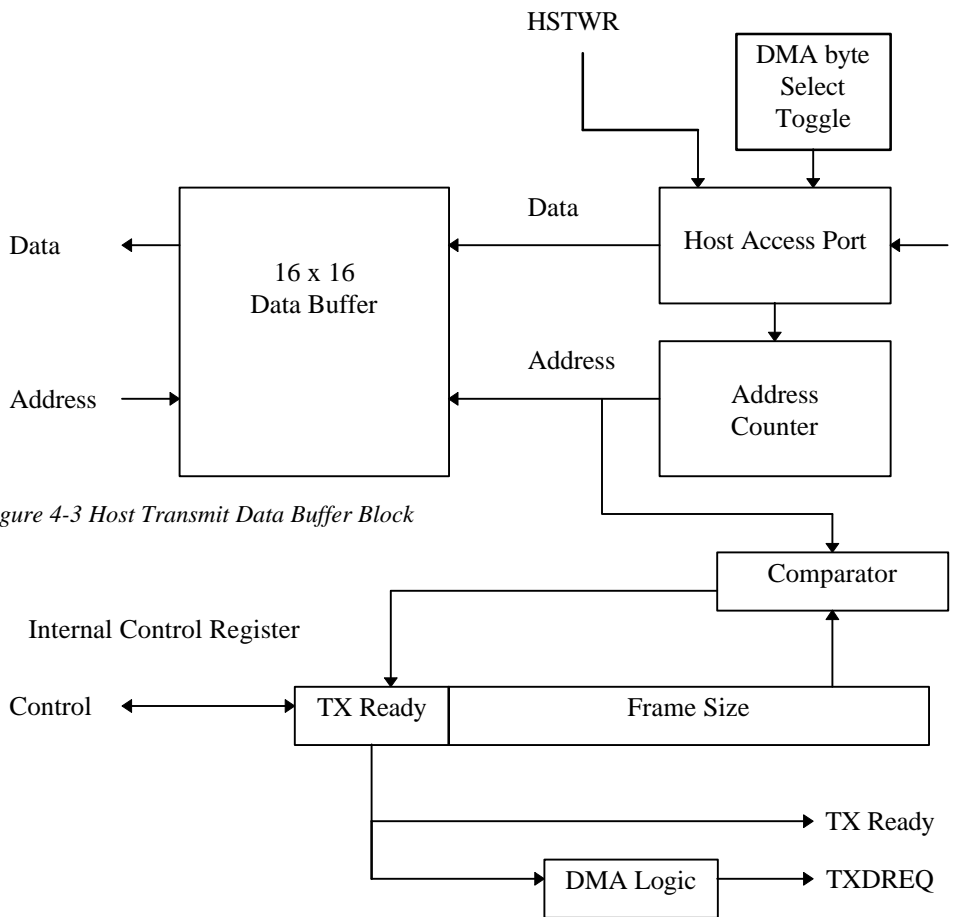


Figure 4-3 Host Transmit Data Buffer Block

TRANSMIT DATA BUFFER INTERNAL CONTROL REGISTER

0000 0000 (15-8)	TX Ready (7)	00 (6-5)	Frame Size (4-0)
---------------------	-----------------	-------------	---------------------

This is an internal register, and is not accessible by the host.

TX Ready When set to zero, the CT8021 has control of the data buffer and can read any location in the buffer.

When set to one, the host has control of the data buffer and may write its contents via the Host Data Buffer access port.

The state of this bit can be checked by the Host at any time since it is reflected by the TX Ready bit in the Hardware Status register. The TX Ready bit in the Transmit Data Buffer drives the TX Ready bit and also TXDREQ via the DMA logic.

After reset, this bit is zero.

Frame Size The Frame Size field determines the number of words contained in the data buffer. It acts as the reference input to the host buffer address comparator.

ADDRESS COUNTER

The 5-bit address counter provides the sequential buffer access address for access by the host via the Host Data Buffer access port. The address counter is reset to zero when the CT8021 writes a 1 to the TX Ready bit of the internal control register causing control of the buffer to transfer to the Host. Each time the host accesses the upper byte of the Host Data Buffer access port the address counter increments by 1.

The data buffer access port is only physically 8-bits wide. When an access to this port is made by a host processor, the byte select address line HSTAB0 is valid and can be used to select the byte accessed. When access is made by a DMA controller, the address decode which selects the data buffer access port is provided by the TXDACKN signal. During a DMA cycle, the HSTAB3-0 address bits are not valid. Generation of an internal HSTAB0 is thus required to select the byte accessed. This is provided by a simple toggle mechanism which changes state on each DMA access. The DMA byte select toggle is set to zero each time the TX Ready bit changes from a zero to a one (this also clears the main address counter). A separate DMA byte select toggle is required for transmit and receive.

COMPARATOR

The 5-bit comparator is active only when the Host has control (the TX Ready bit is set). The comparator compares the value in the address counter with the value in the Frame Size field. If the values are not equal, the host continues to have access to the data buffer. When the values become equal (after the last host access), the TX Ready bit is reset, transferring control back to the CT8021.

TRANSFERRING DATA FROM HOST TO CT8021

The CT8021 determines the size of the data frame to be transferred by some protocol with the Host. At the beginning of data transfer, the TX Ready bit will be zero. The CT8021 then programs the Frame Size and sets the TX Ready bit (in the same write cycle). This causes the TX Ready bit visible in the Hardware Status Register to be set. The host discovers that the TX Ready bit is set (by polling or interrupt). The Host then writes Frame Size words into the Host Transmit Buffer Access Port, after each word is written, the address counter increments. When the Host writes the last word, the address counter is incremented and matches the Frame Size value. This is detected by the comparator which causes the TX Ready bit to be cleared. The CT8021 discovers that it has access to the data buffer and reads the buffer contents, transferring them to the CT8021's local RAM. This process is repeated to transfer the next frame.

For DMA transfers, the TX Ready bit drives the state of the DMA Request signal. When TX Ready is set, the DMA request is asserted. When the buffer becomes full, and buffer control automatically returns to the CT8021, the TX Ready bit is cleared and the DMA Request signal is de-asserted.

4.6 DMA Transfers

The CT8021 is intended to interface directly to one or two external 8-bit DMA controllers connected to the host bus. The DMA controllers may be of the “fly-by” type (used in IBM compatible PCs) or the “flow-through” type (for example -used in the Intel 80186).

In a fly-by DMA transfer, the DMA controller simultaneously generates the address of the source or destination memory location, the read/write signal to the memory, a read/write signal to the peripheral, and a DMA acknowledge signal (which enables the destination or source peripheral). In a “read from peripheral” cycle, the DMA Acknowledge signal plus the peripheral read strobe, cause the peripheral to gate its’ data onto the data bus. The DMA controller provides the destination memory address and asserts the memory write signal causing the peripheral’s data to be written to the memory. In a “write to peripheral cycle”, the DMA controller generates the memory address and asserts the memory read signal causing the appropriate data byte to appear on the data bus. The DMA controller also asserts the DMA Acknowledge signal plus the peripheral write strobe, which causes the peripheral to read the data currently on the data bus. This type of DMA controller requires DMA request **and** DMA acknowledge signals. The address signals are ignored by the peripheral, since they indicate the memory address not the peripheral address. Instead the peripheral uses the DMA acknowledge signal plus the read/write strobe to indicate that it has been selected.

The alternative type of DMA controller which uses a “flow-through” transfer is supported by the normal host processor data transfer interface. In a flow-through transfer, the read/write access to the peripheral and the write/read memory access are performed as two separate cycles, with the DMA controller providing temporary storage of the data. In this type of transfer, the DMA access cycle to the CT8021 is identical to a host processor access cycle. In a “read from peripheral” cycle, the DMA controller first drives the address bus with the address of the peripheral, and asserts HSTCS/ and HSTRD/. The DMA controller temporarily saves the peripheral data and then writes the data to memory by performing a normal memory access cycle. This type of controller requires the DMA request signals **only**, and does not use the DMA acknowledge. When used with this type of controller, the CT8021 DMA acknowledge signals should be connected to VCC via pull-up resistors.

4.6.1 DMA modes

The DMA interface can be operated in two modes : full-duplex and half-duplex.

4.6.1.1 Full-Duplex Mode

In Full-Duplex mode, the RX and TX DMA interfaces operate independently of each other, with physically separate control signals dedicated to the two data transfer directions. Two external DMA channels are required.

4.6.1.2 Half-Duplex Mode

In Half-Duplex mode, only a single external DMA channel is required. The external DMA controller will have only a single DMA Request line available and a single DMA Acknowledge line. Using only these two signals, the DMA controller must be able to transfer data both to and from the CT8021. To accommodate this arrangement without requiring external glue logic, the TXDREQ and RXDREQ and the TXDACKN and RXDACKN pin functions can be swapped by the host using the control bits in the Hardware Control Register. This allows the DMA controller to be hard-wired to one set of DMA pins, and permits software control of the transfer direction associated with those pins.

4.6.2 Burst Mode and Single Cycle Mode Transfers

DMA transfers can be performed in either Burst Mode (continuous) or Single Cycle mode. In Burst Mode, the DMA request signal TXDREQ or RXDREQ remains asserted whenever the appropriate data buffer can accommodate the transfer. Each time the DMA acknowledge signal, TXDACKN + HSTWRN or RXDACKN + HSTRDN, is asserted a byte is transferred. This continues until the Transmit Data Buffer is full or the Receive Data Buffer is empty, at which point the DMA request signal is de-asserted (the appropriate TX or RX Ready bit is cleared). Burst Mode allows for the fastest transfer of data, since the Host is required to perform bus request/bus grant arbitration with its external DMA controller only once per burst. With a 16-word buffer, up to 32 bytes can be transferred in a single burst.

In Single Cycle mode, the DMA request signal, TXDREQ or RXDREQ is asserted then de-asserted once for each byte transferred. This mode requires the Host to re-arbitrate bus access with the external DMA controller for each byte transferred. When the CT8021 is ready to perform a DMA transfer, the DMA request signal is asserted. The DMA controller responds with the DMA acknowledge signal (plus HSTRD/ or HSTWR/) which clocks the byte data transfer and also causes the DMA request signal to be withdrawn. After the byte transfer has completed, the CT8021 delays for a short period of time and then re-asserts the DMA request signal (assuming it has more data to transfer).

Single Cycle Mode

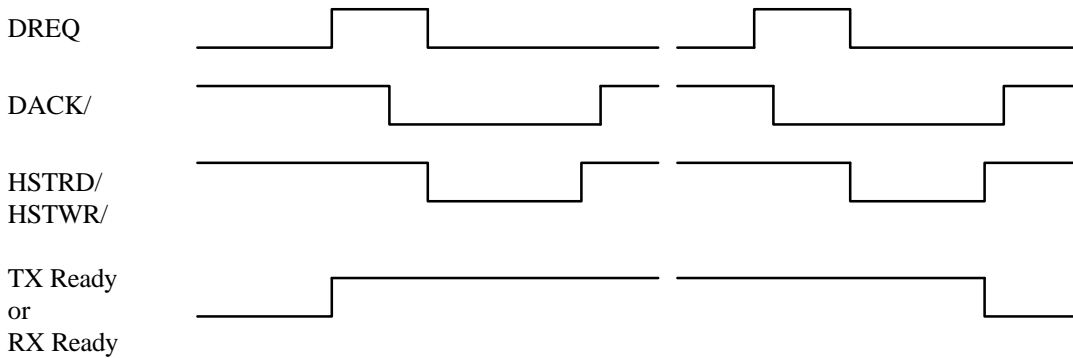


Figure 4-4 Single Cycle DMA mode

In single cycle mode, the negative going edge of HSTRD/ or HSTWR/ during DACK/, causes the DREQ signal to be de-asserted. If RX or TX Ready bit is still set after the end of the DMA cycle, then DREQ will be re-asserted 16 MAINCLOCK clock periods after the positive going HSTRD/ or HSTWR/ edge. At 40 Mips (25ns) this translates to 400 ns. The address counter used to index the data buffer during Host/DMA cycles is incremented by the positive going edge of HSTRD/ or HSTWR/ on alternate DMA cycles (i.e. each time a high byte is accessed). The data buffer block contains a toggle bit which is used to generate a DMA byte select. This toggle bit is used in place of HSTAB0 during a DMA cycle, it changes state at the end of each DMA cycle to provide the correct byte access sequence : low-byte, high-byte, low-byte, high-byte etc. The host address lines HSTAB3-0 are ignored during a DMA cycle. The toggle bit is set to zero, each time the data buffer address counter is reset (when the CT8021 writes a 1 to the RX or TX Ready bit). If the counter reaches the Frame Size limit value, then the RX or TX Ready bit is reset following the positive HSTRD/ or HSTWR/ edge.

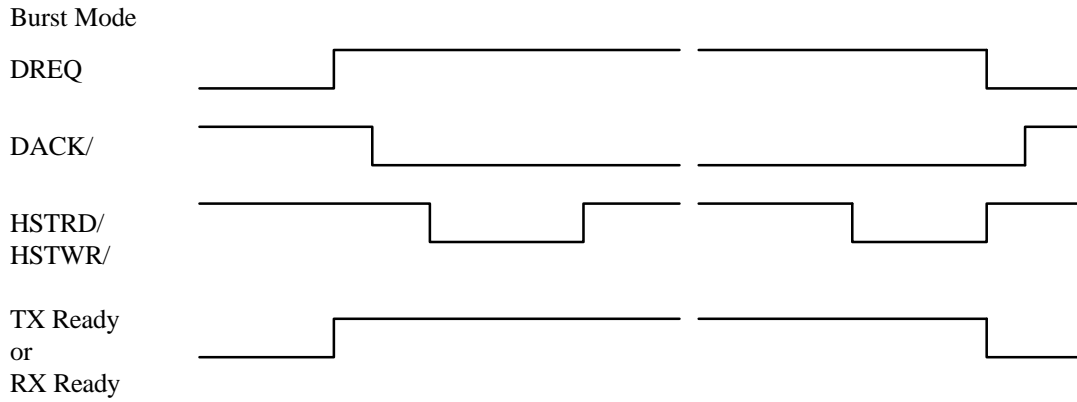


Figure 4-5 Burst Mode DMA

In Burst mode, DREQ remains asserted until the RX or TX Ready bit is cleared by the final counter increment. Byte transfers are clocked by the HSTRD/ or HSTWR/ signals as appropriate. The positive going edge of these signals toggle the DMA byte select. The data buffer address counter increments on alternate cycles, i.e. each time a high byte is accessed. When the counter reaches the Frame Size limit the RX or TX Ready and DREQ signals are cleared.

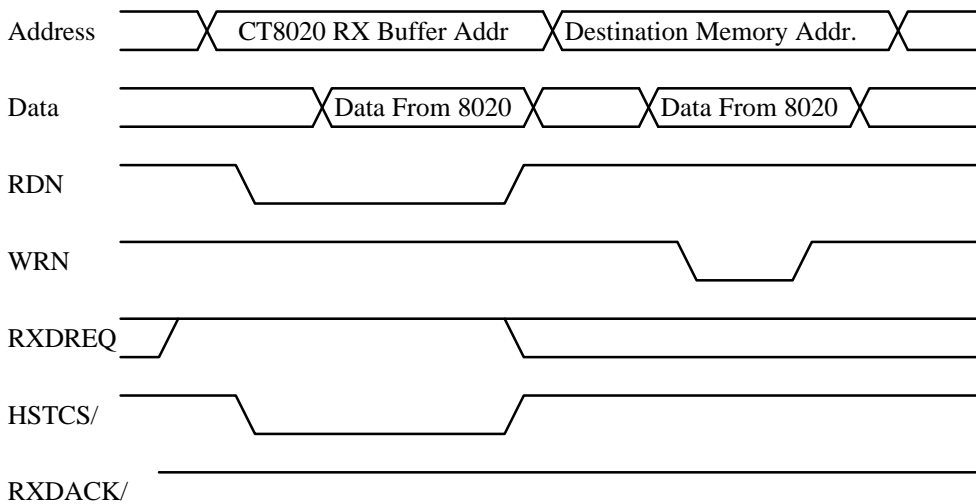
4.6.3 Flow-Through DMA Transfers

PROGRAMMERS NOTE:

A “Flow-Through” DMA transfer is one in which the read and write operations of the DMA transfer take place as separate cycles, with the DMA controller providing intermediate temporary storage of the data. In this type of DMA, the data physically “flows through” the DMA controller:

- DMA controller outputs source address
- DMA controller reads from source
- DMA controller stores data in internal temporary register (single byte)
- DMA controller outputs destination address
- DMA controller writes data to destination.

Figure 4-6 Flow-Through DMA



Example of a CT8021 to Memory DMA transfer using “Flow-Through” DMA. The RXDACK/ signal is not used.

Note that, from the perspective of the CT8021, an access by a “Flow-Through” type DMA controller looks identical to a normal Host access. The Intel 80186 and 80188 processors contain this type of DMA controller.

5. CT8021 CODEC INTERFACE

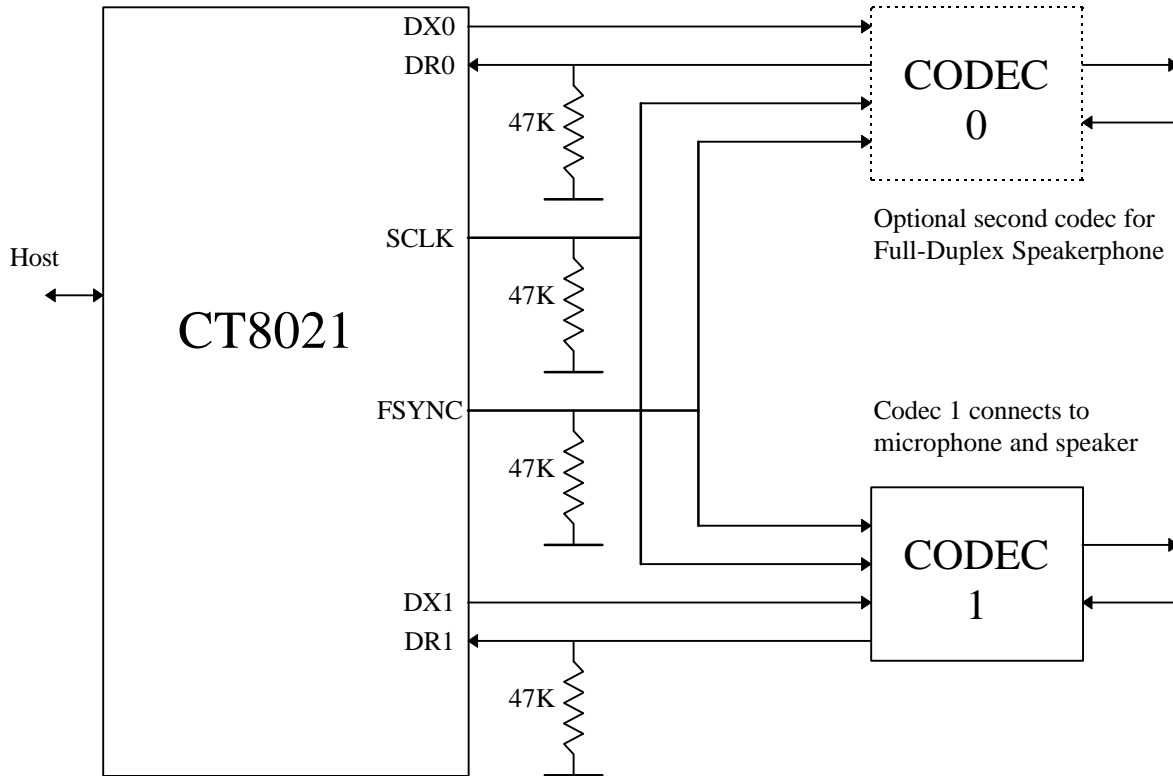


Figure 5-1 CT8021 Codec Interface connection

In Master mode, the CT8020 generates the FSYNC and SCLK signals.

In Slave mode, the FSYNC and SCLK signals are generated externally. In slave mode, the CT8020 FSYNC and SCLK pins are inputs.

Note that during and after reset, the SCLK and FSYNC pins are configured as inputs. As such, they require external pull-down resistors to ensure that a safe and defined logic level is present.

5.1 Codec Options

The CT8021 codec interface supports the following features:

- Codec clock master or slave
- 8-bit A-law/Mu-law or 16-bit linear codec
- Short or Long FSYNC
- programmable SCLK and FSYNC clock rates

5.1.1 Master/Slave

The CT8021 can be operated in codec master or slave mode.

In master mode, the SCLK and FSYNC signal pins are outputs. The clock frequencies for the two signals are generated by programmable integer division from the internal CT8021 DSP core clock. To achieve a SCLK frequency of 2.048 MHz for operation with a A-law or Mu-law codec, the CT8021 internal clock should be 45.056 MHz (4.096 MHz crystal). $45.056 \text{ MHz} = 22 * 2.048 \text{ MHz}$.

In slave mode, the SCLK and FSYNC signals pins are inputs. The SCLK and FSYNC signals must be generated externally. For TrueSpeech® G.723.1 and G.728 operation the FSYNC frequency should be exactly 8.0 KHz. In slave mode it is possible to connect the CT8021 codec interface to a TDM (Time Division Multiplexed) bus.

For support of G.722 7 KHz wideband ADPCM (only), the CT8021 needs to be operated with an external codec that supports operation at 16 Ksamples/sec.

Following reset, the CT8021 codec interface will be in slave mode, so that SCLK and FSYNC are inputs.

5.1.2 8-bit A-law/Mu-law and 16-bit linear codec

The CT8021 codec interface can be programmed by the host to operate in 8-bit or 16-bit mode. In 8-bit mode for each FSYNC, 8 data bits are shifted out of the DX0 and DX1 pins, and 8 data bits are shifted in at the DR0 and DR1 pins. In 16-bit mode for each FSYNC, 16 data bits are shifted out of the DX0 and DX1 pins, and 16 data bits are shifted in at the DR0 and DR1 pins.

In 8-bit mode, the data format may be either Mu-law or A-law.

In 16-bit mode, the data format is 16-bit linear, with the m.s. bit shift in/out first.

5.1.3 Short or Long FSYNC

In master mode, the CT8021 may be operated in short or long FSYNC mode. In short FSYNC mode, the FSYNC pulse is 1 SCLK period wide. The timing of the FSYNC pulse relative to the data bits conforms to the telecom industry standard for short FSYNC for PCM A-law/Mu-law codes when $SCLK = 2.048 \text{ MHz}$ and $FSYNC = 8.00 \text{ KHz}$. In long FSYNC mode, the FSYNC pulse is 8 or 16 SCLK periods wide depending on the data width selected. In this mode, the FSYNC pulse is aligned with the receive and transmit “time slot” of the data bits.

In slave mode, the width of the FSYNC pulse is not important. Selecting long or short FSYNC mode, only affects the assumed timing position of the FSYNC relative to the data bits.

5.1.4 Programmable SCLK and FSYNC rates

In master mode, the SCLK and FSYNC rates are fully programmable by the host. This enables host selection of the codec sample rate based on simple integer division from the internal CT8021 clock.

SCLK is generated by a division-by-(N+1) from the internal CT8021 clock. The divisor may be in the range

$4 \leq (N+1) \leq 32$. If the internal CT8021 clock is 45.056, then setting $N = 21$ will produce a 2.048 MHz SCLK.

FSYNC is generated by a division-by-(M+1) from SCLK. The divisor may be in the range $18 \leq M \leq 1023$. If SCLK = 2.048 MHz and $M = 255$, then FSYNC will be 8.0 KHz.

The table below shows possible clock frequencies

MAINCLOCK = 45.056 MHz (4.096 MHz crystal)			
SCLK	FSYNC	N	M
2.048 MHz	8.000 KHz	21	255
2.048 MHz	11.011 KHz	21	185
2.816 MHz	11.043 KHz	15	254

Note that the AT&T T7525 16-bit linear codec requires that the SCLK frequency is exactly 256 times the FSYNC frequency. When using this codec SCLK must be 2.048 MHz in order to achieve a FSYNC frequency of 8.000 KHz.

For supported SCLK-to-FSYNC ratios, please consult the external codec manufacturer's datasheet. Note that some G.711 A-law/Mu-law codecs provide strapping options which access different available ranges for the SCLK-to-FSYNC ratio (e.g for operation at E1 2.048 MHz 32 channel rate or T1 1.544 MHz 24 channel rate). In general, the choice of SCLK frequency often controls the effective audio bandwidth of the codec's anti-aliasing filter.

For an 8-bit codec, the CT8021 requires that the SCLK frequency is at least 16 times the FSYNC frequency.

For a 16-bit codec, the CT8021 requires that the SCLK frequency is at least 32 times the FSYNC frequency.

The CT8021 may be operated at a lower core clock rate to reduce the power consumption of the device, provided that the clock rate chosen provides sufficient mips for the modes of operation used. Operating the CT8021 at a lower clock rate requires that the host change the SCLK and FSYNC division factors to obtain the desired A-to-D sample rate.

5.1.5 Stop Codec

In Master mode, the STOP CODEC command can be used to halt operation of the codec interface. In this state the FSYNC and SCLK signals are held low. Stop Codec in Slave mode has no effect.

6. HOST DATA TRANSFER MODES

Data transfer between the CT8021 and host can be performed in two ways:

- via the Transmit and Receive Data Buffers (recommended for new designs)
- via the Software Control and Software Status Registers (CT8015 compatible mode)

6.1 Data Transfer using the Data Buffers

This transfer mode is recommend for all new designs.

In this mode, the CT8021 supports DMA transfers as well as low-overhead host processor based transfers. Once the CT8021 playback/record options have be selected and activated, speech frames can be transferred with minimal or zero “control overhead” required for device polling or supervision.

6.1.1 DMA Transfers

The CT8021 provides a full-duplex DMA interface with independent Transmit and Receive, request and acknowledge signals. The host issues the commands to the CT8021 to start playback and/or record, then enables DMA transfers by programming the Hardware Control Register. Once playback/record operation has been setup, the CT8021 can continue to transfer speech frames without further involvement of the host. The host processor can optionally use the buffer management commands to observe and control the status of the speech data streams.

6.1.2 Host Processor Transfers

In systems where no DMA controller is available, the host processor can be used to perform the data transfers. The playback and/or record operation is setup as before, then the host processor can either configure the Hardware Control Register to generate an interrupt to the host on the TX/RX ready condition, or alternatively, the host may simply poll the TX/RX ready status periodically. Once the host is ready to perform the data transfer, it can perform read/write “bursts” of up to 16 words of data without needing to re-check the TX/RX ready status. The data buffers can contain a maximum of 16 words of data. This size is sufficient to accommodate a complete TrueSpeech data frame. This allows microprocessors with “repeat” instruction prefixes (e.g. Intel X86 : REP INS, REP OUTS) to perform block moves of compressed speech frames. When performing record/playback of non-compressed data, where the speech data frame size is greater than 16 words, the host may divide the data transfer into 16-word sub-blocks and re-check the TX/RX ready status between the sub-blocks.

6.2 Data Transfer using the Software Control and Status Registers

This transfer mode is provided for software compatibility with the CT8015 and CT8005.

In this mode, the CT8021 supports data transfers using only the Software Control and Status Registers. This mode provides backwards compatibility with the data transfer protocol used in the CT8005 TrueSpeech 8.5 Messaging Co-Processor and the CT8015 TrueSpeech 8.5 DSVD Co-Processor.

To perform data transfers, the host must interrogate the CT8021 using the playback/record “C2-S2-C3-S3” command/status sequence. When the CT8021 is ready to transfer data, it returns the available transfer count in the “S3” status response. The host then performs the required number of command-write-status-read cycles using the Software Control and Status registers to transfer the data.

6.2.1 Host Control/Status Register Data Transfer Synchronization Modes

The synchronization modes described here, are intended for use with speech data transfers **performed via the Software Control and Status Registers only**. The synchronization options control the manner in which data transfers are coordinated with control operations performed via these registers. When data is transferred via the transmit and receive data buffers, the CT8021 provides dedicated TX/RX ready/DMA signals.

The rate of transfer of data between the host and the CT8021 must, in general, be synchronized with the rate at which speech data is transferred between the CT8021 and its external codec(s). Data is exchanged between the CT8021 and the codec, one sample every 125 micro-seconds (at 8.0 KHz). This data is accumulated into blocks of 240 samples which corresponds to a period of 30 milliseconds. If the channel is operating in a-law/mu-law or linear mode etc., this data block (or frame) is transferred to the host without compression. If the channel is operating in TrueSpeech mode (compressed), the frame is compressed before transfer. The 30 ms period is derived from the external codec clock by counting 240 samples.

The CT8021 devices are intended to be used in pairs, at opposite ends of a communications link. The CT8021 at each end of the link, will have its own external codec and its own independent clock source or crystal. Data is moved between the two devices in real-time via the host processors attached to each CT8021 at each end of the link. Even though the CT8021 clocks will probably be derived from a crystal source, to generate the 8.0 KHz codec sample rate, the precise frequencies of these clocks will most likely be fractionally different. The frequencies may well also vary with temperature and time. As a result of this, one CT8021 at one end of the link will source/consume speech data at a fractionally different rate from the CT8021 at the other end of the link. Over time this difference will accumulate, if left alone. If the playback channel at one end, consumes data more slowly than it arrives, the transmit buffer queue will gradually fill, increasing the apparent transmission delay until the transmit buffer queue becomes full. At this point frames of data will begin to be discarded. If the playback channel consumes data more quickly than it arrives, the transmit buffer will become empty over time until the playback channel becomes starved of data. At this point gaps will be introduced into the outgoing speech, or extra frames of data will have to be synthesized to make up for the short fall in frames.

To assist with dealing with this end-to-end synchronization problem, the CT8021 playback and receive channels can be operated in one of four different sync modes. The sync mode of operation of the playback (transmit) and record (receive) channel can be selected independently. The Codec Sync mode provides for backward software compatibility with the CT8005 TrueSpeech Messaging Co-Processor.

The four sync modes are:

1. Codec Sync Mode
2. Data Sync Mode
3. Poll Sync Mode
4. Host Sync Mode

6.2.1.1 Codec Sync Mode

In Codec sync mode, Host-Codec synchronization is maintained on a 30 ms frame-by-frame basis. The CT8021 does not perform any additional internal buffering. This mode provides minimum delay with the tightest coupling between codec-CT8021 transfers and CT8021-host transfers (there is a 1 frame delay). In this mode, the Host must ensure that it is able to perform a data transfer every 30 ms. Data transfers are performed by the host processor and take place through the Software Control and Status Registers. This mode is compatible with the data transfer protocol used by the CT8005 TrueSpeech Messaging Co-Processor. In this protocol the playback/record command sequence is directly synchronized with the 30ms frame rate by delaying the Status responses until the frame is ready. This causes the CT8021 host interface to “block” or “wait” on a command to transfer data. This blocking means that this mode can only support half-duplex data transfers, i.e. concurrent playback/record is **not** supported in this mode. While the command/status interface is blocked waiting for the next frame, the CT8021 is unable to process other commands. The blocking effect also means that only 1 frame can be transferred to the CT8021 each frame period. As a result of this the effective buffer depth of the CT8021 is 1, regardless of the buffer limit setting. Software designed to support the CT8005 may depend on the blocking feature in order to use the rate at which the CT8005 responds in order to measure the progress of time. Also, this mode has the property that a data frame can always be transferred once the CT8021 has returned a status response (the transfer count returned in the S3 status response is always non-zero). Codec Sync Mode is the default mode of the CT8021 following reset.

6.2.1.2 Data Sync Mode

This mode is intended to operate in a similar manner to the Codec Sync mode, with the difference that additional buffering is provided within the CT8021. This relaxes the requirement that the host be able to respond to a transfer request from the CT8021 every 30 ms (depending on the buffer limit set). This mode provides a degree of compatibility with the CT8005 TrueSpeech Messaging Co-Processor, while also providing additional buffering.

Blocking of the command/status interface can be avoided, in this mode, by using the buffer depth polling commands prior to issuing the data transfer (play/record) commands. By checking the buffer depth, the host can avoid reading from an empty record buffer, or writing to a full playback buffer.

This mode is automatically selected when the CT8021 enters host-to-host (codec-less) compression or decompression mode. This default can be over-riden by changing the sync mode after the start of host-to-host mode.

6.2.1.3 Poll Sync Mode

In Poll Sync mode, the CT8021 maintains internal frame buffers which are tied directly to the host interface transmit/receive data buffers. Available receive data is transferred automatically to the host interface receive buffer unless the buffer is full. Any data placed in the host interface transmit buffer is automatically transferred to the internal transmit buffer unless that buffer is full. This mode is recommended for DMA transfers where there is no frame-by-frame host supervision (no “C2-S2-C3-S3” play/record command sequence is required).

Poll Sync Mode is non-blocking variant of Data Sync Mode. Regardless of the data transfer option selected, the command/status responses occur without waiting to synchronize with the 30 ms frame period. When the record buffer is empty or the playback buffer is full, the CT8021 returns zero as the transfer count in the playback/record “S3” status response. Data may be transferred via the Software Control and Status Registers, via the transmit and receive data buffers using the host processor, or via the transmit and receive data buffers using a DMA controller.

6.2.1.4 Host Sync Mode

Host Sync mode is similar to Poll Sync mode, except that the host is always given permission to perform the transfer it requests, irrespective of the internal transmit and receive state. The CT8021 accommodates this by creating or discarding data as necessary.

6.3 Data Transfer Options Summary

The following table summarizes the data transfer options:

Sync Mode	Data Transfer Options		
	Host via Software Control & Status Registers	Host via Data Buffer	DMA via Data Buffer
Codec	X		
Data	X		
Poll	X	X	X
Host	X		

Poll Sync mode should be selected when performing data transfers via the transmit and receive data buffers. In this mode, data transfer can be performed by the Host processor itself, or using an external DMA controller.

7. PLAY & RECORD DELAY MANAGEMENT

7.1 Delay and Latency Management

The CT8021 provides features that allow the host to manage and control the amount of delay and latency provided by the CT8021. The CT8021 includes data buffering in both the playback and record channels which allows the host to operate with relaxed timing requirements when transferring data to or from the device. Internally, the CT8021 operates within a strict single frame 30 ms time limit when compressing and decompressing speech frames in real-time. The CT8021's buffering capabilities allow the host to operate with programmable additional latency so as to avoid requiring the host to perform data transfers that are synchronized exactly to the 30 ms frame period. By programming the buffer depth for the record and playback channels, the host can select the trade-off made between the time allowed for the host to respond to a request for data transfer and the amount of delay introduced. Also the host is able to manage any accumulation of data, which causes additional delay, that results from differences in precise clock rate between the two ends of a speech link.

7.2 Data under-run and over-run

Data over-run occurs when data arrives at a point faster than data departs from that point. Data under-run occurs when data attempts to leave a point faster than it arrives. In addition to providing buffer depth management support functions such as buffer depth monitoring and variable buffer depth control for host based buffer management, the CT8021 has a built-in or default buffer management scheme. When the host allows the record (receive) buffer to over-run, the CT8021 responds by simply discarding frames until the host causes the record buffer to become non-full. When the host allows the playback (transmit) queue to become empty, the CT8021 responds by repeating the last frame until new data becomes available.

7.3 Buffer Monitoring

The CT8021 provides buffer monitoring functions which allow the host to determine at any point in time, the amount of speech data in the device's transmit and receive queues.

7.4 Frame creation and deletion

The CT8021 provides frame creation and deletion commands to allow the host to control dynamically the transmit and receive speech channel delays. The frame creation function causes the CT8021 to create an extra frame (provided the relevant queue is not full) by duplicating the most recent frame. The frame deletion function causes the CT8021 to discard the most recent frame from its receive or transmit buffers (assuming the buffers are not empty). These functions are an essential part of managing a full duplex communications link for two reasons.

The first is that, since the clock rates at the two ends of the link will always be fractionally different, one end of the link will tend to accumulate data, progressively increasing the end-to-end delay. The other end will consume data more quickly than it arrives. This causes any intentional delay or speech buffer created by the host, for the purpose of "smoothing" or averaging out of the data transmission rate, to be gradually consumed, until the link operates with zero delay.

The creation and deletion functions allow the host to monitor the buffer depths continually and introduce periodic corrections to the overall effective data rate to keep the buffers filled to the desired depth. For

example, if the host detects that the transmit buffer depth has become too low, it can correct this by instructing the CT8021 to create an additional frame.

The second reason is that, since the communications link connecting the two ends of the speech channel may be prone to occasional errors or data corruption, then speech frames will, from time-to-time get lost or damaged and thus discarded. This has the same effect as the data under-run process. Again, the host can use the frame create function to make up for the lost frames. If the host-to-host link includes error correction, this second issue will not be a problem. However, use of error correction schemes either entails a higher bit rate (in the case of forward-error correction) or increased latency (delay) in the case of error correction by error-detection-and-re-transmission. Both the techniques are usually undesirable on real-time speech links. The host-to-host link should include some kind of error detection mechanism (e.g. checksum or CRC) so that the host can detect corruption of speech data frames. Since the speech is very highly compressed, every bit in the speech frame is significant. If a single bit is in error, a major disturbance (noise) may be generated in the speech signal if the damaged frame is decoded. In this situation, the host should detect and discard damaged speech frames and instruct the CT8021 to create a (undamaged) frame to replace it. This process of detection, discarding and replacement of damaged frames is also known as "frame erasure".

Note that the G.723.1 (and G.729 Annex A) includes a built-in frame erasure mechanism for performing replacement of missing or corrupt speech frames. The frame erasure system (for use with the G.723.1 6.3, 5.3, TrueSpeech 4.8 and 4.1 rates, and G.729A) is independent of the mechanism described above. The G.723.1/G.729A frame erasure system performs an intelligent extrapolation of data from previous speech frames in order to replace missing frames. This feature is activated by use of specially coded speech frames.

7.5 Buffer Freezing (Pausing)

The CT8021 supports freezing or pausing of the transmit and receive buffers. Freezing of the transmit (playback) queue enables the host to pre-load frames into the playback channel before the start of playback. In a system where the host deliberately introduces delay into the channel for the purpose of smoothing an un-even data transmission rate, this feature allows the host to make use of the CT8021 internal buffer space to store the "speech reservoir" used for smoothing.

7.6 Variable Buffer Depth

The CT8021 includes 480 bytes each of buffer space for the playback (transmit) and record (receive) queues. In A-law/Mu-law or 8-bit mode, this is sufficient for two 240-byte mu-law frames (60 ms delay) or 15 32-byte TrueSpeech 8.5 frames (450 ms delay). In 16-bit mode, this space can hold only a single 480-byte frame (30ms). By default, the CT8021 will use all of this space. However, the host may wish to limit the amount of buffering the CT8021 uses in order to limit the maximum amount of delay in the speech channel. The CT8021 includes functions to set a limit on the effective buffer depths which is lower than the physical size for just this purpose.

7.7 Silence Generation During Transmit (Playback) Data Under-run

If data under-run occurs during playback, the transmit buffer becomes empty and the CT8021 will default to repeatedly outputting the most recent speech frame up to a (default) limit of 8 times. This behavior is appropriate to “mask” a short 1 or 2 frame period in which playback data is not available. Repeating the most recent frame 1 or 2 times causes the least perceptible break in the continuity of playback. However, if the data under-run condition persists for a longer period of time, the repeated playback of a single frame can cause annoyance to the user. The auto-repeat limit can be programmed by the host in the range 1-15. Programming the auto-repeat limit to zero, will cause auto-repeat to operate continuously. When the auto-repeat limit is exceeded the CT8021 will output silence until more playback data becomes available.

With the G.723.1 and G.729A coders the host may elect to trigger the frame erasure mechanism instead of using the auto-repeat feature.

7.8 Inserting Silence Frames during TrueSpeech Playback

The CT8021 will interpret TrueSpeech 8.5-4.1 data frames that have all words set to zero as silence frames. Each time the CT8021 receives an all zero TrueSpeech frame it will output 30 ms of silence. This applies to all 5 TrueSpeech Rates 8.5, 6.3, 5.3, 4.8 and 4.1. Note that this is a CT8021 specific feature. It is not a part of the standard TrueSpeech or G.723.1 implementation. The G.729A speech coder reserves the “all zero” speech frame for use in triggering frame erasure.

8. Test Modes

The CT8021 supports special modes of operation to facilitate both in-system testing of the device itself and also as an aid to external system testing.

8.1 Test Mode 1: count mode

In this test mode, data received from the external A-law /Mu-law codec by the record channel is discarded and replaced by an incrementing count. This test mode is intended for use in A-law / Mu-law speech mode where the codec receive data is passed directly to the host (without compression, and with record volume set to nominal).

8.2 Test Mode 2: digital milliwatt

In this test mode, data received from the external mu-law codec by the record channel is discarded and replaced by the CCITT G.711 A-law / Mu-law digital milliwatt (0 dBm0) code sequence. This is a sequence of 8 repeating bytes (A-law Mu-law samples) representing a 1 KHz sinewave at 0 dBm0. Selection between A-law and Mu-law is made using the codec configuration command. Digital milliwatt operation is not supported for use with a 16-bit linear codec.

8.3 Test Mode 3: internal loopback

In this test mode, data received from the external A-law / Mu-law codec by the record channel is discarded and replaced by the output data from the transmit (playback) channel. The playback channel is not affected by this mode of operation. The playback speech data is transmitted to the output codec.

Note that receive codec to transmit codec loopback can be performed by the host with the CT8021 operating in normal mode. To do this, the host simply copies frames from the record (receive) channel back to the playback (transmit) channel using A-law, Mu-law, or 16-bit linear modes as appropriate .

8.4 Codec Loopback and Monitoring

The CT8021 also supports codec loop-back and monitoring configurations.

9. CT8021 HOST CONTROL PROTOCOL

9.1 Operational Modes

The CT8021 can be operated in 8 main modes:

1. Idle Mode. In this mode, the CT8021 transmit and receive buffers are empty. Data from the receive codec is ignored, silence is output to the transmit codec.
2. Playback mode. In this mode the playback buffer is active and the TrueSpeech frame decompression function operates (if in TrueSpeech mode). The host should provide data for output in this mode. If no data is available, the most recent frame present in the playback buffer is replicated and output until new data is available. By default, speech is output via codec 1.
3. Record mode. In this mode, the record buffer is active and the TrueSpeech compression function operates (if in TrueSpeech mode), and proceeds to insert speech frames into the record buffer. The host should extract frames from the CT8021 in this mode. If the host does not take data from the CT8021, the record buffer will fill, and then frames will be discarded. By default, speech is input via codec 1.
4. Full Duplex Speech mode. This mode is equivalent to the playback and record modes operating concurrently.
5. Codec Loopback Mode. This mode provides a test mode which enables codec-input to codec-output loopback.
6. Line Monitor Mode. In Line Monitor Mode the CT8021 provides DTMF and Call Progress Tone detection. These services are also available during record and playback operations.
7. Full Duplex Speech mode plus Acoustic Echo Canceller (AEC) (DSVD Speakerphone mode). In this mode the AEC is active and attempts to remove the acoustic echo of the speaker output that is picked up at the microphone input.
8. Full Duplex (Standalone) Speakerphone mode. In this mode the CT8021 operates as a Full Duplex Speakerphone using an analog telephone line connection. This requires operation with two external codecs. The codec0 connection must be used for the telephone line audio (analog) input/output. The codec1 connection must be used for the speaker-microphone audio input/output. In Full Duplex Speakerphone mode, the CT8021 attempts to cancel the speaker-to-microphone acoustic echo and also the telephone line transmit to telephone line receive electrical echo. See Appendix A.

Note: The terms “**Standalone Speakerphone**” and “**DSVD Speakerphone**” are used throughout this document to distinguish between Speakerphone operation in a conventional analog telephone environment and Speakerphone operation in a “hands-free” DSVD or Video Conferencing system (compressed digital speech via a modem or LAN).

9.2 Basic Protocol

The Host controls the CT8021 via a command-response protocol. For each command the Host writes to the CT8021, the CT8021 will generate a status response. The Host should read the response before issuing the next command.

Commands are written to the CT8021 via the Software Control Register (SCR). The CONTROL READY bit in the Hardware Status Register (HSR) is set to indicate that the CT8021 is ready to accept a command. The Host should only write to the Software Control Register when the CONTROL READY bit is set. The CT8021 will respond to the command by writing a status response to the Software Status Register (SSR). The Host should wait for the STATUS READY bit in the Hardware Status Register to be set before reading from the Software Status Register. Reading from the Software Status Register will clear the STATUS READY bit.

Commands to the CT8021 always result in an acknowledgment via the Software Status Register. The CT8021 registers are 16-bit. The Host accesses these registers using two 8-bit cycles: low byte then high byte.

The response time for most commands is usually in the 5-10 **microseconds** range. However there are a small number of commands with a significantly longer response times up to 50-60 **milliseconds**. Notable examples of this are the commands for starting playback and record, where the command delays its response in order to synchronize with the internal speech frame period. See the detailed command descriptions for more information.

Note: actual command value constants are presented in expanded 16-bit binary form and also where appropriate in hexadecimal form with the "X" value used to indicate user selected values. For example - the command 10XXH indicates that the "base" command is hexadecimal 1000H, but that the least significant 8 bits of the 16-bit command are user selected dependent on the command options desired.

9.3 Reset & Start-up Sequence

After RESET, the CT8021 performs internal initialization operations and then sets the CONTROL READY bit in the Hardware Status Register. The host should wait for this bit to be set before issuing the first command. After initialization the CT8021 enters idle mode and awaits a command from the host. The host should program the codec configuration before attempting further operations.

Operating Start-up Sequence:

1. Power-on or Reset.
2. Host polls Hardware Status Register and waits for CONTROL READY indication.
3. Host writes first command (IDLE = 0000H) to Software Control Register.
4. CT8021 generates command response (0000H) and sets STATUS READY.
5. Host waits for STATUS READY then reads response from Software Status Register.
6. Host writes CODEC configuration command(s) to Software Control Register
7. CT8021 configures the codec (master/slave, A-law/Mu-law/16-bit and sample rate) and generates a status response.
8. Host waits for STATUS READY then reads response from Software Status Register.

Optional:

9. Host writes Get Device Identification Code = 3400H to the Software Control Register.
10. CT8021 responds with device code = 8021H and sets STATUS READY.
11. Host waits for STATUS READY then reads response from Software Status Register.

9.4 IDLE

Command : 0000H

Status: 0000H

The IDLE command instructs the CT8021 to enter IDLE mode. If the CT8021 is in either RECORD, PLAYBACK, FULL DUPLEX, or SPEAKERPHONE modes, speech processing is stopped. In IDLE mode, the CT8021 performs no processing except to wait for a command from the host.

IDLE should be the first command written to the CT8021 following reset.

9.5 CODEC CONFIGURATION

9.5.1 Codec Configuration Command

The CT8021 supports connection to two external G.711 A-law, Mu-law or 16-bit linear codecs labeled codec0 and codec1. When 2 codecs are used, both codecs must be of the same type. Codec 1 is normally required for DSVD operation (record/playback). Codec 0 is optional and is required only for standalone speakerphone mode. The data source for the record channel can be selected to be either codec, the data output for the playback channel can be selected to go to either codec or both codecs simultaneously.

After RESET, the Host MUST program the codec configuration before attempting any other operations (with the exception of Host-to-Host compression/decompression). Note that certain codecs may impose additional restrictions on the choice of the CT8021s external crystal or clock source frequency.

Command C1 (= C4XXH or C5XXH)

1100	010	Width	Law	Master	Long	OUTPUT CODEC	INPUT CODEC	Set Rate	0
(15-12)	(11-9)	(8)	(7)	(6)	(5)	(4-3)	(2)	(1)	0

Status S1

1100	010	Width	Law	Master	Long	OUTPUT CODEC	INPUT CODEC	Set Rate	0
(15-12)	(11-9)	(8)	(7)	(6)	(5)	(4-3)	(2)	(1)	0

Command C2 (only if Set Rate = 1, not required for slave mode operation)

WIDE	FSYNC_RATE	CO_RATE
(15)	(14-5)	(4-0)

Status S2 (only if Set Rate = 1)

WIDE	FSYNC_RATE	CO_RATE
(15)	(14-5)	(4-0)

Width	0 selects 8-bit A-law or Mu-law codec 1 selects 16-bit linear codec
Law	0 selects external Mu-law codec 1 selects external A-law codec (also affects Mu-law /A-law selection of RECMODE and PLAYMODE)
Master	0 selects slave mode (FSYNC and SCLK are inputs - default after reset) 1 selects master mode (FSYNC and SCLK are outputs)
Long	0 selects short frame sync mode when WIDE=0 1 selects long frame sync mode when WIDE=1
OUTPUT CODEC	00: default output codec routing (playback to codec 1) 01: output signal goes to codec 1 only 10: output signal goes to codec 0 only 11: output signal goes to both codec 0 and codec 1
INPUT CODEC	0 : normal input codec routing (record from codec 1) 1: input from codec 0 and codec 1 are exchanged (record from codec 0)
Set Rate	0: no action 1: next command word programs codec sample rate dividers
CO_RATE	This field sets the division factor used to divide the main 45.056 MHz CT8021 clock to generate SCLK. Division factor is (CO_RATE+1). For example $45.056 \text{ MHz} / (21+1) = 2.048 \text{ MHz SCLK}$. The 45.056 MHz main clock is normally generated by the x11 PLL from an external 4.096 MHz crystal or clock input. CO_RATE must be greater than 4. (master mode only)
FSYNC_RATE	This field sets the division factor used to divide the SCLK clock to generate FSYNC. Division factor is (FSYNC_RATE+1). For Example $2.048 \text{ MHz} / (255+1) = 8.0 \text{ KHz}$. F_SYNC_RATE must be greater than 17. (master mode only)
WIDE	0: FSYNC is 1 SCLK period wide 1: FSYNC is 8 SCLK periods wide if Width=0 or 16 SCLK periods wide if Width=1. Note that setting the WIDE bit to 1 only has an effect if the LONG bit is also set.

Before entering Standalone Speakerphone mode, Output Codec must be set to 00 and Input Codec set to 0.

9.5.2 Example codec configurations

C1	C2	SCLK	FSYNC	Description
C442H	1FF5H	2.048 MHz	8.000 KHz	Texas Instruments Mu-law TP3054 codec Short Frame Sync pulse Master Mode CO_RATE = 21 FSYNC_RATE = 255 external crystal = 4.096 MHz
C542H	1FF5H	2.048 MHz	8.000 KHz	AT&T 16-bit linear codec T7525 Short Frame Sync pulse Master Mode CO_RATE = 21 FSYNC_RATE = 255 external crystal = 4.096 MHz
C462H	9FF5H	2.048 MHz	8.000 KHz	OKI Semiconductor Mu-law MSM7543 codec Long (+ wide) Frame Sync pulse Master Mode CO_RATE = 21 FSYNC_RATE = 255 external crystal = 4.096 MHz

Note that the AT&T T7525 16-bit linear codec **requires** that the FSYNC frequency is exactly 1/256 of the SCLK frequency. When using this codec, the FSYNC_RATE value **must** be set to 255. In order to accommodate this requirement, when operating in Master Mode, the CT8021 **must** be driven from a crystal or clock source that is an exact multiple of 2.048 MHz (so that FSYNC operates at 8KHz). For 8-bit codecs the SCLK frequency must be at least equal to FSYNC x 16. For 16-bit codecs the SCLK frequency must be at least equal to FSYNC x 32.

Other codecs may permit a range of values for the FSYNC to SCLK ratio, however, note that for some codecs (e.g. TP3054 or TP3057) the actual usable range of ratios may depend on selection of the default or nominal ratio based on certain control pins. Depending on the operational mode selected, the TP3054 and TP3057 can support nominal ratios of 1:256, 1:193 and 1:192 using nominal 2.048 MHz, 1.544 MHz and 1.536 MHz values for SCLK (for 32 channel E1 mode or 24 channel T1 mode). Informal tests have shown that the codecs can support ratio ranges of at least +/- 5% centered about these nominals.

Note that the PLL multiplication factor used to scale the XIN frequency is 11 for the CT8021. With a 4.096 MHz crystal or clock at XIN and the PLL enabled, the CT8021 will operate at 45.056 mips.

9.6 Setting the Base Frame Size

The CT8021 supports programmable base frame sizes to support operation with different speech compression algorithms which may operate on different fundamental frame sizes and frame periods (e.g. 30ms or 10ms). The following table shows the base frame sizes recommended for different speech compression algorithms in combinations with various other CT8021 features. Note the higher processing overhead incurred when operating on smaller speech frames leaves less processing resources available for the operation of additional features like AEC. The “fundamental” frame and period sizes shown are the basic sizes of the corresponding speech compression algorithm.

Speech Compression type	Fundamental Algorithm frame size (samples)	Fundamental frame period (milliseconds)	Minimum Base size recommend for operation with the CT8021 (samples) See Note 1	Minimum Frame period recommended for operation with the CT8021	Compressed speech block size (16-bit words)
G.723.1 6.3/5.3 (all feature combinations)	240	30 ms	240	30 ms	12/10
TrueSpeech 8.5 (all feature combinations)	240	30 ms	240	30 ms	16
TrueSpeech 4.8/4.1 (all feature combinations)	240	30 ms	240	30 ms	9/8
G.728 only (note 3,4)	20	2.5 ms	40	5 ms	8 (note 2)
G.728 plus 10 ms AEC (note 3)	20	2.5 ms	80	10 ms	16 (note 2)
G.728 plus 20 ms AEC (note 3)	20	2.5 ms	240	30 ms	48 (note 2)
G.728 plus 10ms AEC with DTMF and call progress tone detectors (note 3)	20	2.5 ms	240	30 ms	48 (note 2)
G.729A+B 8 kbps	80	10 ms	80	10 ms	6 (includes 1 word frame type indicator)
G.722 7 KHz wideband 64 kbps ADPCM			80	10 ms	40

Note 1: that for correct operation of the CT8021, the base frame size selected must be an integer multiple of the fundamental frame size for the speech compression algorithm(s) used.

Note 2: The output of the G.728 algorithm for each 20 sample (2.5 ms) input block is a set for four 10-bit vectors. For an input block size of 80 samples (10 ms), four blocks are concatenated together to generate an output of 16 10-bit vectors. The 10-bit vectors are output as 10-bits right-justified within 16 bits with the most significant 6 bits of each 16-bit word set to zero..

Note 3: Transfers of compressed speech data between the host controller and the CT8021 should be made via the CT8021s Transmit/Receive Data Buffer. Use of the CT8015 style transfer protocol consumes additional CT8021 processing resources during data transfer, leaving less processing power available for compression + AEC etc.

Note 4: The CT8021 will support full duplex G.728 compression/decompression using 20 sample frames (2.5 ms). With this very small frame size, the CT8021 DSP is 100% loaded when running G.728 only. This mode is only possible when operating the device at 45.056 mips, using Transmit/Receive Data Buffer transfers, and with 16-bit wide external data sram.

G.728 speech compression algorithm, has a fundamental frame size of only 2.5 ms. It is generally not practical to implement systems which operate on such a small frame due to the overhead associated with moving data around a system in very small chunks. Another factor that affects the G.728 speech coder is that G.728 requires a relatively large amount of mips to run the coder in real-time. The CT8021 design is optimized to operate the G.728 algorithm using a frame size of 10 ms or larger. This frame size allows the CT8021 enough mips so that it is also possible to run the AEC concurrently with the G.728 coder. If the DTMF and Call Progress tone detectors are also required to operate concurrently, then the frame size should be increased to 30 ms.

G.729 Annex A operates at exactly 8.0 Kbits/sec. With a 10ms frame size this rate should produce 80 bits every 10ms. 80 bits of data requires exactly 5 16-bit words. However, with the inclusion of the G.729 Annex B VAD/CNG “silence” compression, the combined G.729AB coder can produce 3 different types of speech frame. These are: the normal speech 80 bit speech frame, a 16 bit “noisy silence” frame, and a “no transmit” indication frame. In order for the CT8021 to indicate the G729AB frame type, an extra frame type indicator word is added to the beginning of each speech block so that the total size of each compressed speech block is 6 words instead of 5.

The base frame size selected affects the operation of the entire CT8021 device. The same base frame size is used for both the record and playback channels.

The DTMF and Call Progress Tone detectors themselves are designed to operate with frame sizes of 30, 20 or 10 ms. The status from these detectors is updated and reported after every frame processed, regardless of the actual frame size.

To set the base frame size use the following command:

Command	07XXH
Status	07XXH or 0000H

Where XX selects the desired frame size (for example 07F0H selects a base frame size of 240 (decimal) for 30ms operation, 0750H selects a base frame size of 80 (decimal) for 10ms operation)

Note that the base frame size can only be changed when the CT8021 is in the IDLE state and the AEC is in the OFF state. The host must ensure that the base frame size selected matches the requirements of the speech compression algorithms that the host activates. If the command to set the base frame size is completed successfully, the status response is 07XXH, echoing the command sent. If an attempt is made to change the base frame size when the CT8021 is not in the idle state, the status response will be 0000H and the actual base frame size in the device is not changed.

For the uncompressed modes (G.711 Mu-law/A-law, and 8-bit and 16-bit linear) any frame size may be used. Changing the frames size also causes corresponding changes in the size of the data block transferred to the host:

Uncompressed data type	Frame Size Selected	Size of data transfer block (16-bit words)
16-bit linear	240 (30 ms XX=F0)	240
16-bit linear	160 (20 ms XX=A0)	160
16-bit linear	80 (10 ms XX=50)	80
8-bit linear/Mu-law/A-law	240 (30 ms XX=F0)	120
8-bit linear/Mu-law/A-law	160 (20 ms XX=A0)	80
8-bit linear/Mu-law/A-law	80 (10 ms XX=50)	40

9.7 RECORD

Command (C1 = 1XXXH):

0001	11	RECMODE	0	0	000	TFR Mode
(15-12)	(11-10)	(9-7)	(6)	(5)	(4-2)	(1-0)

Status (S1):

0001	11	RECMODE	0	0	000	TFR Mode
(15-12)	(11-10)	(9-7)	(6)	(5)	(4-2)	(1-0)

9.7.1 Transfer Mode

The TFR Mode field is used to select the mode of data transfer used:

TFR Mode	Data Transfer Mode
00	Data Transfer via Software Command and Status Registers (CT8015/CT8005 protocol mode). See section on Host Data Transfer Modes and Record Protocol. Not recommended for new designs.
01	Reserved
10	Reserved
11	Data Transfer via Host Transmit Data Buffer Access Port. Poll Sync Mode should be selected when using this data transfer option. This data transfer mode must be selected if performing DMA transfers using the TXDREQ and TXDACKN signals. See section on Host Data Transfer Modes and Record Protocol.

The following C2, C3 and Data Transfer commands are required **only** if TFR Mode = 00 (CT8015 / CT8005 protocol mode). See section on Record Protocol. These commands are used only if transferring speech data via the Software Control and Status Registers.

For new designs using TFR Mode = [binary] 11 is recommended.

Command (C2 = 1000H):

0001	0000	0000	0000
(15-12)	(11-8)	(7-4)	(3-0)

Status (S2):

0001	F3	F2	F1	F0	0	PEAK	DTMF VALID	CPF VALID	DTMF DIGIT
(15-12)	(11)	(10)	(9)	(8)	(7)	(6)	(5)	(4)	(3-0)

Command (C3 =10XXH):

0001	0000	REQUEST NUMBER OF WORDS TO TRANSFER
(15-12)	(11-8)	(7-0)

Status (S3):

0001	0000	ACTUAL NUMBER OF WORDS TO TRANSFER
(15-12)	(11-8)	(7-0)

The NUMBER OF WORDS TO TRANSFER should either be set to the (compressed) frame size of the selected speech mode, or set to zero to indicate no transfer is to take place. The ACTUAL NUMBER OF WORDS TO TRANSFER will be the compressed frame size if data is available for transfer, or it will be zero to indicate that no data is available.

Data Transfer Command:

0001	0000 0000 0000
(15-12)	(11-0)

Data Transfer Status:

16-bit speech data word
(15-0)

9.7.2 Recording Modes

RECMODE	Frame Size (words)	Recording Modes
000	48-4	Compressed Speech: TrueSpeech (8.5, G.723.1 6.3/5.3, 4.8 and 4.1), G.728 etc. See : Select TrueSpeech Record Rate command
001		Reserved
010		Reserved
011		Reserved
100	120	64 Kbit/sec A-law / Mu-law PCM (120 words per 30ms data frame)
101	240	128 Kbit/sec 16-bit linear (240 words per 30ms data frame)
110	120	64 Kbits/sec 8-bit signed linear (120 words per 30 ms data frame)
111	120	64 Kbits/sec WSS 8-bit unsigned linear (120 words per 30 ms data frame)

Notes:

1. All data rates assume that the codec connected to the CT8021 operates at 8000 samples/sec
2. The 30 ms frame interval is based on counting 240 samples from the codec. See also the section on setting the base frame size
3. Mode 100, PCM A-law / Mu-law is a “pass through mode”, where the data from the codec is passed to the host entirely unmodified, (provided that the record volume is set to 0100H).
4. 16 bit linear data is signed integer (2’s complement form), positive full scale is 7FFFH, negative full scale is 8000H, zero is 0000H
5. 8 bit linear data is signed byte (2’s complement form), positive full scale is 7FH, negative full scale is 80H, zero is 00H

6. 8 bit linear WSS data is unsigned byte, positive full scale is FFH, negative full scale is 00H, zero is 80H. This is the format used for 8-bit WAVE format data (Microsoft Windows Sound System). Data can be converted between the two 8-bit signed/unsigned formats by simply inverting the ms bit.
7. The compressed speech rate for TrueSpeech 8.5,6.3/5.3 (G.723.1),4.8 or 4.1 kbps and G.728 etc. is selected using the Select TrueSpeech Rate command
8. Selection between A-law / Mu-law is made using the codec configuration command. With an external Mu-law codec configured, RECMODE = 100 selects Mu-law data. With an external A-law codec configured, RECMODE = 100 selects A-law data.
9. The RECMODE and PLAYMODE bit assignments are different. See Playback command

9.7.3 Peak Level Indication

The PEAK bit in the S2 status response is set if the average input level for the current speech frame exceeds the threshold configured using the Set Peak Threshold command. This bit can be used by the host to perform silence detection during recording. Note that the Record signal level can also be accessed using the Read Record Level command. This is the recommended method of determining the record input signal level, when TFR Mode = [binary] 11 and the C2-S2, C3-S3 sequence is not used.

CPF F3-0 and DTMF Detection: refer to section on Line Monitoring

9.7.4 Record Protocol

Example 1 : TrueSpeech 6.3 Record via Host Receive Data Buffer Access Port - Host Polling

This example shows how to start recording using TrueSpeech 6.3 with data transfers performed via the Host Receive Data Buffer Access Port. All data transfers in this example are performed by the host controller. The Host polls the Hardware Status Register to perform data synchronization.

Starting Record

1. CT8021 is in IDLE or PLAYBACK state.
2. Host checks for CONTROL READY state in Hardware Status Register.
3. Host selects TrueSpeech 6.3 recording by writing the Select TrueSpeech Record Rate command = 5131H to the Software Control Register.
4. CT8021 responds via the Software Status Register
5. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
6. Host selects Poll Sync Mode for record using the command 5102H.
7. CT8021 responds via the Software Status Register.
8. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
9. Host selects Record by issuing the RECORD “C1” command = 1C03H via the Software Control Register. This also informs the CT8021 that data transfers will occur via the Host Receive Data Buffer Access Port (TFR Mode = [binary] 11).
10. CT8021 performs internal synchronization then responds with the RECORD “S1” status response after a delay of up to 2 speech frame periods.
11. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.

The CT8021 is now in record mode. The CT8021 will produce a compressed speech frame every 30 ms (assuming the base frame size is set to 240). The following sequence is used to transfer each compressed speech data frame from the CT8021 to the host:

1. Host waits for the RX Ready bit to be set in the Hardware Status Register
2. Host reads 12 words (for TrueSpeech 6.3) from the Host Receive Data Buffer Access Port.

Note that for transfer of speech blocks larger than 16 words via the Host Receive Data Buffer Access Port, the transfer will be split into sub-blocks of 16 words each or less. The CT8021 will create a small pause between each sub-block during which it will re-fill the data access port (approx 5 microseconds).

To terminate recording:

1. Host writes IDLE = 0000H command or STOP RECORD = 5120H command to the Software Control Register.
2. CT8021 terminates record and clears RX Ready if set (in Hardware Status Register).
3. CT8021 writes status response to Software Status Register.
4. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.

Note that after writing the IDLE or STOP RECORD command to terminate recording, the host should not attempt to access the Host Receive Data Buffer Access Port even if the RX Ready bit is set.

Example 2 : TrueSpeech 5.3 Record via Host Receive Data Buffer Access Port - Host Interrupt

This example shows how to start recording using TrueSpeech 5.3 with data transfers performed via the Host Receive Data Buffer Access Port. All data transfers in this example are performed by the host controller. The Host uses an interrupt service routine (ISR) to perform data transfer.

Starting Record

1. CT8021 is in IDLE state or PLAYBACK.
2. Host checks for CONTROL READY state in Hardware Status Register.
3. Host selects TrueSpeech 5.3 recording by writing the Select TrueSpeech Record Rate command = 5132H to the Software Control Register.
4. CT8021 responds via the Software Status Register
5. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
6. Host selects Poll Sync Mode for record using the command 5102H.
7. CT8021 responds via the Software Status Register.
8. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
9. Host selects Record by issuing the RECORD "C1" command = 1C03H via the Software Control Register. This also informs the CT8021 that data transfers will occur via the Host Receive Data Buffer Access Port (TFR Mode = [binary] 11).
10. CT8021 performs internal synchronization then responds with the RECORD "S1" status response after a delay of up to 2 speech frame periods.
11. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
12. Host sets up an ISR to service interrupts from the CT8021.
13. Host programs the Hardware Control Register to generate an interrupt from the CT8021 on the RX Ready condition by writing 0401H (Master Enable set, RX Ready IE set).

The CT8021 is now in record mode. The CT8021 will produce a compressed speech frame every 30 ms (assuming the base frame size is set to 240). The following sequence is used by the Host ISR to transfer each compressed speech data frame from the CT8021 to the host:

1. Host gets interrupt from CT8021 (optionally checks that RX Ready is set).
2. Host reads 10 words (for TrueSpeech 5.3) from the Host Receive Data Buffer Access Port.

To terminate recording:

1. Host disables interrupts from the CT8021 by writing 0000H to the Hardware Control Register.
2. Host writes IDLE = 0000H command or STOP RECORD = 5120H command to the Software Control Register.
3. CT8021 terminates record and clears RX Ready if set (in Hardware Status Register).
4. CT8021 writes status response to Software Status Register.
5. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.

Note that after writing the IDLE or STOP RECORD command to terminate recording, the host should not attempt to access the Host Receive Data Buffer Access Port even if the RX Ready bit is set.

Example 3 : Linear 16-bit Record via Host Receive Data Buffer Access Port - Using DMA

This example shows how to start recording in uncompressed 16-bit Linear format speech with data transfers performed via the Host Receive Data Buffer Access Port. All data transfers in this example are performed by an external DMA controller.

Starting Record

1. CT8021 is in IDLE state or PLAYBACK state.
2. Host checks for CONTROL READY state in Hardware Status Register.
3. Host selects Poll Sync Mode for record using the command 5102H.
4. CT8021 responds via the Software Status Register.
5. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
6. Host selects Record by issuing the RECORD "C1" command = 1E83H (RECMODE = [binary] 101) via the Software Control Register. This also informs the CT8021 that data transfers will occur via the Host Receive Data Buffer Access Port (TFR Mode = [binary] 11).
7. CT8021 performs internal synchronization then responds with the RECORD "S1" status response after a delay of up to 2 speech frame periods.
8. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
9. Host programs external DMA controller to accept DMA requests from the CT8021 (See DMA Direction bit in Hardware Control Register). Host configures a data buffer in the Host memory to receive the DMA-ed speech data. The DMA memory buffer should be an integral multiple of the speech frame size. In the case of uncompressed 16-bit Linear, this should be N*480 (bytes).
10. Host sets up an ISR to handle an end-of-transfer (EOT) interrupt from the DMA controller when it reaches the end of the DMA buffer. Alternatively, if the DMA supports circular-buffering (auto-initialize), the Host can poll the DMA to determine when the end of the buffer is reached.
11. Host programs the Hardware Control Register to generate a DMA request from the CT8021 on the RX Ready condition by writing 0050H (RX DMA Burst Mode, RX DMA enable) - or 0150H (RX DMA Burst Mode, RX DMA enable, DMA Direction bit set).

The CT8021 is now in record mode. The CT8021 will produce a uncompressed speech frame every 30 ms (assuming that the base frame size is set to 240). The following sequence occurs between the CT8021 and the DMA controller to transfer data:

1. CT8021 has 240 word (480 byte) uncompressed 16-bit speech data frame ready for transfer.
2. CT8021 transfers first 16 words into the Host Receive Data Buffer.
3. CT8021 asserts RX Ready.
4. RX Ready drives RXDREQ.
5. DMA controller responds to RXDREQ by asserting RXDACKN.
6. DMA controller performs a burst transfer of 16 words (32 bytes) from the CT8021 to the Host DMA memory buffer.
7. CT8021 de-asserts RX Ready.
8. CT8021 transfers next 16 words into the Host Receive Data Buffer.
9. CT8021 asserts RX Ready.
10. RX Ready drives RXDREQ.
11. DMA controller responds to RXDREQ by asserting RXDACKN.
12. DMA controller performs a burst transfer of 16 words (32 bytes) from the CT8021 to the Host DMA memory buffer.
13. CT8021 de-asserts RX Ready.
14. Repeat 8-13 until all 240 words of the speech data frame have been transferred.

The Host processor also needs to supervise the operation of the DMA controller and arrange for the transfer of the DMA-ed data to its final destination (e.g. transfer to disk). When the DMA controller reaches the end of the DMA buffer, the Host must process the data (transfer to disk). The Host can either arrange to receive an EOT interrupt from the DMA controller, or can periodically poll the DMA's internal transfer count register to determine the status of the DMA.

To terminate recording:

1. Host disables DMA requests from the CT8021 by writing 0000H to the Hardware Control Register.
2. Host writes IDLE = 0000H command or STOP RECORD = 5120H command to the Software Control Register.
3. CT8021 terminates record and clears RX Ready if set (in Hardware Status Register).
4. CT8021 writes status response to Software Status Register.
5. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.

Note that after writing the IDLE or STOP RECORD command to terminate recording, the host/DMA should not attempt to access the Host Receive Data Buffer Access Port even if the RX Ready bit is set. When "cleaning up" at the end of recording, the Host should ensure that any partial frame at the end of the recording is discarded. When the recorded data is sent back to the CT8021 for playback, the CT8021 can only playback complete frames. It is recommended that the Host synchronizes the termination of recording with the end of transfer of a complete speech frame. This synchronization can be achieved by examining the DMA's transfer count register, or by performing the final speech data frame transfer using the Host processor.

Example 4: TrueSpeech 8.5 Recording using CT8005/CT8015 protocol via Software Control and Status Registers (Requires download of external TrueSpeech 8.5 coder module).

This example shows how to perform TrueSpeech 8.5 recording with TFR Mode = 00 using the CT8015 compatible protocol. Data is transferred via the Software Control and Status Registers. All data transfers are performed by the Host processor. The example shown uses the CT8015 Poll Sync Mode.

1. CT8021 is in IDLE or PLAYBACK state.
2. Host checks for CONTROL READY state in Hardware Status Register.
3. Host selects TrueSpeech 8.5 recording by writing the Select TrueSpeech Record Rate command = 5130H to the Software Control Register.
4. CT8021 responds via the Software Status Register
5. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
6. Host selects Poll Sync Mode for record using the command 5102H.
7. CT8021 responds via the Software Status Register.
8. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
9. Host writes Record C1 command = 1C00H (TrueSpeech 8.5) to the Software Control Register.
10. CT8021 activates record mode and performs internal synchronizations (1-2 frame delay) .
11. CT8021 responds with Record status S1 = 1C00H.
12. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
- 13. Host sends Record C2 command = 1000H.**
- 14. CT8021 responds with Record S2 status = 10X0H, where X is defined by the PEAK bit state.**
- 15. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.**
- 16. Host sends Record C3 with the number of (16 bit) words it wishes to transfer = 1010H.**

17. **CT8021 checks number of words ready to transfer.**
18. **CT8021 responds with Record S3 = 1000H or 1010H. If number of words is not zero DATAFLAG/ signal is asserted to indicate the beginning of data transfer.**
19. **Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.**
20. **If number of words to transfer is zero (S3 = 1000H), Host goes back to step 13.**
21. **Optional: Host waits for DATAFLAG/ signal to be asserted**
22. **Host writes data transfer command = 1000H to Software Control Register**
23. **CT8021 responds with 16-bit data word in Software Status Register**
24. **Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.**
25. **Repeat steps 22-24 until all (16) data words are transferred**
26. **CT8021 de-asserts DATAFLAG/ signal to indicate the end of data transfer**
27. **Optional: Host waits for DATAFLAG/ to be de-asserted**
28. **Repeat steps 13-27 until recording finished**
29. To end recording host writes IDLE command or Record Stop command and waits for status response.

Note: **bold text** indicates the record software data transfer loop. By default, The DATAFLAG signal on external pin GPIO0 is disabled. See section on GPIO pins for more information.

9.7.5 Select TrueSpeech/G.723.1/G.728 Record Rate

This command allows the host to select the TrueSpeech data rate used in compressed record mode. The standard CT8021 (built-in) available rates are: TrueSpeech 6.3, 5.3, 4.8 and 4.1 Kbps and G.728 16 Kbps only. This command should be issued prior to starting record using the RECORD command. The TrueSpeech Record Rate remains in effect until the next TrueSpeech Record Rate command.

Command (513XH):

0101	0001	0011	RATE
(15-12)	(11-8)	(7-4)	(3-0)

Status (513XH):

0101	0001	0011	RATE
(15-12)	(11-8)	(7-4)	(3-0)

RATE	TrueSpeech Compression Rate	Compressed Frame Size (16-bit words)
0	8.5 (see note 1)	16
1	6.3 (G.723.1) (default)	12
2	5.3 (G.723.1)	10
3	4.8	9
4	4.1	8
5	G.728	4,8,16,32 or 48 (2.5,5,10,20 or 30 ms)
6	G.729AB (see note 1)	6 (10 ms)
7	G.722 (see note 1)	40 (10 ms)
8-15	reserved	

The Frame Size value indicates the number of 16-bit words in a compressed speech frame for each coding rate. When any of the TrueSpeech compression rates are selected, the frame size is always 16 words or less. Since the size of the CT8021 hardware buffer within the Host port is 16 words, the Host can “burst” read the complete speech frame of 16-8 words without needing to check the RX Ready bit before each individual word is read. Where the frame size is larger than 16 words, the CT8021 splits the data transfer into sub-blocks of 16 words or less.

Notes:

- The CT8021 supports the TrueSpeech 8.5, G.729-A/B and G.722 coders only as external downloadable software expansion modules. The host must ensure that the appropriate module is loaded to the CT8021 external program/data memory before attempting to activate any of these coders. Contact DSP Group for details on availability for these coders.**

9.7.6 Enable VAD

The G.723.1 / TrueSpeech 6.3, 5.3, 4.8 and 4.1 rates support VAD (Voice Activity Detection) and CNG (comfort noise generation) as defined in the ITU-T G.723.1 Annex A specification. This provides a form of built-in “silence compression”. On the speech encode (record) side the VAD feature can be selectively enabled or disabled. On the decode (playback) side, “comfort noise” generation (CNG) is automatic and is triggered by specially formatted speech frames that are generated by the encoder when VAD is enabled. By default, the encoder VAD is disabled.

Note that the size of the physical data transfer performed for each speech frame when reading or writing to the CT8021 is determined by the initial speech mode selection. For example if the record session is started in 6.3 mode, then all data transfers will consist of 12 word frames. The VAD/CNG frames which contain a smaller amount of significant data (2 words or 2 bits) have the unused portions of the frame padded with zeros.

To enable VAD:

Command	513FH
Status	513FH

To disable VAD:

Command	513EH
Status	513EH

9.7.7 Dynamic Switching between TrueSpeech/G.723.1 6.3 and 5.3

The CT8021 supports on-the-fly switching between the TrueSpeech 6.3 and 5.3 data rates as required by the ITU G.723.1 specification. To enable switching between these two rates, the RECORD operation **must** be started with the 6.3 rate selected. This sets the frames size used for CT8021-to-Host transfer to 12 words. When dynamically switching between the 6.3 and 5.3 rates the frames size remains fixed throughout the record process. The 12 word frame size It is used for both the 6.3 **and** 5.3 speech data frames, even though the 5.3 speech frame only contains 10 words. This prevents synchronization problems between the time at which the compression rate is changed and the time at which the compressed data is read out from the CT8021. When a 5.3 frame is read from the CT8021s Hardware Receive Buffers after on-the-fly switching from the 6.3 rate, the Host must read 12 words instead of 10. In this case the last two words of data may be discarded.

Switching between the 6.3 and 5.3 rates is performed using the Select TrueSpeech Record Rate command (above). To support on-the-fly switching, the CT8021 will accept this command after the record process has been started. If the CT8021 is instructed to make an unsupported rate switch (for example between TrueSpeech 6.3 and TrueSpeech 8.5), the status response from the Select TrueSpeech Record Rate will be zero, and the CT8021 record operation will continue to operate with the old TrueSpeech rate.

The l.s two bits of the first data word of a G.723.1 6.3 or 5.3 speech frame are encoded to indicate the speech frame type.

During Playback (de-compression) the CT8021 will test the two l.s. significant control bits only when the current host selected TrueSpeech rate is 6.3 or 5.3. In this case, the CT8021 will automatically perform the correct decoding of the speech frames independently of which rate is selected by the host . To support on-the-fly switching for Playback, Playback must be started with the 6.3 rate selected.

This fixes the frame size used for Host-to-CT8021 transfer at 12 words. When transferring a 5.3 speech frame into the CT8021 when the 6.3 rate was initially selected, the Host should append two dummy words (zeros) at the end of the 10 word TrueSpeech 5.3 frame.

To switch between other speech rates, the Host must stop and re-start playback/record. The Host is responsible for synchronizing the rate switch between playback and record sources.

9.7.8 G.723.1 Frame Type Encoding

The least significant two bits of the first word of the TrueSpeech 6.3 and 5.3 frames are defined by the ITU-T G.723.1 v5.1 (final formal release) specification to indicate the speech data rate as shown:

Bit 1-0	Frame Type	Data Rate	Number of significant data words
00	0	TrueSpeech 6.3 data frame (12 words)	12
01	1	TrueSpeech 5.3 data frame (10 words)	10
10	2	Silence / Comfort Noise Generation frame (only first 2 words contain useful data). All other data words are set to zero	2
11	3	Repeat last CNG frame (only 1.s. 2 bits of first data word are used). All other data words are set to zero	2 bits

Compatibility Note: the Bit 1-0 encoding of the CT8021 is different from that used in older firmware versions of the CT8020 which implemented pre-release versions of G.723.1, (firmware revisions 0109 and 0112 part numbers CT8020A11AQC and CT8020A11BQC). The full production release, CT8020D11AQC (firmware revision 0114) implements the final v5.1 G.723.1 release as do all versions of the CT8021.

The Host can examine these two bits to distinguish between 6.3 and 5.3 frames (only) to determine the data rate and the number of valid data words in the frame. This bit encoding applies **only** to the 6.3 and 5.3 TrueSpeech frames. There is no bit encoding provided for the 8.5, 4.8 and 4.1 rates.

When implementing a real-time digital compressed speech link, the host and data transport service should be able to transfer all four sub-types of speech frame end-to-end. The host may take advantage of the smaller effective frame size of the silence frames by using the unused portion of the frame for transporting non-speech data. However, before writing the silence frames back to the decoder, the host should restore the unused portions of the silence frames to zero.

The CT8021 implements additional non-standard frame encodings as follows:

1. A speech frame which consists of all zeros will be interpreted as an instruction to the playback channel to output pure silence for one frame period. This applies to all TrueSpeech rates, including 8.5, 6.3, 5.3, 4.8 and 4.1.
2. A speech frame which has the first two data words set to FFFFH, with the remaining words set to all zeros, will be interpreted as an instruction to activate the G.723.1 frame erasure mechanism. This should be used where the host needs to instruct the CT8021 playback channel to replace a dropped or corrupt speech frame by extrapolating speech data from previous speech frames. This feature is available only with the TrueSpeech 6.3, 5.3, 4.8 and 4.1 data rates.

When reading or writing speech frames, to or from the CT8021, the number of physical words transferred for each individual frame is fixed during each record/playback session. For example, if the record/playback session is started in 6.3 mode, then all frames will contain 12 words.

Frames which use less than this number of words for actual data will have the unused portion of the frame set to zero. This approach simplifies the coding of low level driver software such that it does not need to deal with variable sized data frames. This is particularly significant when dealing with DMA type transfer.

9.7.9 Programming the Record Peak Threshold

The PEAK bit in the Record mode status S2 response can be used by the host to implement silence detection during record. The PEAK bit is set whenever the average signal level for the most recent record frame is greater than the Input Peak Threshold. Although the CT8021 includes a record data frame buffer which can hold a number of record data frames, the PEAK indication provided always refers to the most recent frame recorded. In most cases, where the host services the CT8021 record data frame buffer every frame period (30 ms), there should never be more than one data frame in the buffer. The signal level measured is the input signal level before any scaling occurs due to the Record Volume Control or AGC.

Command C1 (CF99H):

1100 1111 1001 1	001
(15-3)	(2-0)

Status S1:

1100 1111 1001 1	001
(15-3)	(2-0)

Command C2:

Threshold Value
(15-0)

Status S2:

Threshold Value
(15-0)

By default, the Threshold Value is set to 0100H.

9.7.10 Reading the Record Level Value and Threshold Value

The Threshold Value programmed, together with the actual current frame average Level Value can be read using the command:

Command (CF88H or CF89H):

1100 1111 1000 1	00X
(15-3)	(2-0)

Status:

Threshold or Level Value
(15-0)

Where X = 0 selects reading the Level Value.
 X = 1 selects reading the Threshold Value.

The level value is calculated using the following algorithm:

LEVEL = 0

FOR each of 240 INPUT samples per frame
 IF INPUT < 0 THEN INPUT = -1 * INPUT
 LEVEL = LEVEL + INPUT

LEVEL = LEVEL / 256

Note that this is an average signal level (not r.m.s.). The division by 256 instead of 240 leads to a slight understatement of the true average level. The fractional error is $240/256 = 0.9375$, or 6.25%. Dividing by 256 instead of 240 requires less processing by the CT8021 device.

When the CT8021 is operated with a base frame size that is not set to 240, the level calculation is adjusted to report the same level as if the base frame size was set to 240. However since the CT8021 will update and report the level for each frame processed, with a smaller base frame size the CT8021 will report the level measured over a shorter period. This may result in more frame-by-frame variation in the signal level measured. When measuring the level of a signal with lower frequency components, or with a low frequency modulation, more “ripple” may be observed in the reported signal level. One example of this would be reporting the signal level of a 350+440 Hz dial tone.

Threshold Level Scale

The threshold level scale is relative to the maximum positive full scale value of 7FFFH (decimal 32767). This corresponds to an input signal level of approximately +3dBm0 from the external mu-law codec.

The precise input signal level scale can be calculated thus:

$$\text{Input Level in dBm0} = +3.17 - 20 \log(\text{base } 10) \left(\frac{\text{input level}}{32767} * 256/240 \right)$$

so the default threshold level of 0100H = 256 decimal, corresponds to -38.4 dBm0.

To calculate the actual average input signal level, read the LEVEL from the device and then perform the following calculation:

$$\text{CORRECTED LEVEL} = (\text{LEVEL} * 256) / 240$$

$$\text{input level in dBm0} = 3.17 - 20 * \log_{10}(\text{CORRECTED LEVEL} / 32767)$$

9.8 PLAYBACK

Command (C1 = 2CXXH):

0010	11	000	PLAYMODE	00	TFR Mode
(15-12)	(11-10)	(9-7)	(6-4)	(3-2)	(1-0)

Status (S1):

0010	11	000	PLAYMODE	00	TFR Mode
(15-12)	(11-10)	(9-7)	(6-4)	(3-2)	(1-0)

9.8.1 Transfer Mode

The TFR Mode field is used to select the mode of data transfer used:

TFR Mode	Data Transfer Mode
00	Data Transfer via Software Command and Status Registers (CT8015/CT8005 protocol mode). See section on Host Data Transfer Modes and Record Protocol.
01	Equivalent to TFR Mode 11, with the addition that data frame alignment is strictly maintained. See section called Playback Frame Alignment Using Transmit Data Buffer for more details. Not available in some older versions of the CT8020.
10	Reserved
11	Data Transfer via Host Transmit Data Buffer Access Port. Poll Sync Mode should be selected when using this data transfer option. This data transfer mode must be selected if performing DMA transfers using the TXDREQ and TXDACKN signals. See section on Host Data Transfer Modes and Record Protocol.

For new designs using TFR Mode = [binary] 11 or 01 is recommended.

The **C2-S2**, **C3-S3** commands detailed below are used **once** at the beginning of playback (or host-to-host decompression) if TFR Mode = [binary] 11 or 01. If TFR Mode = 00, the C2-S2, C3-S3 commands are used to transfer playback speech data via the Software Control and Status Register.

The Data Transfer command is required only if TFR Mode = 00 (CT8015 / CT8005 protocol mode). This command is used only if transferring speech data via the Software Control and Status Registers.

See section on Playback Protocol.

Command (C2 = 2000H):

0010	0000	0000	0000
(15-12)	(11-8)	(7-4)	(3-0)

Status (S2):

0001	F3	F2	F1	F0	0	0	DTMF VALID	CPF VALID	DTMF DIGIT
(15-12)	(11)	(10)	(9)	(8)	(7)	(6)	(5)	(4)	(3-0)

For information on CPF F3-0 and DTMF detection please refer to section on Line Monitoring.

Command (C3= 20XXH):

0010	0000	REQUEST NUMBER OF WORDS TO TRANSFER
(15-12)	(11-8)	(7-0)

Status (S3):

0010	0	0	00	ACTUAL NUMBER OF WORDS TO TRANSFER
(15-12)	(11)	(10)	(9-8)	(7-0)

The NUMBER OF WORDS TO TRANSFER should either be set to the (compressed) frame size of the selected speech mode, or set to zero to indicate no transfer is to take place. The ACTUAL NUMBER OF WORDS TO TRANSFER will be the compressed frame size if the device is ready to accept data for transfer, or it will be zero to indicate that the device is not ready for data transfer.

Data Transfer Command:

16-bit speech data word
(15-0)

Data Transfer Status:

16-bit speech data word
(15-0)

9.8.2 Playback Modes

PLAYMODE	Frame Size (words)	Playback Modes
000	16-8	Compressed Speech: TrueSpeech (8.5, G.723.1 6.3/5.3, 4.8 and 4.1), G.728 etc. See : Select TrueSpeech Playback Rate command
001		Reserved
010	120	64 Kbit/sec A-law / Mu-law PCM (120 words per 30ms data frame)
011		Reserved
100	120	64 Kbits/sec 8-bit signed linear (120 words per 30 ms data frame)
101	120	64 Kbits/sec 8-bit unsigned linear (120 words per 30 ms data frame)
110	240	128 Kbits/sec 16-bit signed linear (240 words per 30 ms data frame)
111		Reserved

Notes:

1. All data rates assume that the codec connected to the CT8021 operates at 8000 samples/sec
2. The 30 ms frame interval is based on counting 240 samples from the codec (see also command to set base frame size).
3. Mode 010, PCM mu-law is a “pass through mode”, where the data from the host is passed to the codec entirely unmodified, (provided that the playback volume is set to 0100H).
4. 16 bit linear data is signed integer (2’s complement form), positive full scale is 7FFFH, negative full scale is 8000H, zero is 0000H
5. 8 bit linear data is signed byte (2’s complement form), positive full scale is 7FH, negative full scale is 80H, zero is 00H
6. 8 bit linear WSS data is unsigned byte, positive full scale is FFH, negative full scale is 00H, zero is 80H. This is the format used for 8-bit WAVE format data (Windows Sound System). Data can be converted between the two 8-bit formats by simply inverting the ms bit.
7. The TrueSpeech rate 8.5,6.3,5.3,4.8 or 4.1 or G.728 etc. is selected using the Select TrueSpeech Playback Rate command.
8. Selection between A-law / Mu-law is made using the codec configuration command. With an external Mu-law codec configured, RECMODE = 100 selects Mu-law data. With an external A-law codec configured, RECMODE = 100 selects A-law data.

9.8.3 Playback Protocol

Example 1 : TrueSpeech 6.3 Playback via Host Transmit Data Buffer Access Port - Host Polling

This example shows how to start playback using TrueSpeech 6.3 with data transfers performed via the Host Transmit Data Buffer Access Port. All data transfers in this example are performed by the Host controller. The Host polls the Hardware Status Register to perform data synchronization.

Starting Playback

1. CT8021 is in IDLE or RECORD state.
2. Host checks for CONTROL READY state in Hardware Status Register.
3. Host selects TrueSpeech 6.3 playback by writing the Select TrueSpeech Playback Rate command = 5231H to the Software Control Register.
4. CT8021 responds via the Software Status Register.
5. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
6. Host selects Poll Sync Mode for playback using the command 5202H.
7. CT8021 responds via the Software Status Register.
8. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
9. Host selects Playback by issuing the PLAYBACK “C1” command = 2C03H or 2C01H via the Software Control Register. This also informs the CT8021 that data transfers will occur via the Host Transmit Data Buffer Access Port (TFR Mode = [binary] 11 or 01).
10. CT8021 responds via the Software Status Register after a delay of up to 2 speech frame periods.
11. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register S1 = 2C03H or 2C01H. This clears the STATUS READY bit.
12. Host writes PLAYBACK “C2” command = 2000H to the Software Control Register.
13. CT8021 responds via the Software Status Register.
14. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register S2 = 2000H. This clears the STATUS READY bit.
15. Host writes PLAYBACK “C3” command = 200CH to the Software Control Register. This command includes the requested number of words per frame to transfer. In this case (TrueSpeech 6.3) this is 12 = 0CH.
16. CT8021 responds via the Software Status Register.
17. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register S3 = 200CH. This clears the STATUS READY bit.

The CT8021 is now in playback mode. The CT8021 will be ready to consume a compressed speech frame every 30 ms (assuming the base frame size is set to 240). The following sequence is used to transfer each compressed speech frame from the Host to the CT8021:

1. Host waits for TX Ready bit to be set in the Hardware Status Register
2. Host writes 12 words (for TrueSpeech 6.3) to the Host Transmit Data Buffer Access Port.

Note that for transfer of speech blocks larger than 16 words via the Host Transmit Data Buffer Access Port, the transfer will be split into sub-blocks of 16 words each or less. The CT8021 will create a small pause between each sub-block during which it will move the sub-block from data access port (approx 5 microseconds).

The CT8021 includes can buffer up to 20 TrueSpeech 6.3 speech data frames (600 ms). At the start of playback, the CT8021 will keep re-asserting the TX Ready condition until the internal 480 byte buffer is full, and the CT8021 has “pre-fetched” 20 frames. If the host needs to reduce the number of speech data frames “pre-fetched” by the CT8021, it can do so using the playback Set Buffer Depth command (528XH). This can be set so as to reduce the effective size of the internal buffer. See also TFR_MODE 01 and the section on playback data alignment.

To terminate playback:

1. Host writes IDLE = 0000H command or STOP PLAYBACK = 5220H command to the Software Control Register.
2. CT8021 terminates playback, discards data in its internal playback buffer, and clears TX Ready if set (in the Hardware Status Register).
3. CT8021 writes status response to the Software Status Register.
4. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.

When the IDLE or STOP PLAYBACK command is used to terminate playback, the CT8021 will discard data in its internal playback buffer. In some cases, this may not be the desired behavior. For example, when playing back TrueSpeech 6.3 data stored in a disk file, if the IDLE or STOP PLAYBACK command is issued immediately on encountering end-of-file (EOF), then the last 20 frames (600 ms) of the message will not be played. The Host may wish to make use of the playback Buffer Monitoring command (5212H). Using this command, the Host can poll the CT8021 and wait until all the internally stored speech data has been played out before issuing the IDLE or STOP PLAYBACK command.

Note that after writing the IDLE or STOP PLAYBACK command to terminate playback, the host should not attempt to access the Host Transmit Data Buffer Access Port even if the TX Ready bit is set.

Example 2 : TrueSpeech 5.3 Playback via Host Receive Data Buffer Access Port - Host Interrupt

This example shows how to start playback using TrueSpeech 5.3 with data transfers performed via the Host Transmit Data Buffer Access Port. All data transfers in this example are performed by the host controller. The Host uses an interrupt service routine (ISR) to perform data transfer.

Starting Playback

1. CT8021 is in IDLE state or RECORD.
2. Host checks for CONTROL READY state in Hardware Status Register.
3. Host selects TrueSpeech 5.3 playback by writing the Select TrueSpeech Playback Rate command = 5232H to the Software Control Register.
4. CT8021 responds via the Software Status Register
5. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
6. Host selects Poll Sync Mode for playback using the command 5202H.
7. CT8021 responds via the Software Status Register.
8. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
9. Host selects Playback by issuing the PLAYBACK “C1” command = 2C03H or 2C01H via the Software Control Register. This also informs the CT8021 that data transfers will occur via the Host Transmit Data Buffer Access Port (TFR Mode = [binary] 11 or 01).
10. CT8021 responds via the Software Status Register after a delay of up to 2 speech frame periods.
11. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register S1 = 2C03H or 2C01H. This clears the STATUS READY bit.

12. Host writes PLAYBACK “C2” command = 2000H to the Software Control Register.
13. CT8021 responds via the Software Status Register.
14. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register S2 = 2000H. This clears the STATUS READY bit.
15. Host writes PLAYBACK “C3” command = 200AH to the Software Control Register. This command includes the requested number of words per frame to transfer. In this case (TrueSpeech 5.3) this is 10 = 0AH.
16. CT8021 responds via the Software Status Register.
17. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register S3 = 200AH. This clears the STATUS READY bit.
18. Host sets up an ISR to service interrupts from the CT8021.
19. Host programs the Hardware Control Register to generate an interrupt from the CT8021 on the TX Ready condition by writing 0402H (Master Enable set, TX Ready IE set).

The CT8021 is now in playback mode. The CT8021 will consume a compressed speech frame every 30 ms. The following sequence is used by the Host ISR to transfer each compressed speech data frame from the Host to the CT8021:

1. Host gets interrupt from CT8021 (optionally checks that TX Ready is set).
2. Host writes 10 words (for TrueSpeech 5.3) into the Host Transmit Data Buffer Access Port.

The CT8021 can buffer upto 20 TrueSpeech 5.3 speech data frames (600 ms). At the start of playback, the CT8021 will keep re-asserting the TX Ready condition until the internal playback buffer is full, and the CT8021 has “pre-fetched” 20 frames. If the host needs to reduce the number of speech data frames “pre-fetched” by the CT8021, it can do so using the playback Set Buffer Depth command (528XH). This can be set so as to reduce the effective size of the internal buffer.

To terminate playback:

1. Host disables interrupts from the CT8021 by writing 0000H to the Hardware Control Register.
2. Host writes IDLE = 0000H command or STOP PLAYBACK = 5220H command to the Software Control Register.
3. CT8021 terminates playback, discards data in its internal playback buffer and clears TX Ready if set (in Hardware Status Register).
4. CT8021 writes status response to Software Status Register.
5. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.

Note that after writing the IDLE or STOP PLAYBACK command to terminate playback, the host should not attempt to access the Host Transmit Data Buffer Access Port even if the TX Ready bit is set.

When the IDLE or STOP PLAYBACK command is used to terminate playback, the CT8021 will discard data in its internal playback buffer. In some cases, this may not be the desired behavior. For example, when playing back TrueSpeech 5.3 data stored in a disk file, if the IDLE or STOP PLAYBACK command is issued immediately on encountering end-of-file (EOF), then the last 24 frames (720 ms) of the message will not be played. The Host may wish to make use of the playback Buffer Monitoring command (5212H). Using this command, the Host can poll the CT8021 and wait until all the internally stored speech data has been played out before issuing the IDLE or STOP PLAYBACK command.

Example 3 : Linear 16-bit Playback via Host Transmit Data Buffer Access Port - Using DMA

This example shows how to start playback in uncompressed 16-bit Linear format speech with data transfers performed via the Host Transmit Data Buffer Access Port. All data transfers in this example are performed by an external DMA controller.

Starting Playback

1. CT8021 is in IDLE state or RECORD state.
2. Host checks for CONTROL READY state in Hardware Status Register.
3. Host selects Poll Sync Mode for playback using the command 5202H.
4. CT8021 responds via the Software Status Register.
5. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
6. Host selects Playback by issuing the PLAYBACK “C1” command = 2C63H or 2C61H (PLAYMODE = [binary] 110) via the Software Control Register. This also informs the CT8021 that data transfers will occur via the Host Transmit Data Buffer Access Port (TFR Mode = [binary] 11 or 01).
7. CT8021 responds via the Software Status Register after a delay of up to 2 speech frame periods.
8. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register S1 = 2C63H or 2C61H. This clears the STATUS READY bit.
9. Host writes PLAYBACK “C2” command = 2000H to the Software Control Register.
10. CT8021 responds via the Software Status Register.
11. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register S2 = 2000H. This clears the STATUS READY bit.
12. Host writes PLAYBACK “C3” command = 20F0H to the Software Control Register. This command includes the requested number of words per frame to transfer. In this case (uncompressed 16-bit linear) this is 240 = F0H.
13. CT8021 responds via the Software Status Register.
14. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register S3 = 2000H. This clears the STATUS READY bit. The S3 response sets the actual number of words to transfer to 00H. This transfer count indicates the number of words to be transferred via the Software Control and Status Registers. Since this example uses TFR Mode = [binary] 11, no data is transferred via the Software Control and Status Registers, instead the data transfer takes place via the Host Transmit Data Buffer Access Port.
15. Host programs external DMA controller to accept DMA requests from the CT8021 (See DMA Direction bit in Hardware Control Register). Host configures a data buffer in the Host memory to store the speech data ready for DMA. The DMA memory buffer should be an integral multiple of the speech frame size. In the case of uncompressed 16-bit Linear, this should be N*480 (bytes).
16. Host sets up an ISR to handle an end-of-transfer (EOT) interrupt from the DMA controller when it reaches the end of the DMA buffer. Alternatively, if the DMA supports circular-buffering (auto-initialize), the Host can poll the DMA to determine when the end of the buffer is reached.
17. Host programs the Hardware Control Register to generate a DMA request from the CT8021 on the TX Ready condition by writing 00A0H (TX DMA Burst Mode, TX DMA enable) - or 01A0H (TX DMA Burst Mode, TX DMA enable, DMA Direction bit set).

The CT8021 is now in playback mode. The CT8021 will consume an uncompressed speech frame every 30 ms. The following sequence occurs between the CT8021 and the DMA controller to transfer data:

1. CT8021 is ready to receive 240 words (480 bytes) of uncompressed 16-bit speech data.
2. CT8021 asserts TX Ready.
3. TX Ready drives TXDREQ.
4. DMA controller responds to TXDREQ by asserting TXDACKN.

5. DMA controller performs a burst transfer of 16 words (32 bytes) from the Host DMA memory buffer to the CT8021.
6. CT8021 de-asserts TX Ready.
7. CT8021 transfers 16 words from the Host Transmit Data Buffer into its internal playback buffer.
8. Repeat 2-7 until all 240 words of the speech data frame have been transferred.

The Host processor also needs to supervise the operation of the DMA controller and arrange for the re-filling of the DMA buffer during playback (e.g. read from disk). When the DMA controller reaches the end of the DMA buffer, the Host must re-fill the data (read from disk). The Host can either arrange to receive an EOT interrupt from the DMA controller, or can periodically poll the DMA's internal transfer count register to determine the status of the DMA.

To terminate playback:

1. Host disables DMA requests from the CT8021 by writing 0000H to the Hardware Control Register.
2. Host writes IDLE = 0000H command or STOP PLAYBACK = 5220H command to the Software Control Register.
3. CT8021 terminates playback and clears TX Ready if set (in Hardware Status Register).
4. CT8021 writes status response to Software Status Register.
5. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.

Note that after writing the IDLE or STOP PLAYBACK command to terminate playback, the host/DMA should not attempt to access the Host Transmit Data Buffer Access Port even if the TX Ready bit is set.

Example 4: TrueSpeech 8.5 Playback using CT8005/CT8015 protocol via Software Control and Status Registers (Requires download of external TrueSpeech 8.5 coder module).

This example shows how to perform TrueSpeech 8.5 playback with TFR Mode = 00 using the CT8015 compatible protocol. Data is transferred via the Software Control and Status Registers. All data transfers are performed by the Host processor. The example shown uses the CT8015 Poll Sync Mode.

1. CT8021 is in IDLE or PLAYBACK state.
2. Host checks for CONTROL READY state in Hardware Status Register.
3. Host selects TrueSpeech 8.5 playback by writing the Select TrueSpeech Playback Rate command = 5230H to the Software Control Register.
4. CT8021 responds via the Software Status Register
5. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
6. Host selects Poll Sync Mode for playback using the command 5202H.
7. CT8021 responds via the Software Status Register.
8. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
9. Host writes Playback C1 command = 2C00H (TrueSpeech 8.5) to the Software Control Register.
10. CT8021 activates playback mode and performs internal synchronizations (1-2 frame delay) .
11. CT8021 responds with Playback status S1 = 2C00H after a delay of up to 2 speech frame periods.
12. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
- 13. Host sends Playback C2 command = 2000H.**
- 14. CT8021 responds with Playback S2 status = 2000H.**
- 15. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.**

16. Host sends Playback C3 with the number of (16 bit) words it wishes to transfer = 2010H. For TrueSpeech 8.5 the requested transfer count should be 16 = 10H.
17. CT8021 checks number of words it is able ready to transfer.
18. CT8021 responds with Playback S3 = 2000H or 2010H. If number of words is not zero DATAFLAG/ signal is asserted to indicate the beginning of data transfer.
19. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
20. If number of words to transfer is zero (S3 = 2000H), Host goes back to step 13.
21. **Optional:** Host waits for DATAFLAG/ signal to be asserted
22. Host writes 16-bit data word to Software Control Register
23. CT8021 responds by echoing the 16-bit data word in Software Status Register
24. Host waits for STATUS READY indication in the Hardware Status Register, then reads the CT8021 response from the Software Status Register. This clears the STATUS READY bit.
25. Repeat steps 22-24 until all (16) data words are transferred
26. CT8021 de-asserts DATAFLAG/ signal to indicate the end of data transfer
27. **Optional:** Host waits for DATAFLAG/ to be de-asserted
28. **Repeat steps 13-27 until playback finished**
29. To end playback host writes IDLE command or Playback Stop command and waits for status response.

Note: **bold text** indicates the playback software data transfer loop. By default, The DATAFLAG signal on external pin GPIO0 is disabled. See section on GPIO pins for more information.

9.8.4 Playback Frame Alignment Using Transmit Data Buffer

CT8021 provides an additional TFR_MODE = 01 [binary] which maintains strict 30ms data frame alignment for all playback rates. This mode was not available in older versions of the CT8020. The following section describes non-frame-aligned playback using TFR_MODE = 11 [binary]

If performing Playback via the Transmit Data Buffer with TFR_MODE = 11 [binary], the following should be noted:

When the CT8021 completes the transfer of a speech data frame from the Transmit Data Buffer into its playback queue (in the external data sram), the CT8021 immediately returns the (empty) Transmit Data Buffer to the host and grants the host access to this buffer. This causes the TX READY signal (and interrupt) to be re-asserted. This happens even when the CT8021's internal memory playback buffer is full.

In the case of **compressed** speech playback (TrueSpeech 8.5 - 4.1, G.728-10ms), the effect is simply to allow the host to transfer 1 extra compressed speech frame to the CT8021. When the host writes the next compressed speech frame to the CT8021, the CT8021 is unable to empty the Transmit Data Buffer into its internal memory since this memory is full. In this case, the speech data remains in the Transmit Data Buffer until the next 30ms frame period when space becomes available in the internal memory. In this “memory full + transmit data buffer full” state, the TX READY will not be set. This provides an indication to the host that the CT8021 is not ready to accept more data. The effect of all this is to allow the CT8021 to effectively buffer 1 extra frame. This “extra” frame is not included in the frame count returned by the Buffer Monitoring Command.

In the case of **non-compressed** speech (and G.728-20ms or G.728-30ms), the effect of this behavior is not as simple. With non-compressed playback (16-bit linear, 8-bit linear and 8-bit mu-law/A-law), the empty Transmit Data Buffer is also returned to the host at the end of transfer of each complete speech frame (120 or 240 words). However in this case, the Transmit Data Buffer is not large enough to take the entire next frame. Instead it is able to hold only the first 16 words of the next frame. The result of this is that the frame by frame transfer of data to the CT8021 will occur with a 16 word offset relative to the 120 or 240 word frame boundaries.

For example:

In 16-bit linear playback the 240 word speech frame is broken up into 15 sub-frames of 16 words each for transfer via the Transmit Data Buffer. So the pattern of data transfer, broken down into 30 ms frame periods, will appear as follows:

Playback frame number	Data transferred	Number of sub-frames transferred
1	240 words of frame 1 + 16 words of frame 2	16 (256 words)
2	224 words of frame 2 + 16 words of frame 3	15 (240 words)
3	224 words of frame 3 + 16 words of frame 4	15 (240 words)
Final Frame	224 words of final frame	14 (224 words)

In 8-bit playback (all types), the 120 word speech frame is broken up into 7 sub-frames of 16 words plus 1 sub-frame of 8 words.

Playback frame number	Data transferred	Number of sub-frames transferred
1	120 words of frame 1 (16,16,16,16,16,16,16,8) + 16 words of frame 2 (16)	9 (136 words)
2	104 words of frame 2 (16,16,16,16,16,16,8) + 16 words of frame 3 (16)	8 (120 words)
3	104 words of frame 3 (16,16,16,16,16,16,8) + 16 words of frame 4 (16)	8 (120 words)
Final Frame	104 words of final frame (16,16,16,16,16,16,8)	7 (104 words)

Of particular significance is the sequence of sub-frame sizes transferred by the host to the CT8021 each 30 ms period. For 8-bit playback as a result of the playback frame alignment offset, the sequence per frame is: 16,16,16,16,16,16,8,16.

In view of this it is recommended that host software performing 8-bit playback should re-check the status of the TX READY bit at least every 8 words transferred to the CT8021.

9.8.5 Select TrueSpeech Playback Rate

This command allows the host to select the TrueSpeech data rate used in compressed playback mode. The standard built-in available rates are: TrueSpeech 6.3, 5.3, 4.8 and 4.1 Kbps and G.728 16 Kbps. This command should be issued prior to starting playback using the PLAYBACK command. The TrueSpeech Playback Rate remains in effect until the next TrueSpeech Playback Rate command.

Command (523XH):

0101	0010	0011	RATE
(15-12)	(11-8)	(7-4)	(3-0)

Status (523XH):

0101	0010	0011	RATE
(15-12)	(11-8)	(7-4)	(3-0)

RATE	TrueSpeech Compression Rate	Compressed Frame Size (16-bit words)
0	8.5 (see note 1)	16
1	6.3 (G.723.1) (default)	12
2	5.3 (G.723.1)	10
3	4.8	9
4	4.1	8
5	G.728	16,32 or 48 (10,20 or 30 ms)
6	G.729AB (see note 1)	6 (10 ms)
7	G.722 (see note 1)	40 (10 ms)
8-15	reserved	

The Frame Size value indicates the number of 16-bit words in a compressed speech frame for each coding rate. When any of the TrueSpeech compression rates are selected, the frame size is always 16 words or less. Since the size of the CT8021 hardware buffer within the Host port is 16 words, the Host can “burst” write the complete speech frame of 16-8 words without needing to check the RX Ready bit before each individual word is read. Where the frame size is larger than 16 words, the CT8021 splits the data transfer into sub-blocks of 16 words or less.

Notes:

- The CT8021 supports the TrueSpeech 8.5, G.729AB and G.722 coders only as external downloadable software expansion modules. The host must ensure that the appropriate module is loaded to the CT8021 external program/data memory before attempting to activate any of these coders. Contact DSP Group for information.**

The l.s. two bits of the first data word of a G.723.1 6.3 or 5.3 speech frame are encoded to indicate the speech frame type.

During Playback (de-compression) the CT8021 will test the two l.s. significant control bits only when the current host selected TrueSpeech rate is 6.3 or 5.3. In this case, the CT8021 will automatically perform the correct decoding of the speech frames independently of which rate is selected by the host . This includes automatic decoding of the G.723.1 CNG (Comfort Noise Generation or “silence”) frames.

To support on-the-fly switching for Playback, Playback must be started with the 6.3 rate selected. This fixes the frame size used for Host-to-CT8021 transfer at 12 words. When transferring a 5.3 speech frame into the CT8021 when the 6.3 rate was initially selected, the Host should append two dummy words (zeros) at the end of the 10 word TrueSpeech 5.3 frame.

See also section on on-the-fly switching between TrueSpeech 6.3 and 5.3 data rates.

9.8.6 Inserting Silence During Compressed Speech Playback

When the CT8021 is performing playback in any of the TrueSpeech modes (8.5, 6.3, 5.3, 4.8 or 4.1) it will interpret a speech frame that contains all zeros as an instruction to output 30 ms (one frame period) of pure silence.

9.8.7 Activating The G.723.1 Frame Erasure Mechanism

When the CT8021 is performing playback of G.723.1 / TrueSpeech 6.3, 5.3, 4.8 or 4.1 speech frames, it will interpret a speech frames that has the first two words set to FFFFH and the remainder of the speech frame set to all zeros, as an instruction to perform frame erasure. When Frame Erasure is activated, the CT8021 attempts to construct a replacement for a missing speech frame by extrapolation from previous valid speech frames. The sending of a Frame Erasure frame to the CT8021 based upon detection of dropped or corrupted speech frames is the responsibility of the host software and system communication protocol. The G.723.1 speech frames themselves do not include any built-in error detection mechanism.

9.8.8 G.723.1 Comfort Noise Generation (CNG)

In accordance with Annex A of the ITU G.723.1 specification, the CT8021 will automatically decode CNG frames generated by the VAD (Voice Activity Detector) of a G.723.1 encoder. Decoding of CNG frames is based on examination of the l.s. 2 bits of the first data word of each G.723.1 speech frame. The purpose of the VAD/CNG frame in the G.723.1 Annex A specification is to provide approximate reconstruction of the background noise level and energy spectrum during periods of “silence” that occur between periods of active speech.

The least significant two bits of the first word of the TrueSpeech 6.3 and 5.3 frames are defined by the ITU-T G.723.1 v5.1 (final formal release) specification to indicate the speech data rate as shown:

Bit 1-0	Frame Type	Data Rate	Number of significant data words
00	0	TrueSpeech 6.3 data frame (12 words)	12
01	1	TrueSpeech 5.3 data frame (10 words)	10
10	2	Silence / Comfort Noise Generation frame (only first 2 words contain useful data) All other data words are set to zero	2
11	3	Repeat last CNG frame (only l.s. 2 bits of first data word are used) All other data words are set to zero	2 bits

Compatibility Note: the Bit 1-0 encoding of the CT8021 is different from that used in older firmware versions of the CT8020 which implemented pre-release versions of G.723.1, (firmware revisions 0109 and 0112 part numbers CT8020A11AQC and CT8020A11BQC). The full production release, CT8020D11AQC (firmware revision 0114) implements the final v5.1 G.723.1 release as do all versions of the CT8021.

When sending a type 3 frame to the CT8021 playback channel (l.s. two bits = 11), please note the requirement to distinguish this type of frame from the Frame Erasure frame (FFFF FFFFH). All the unused bits in the type 3 frame should be set to zero. Although the CNG frames require fewer bits for actual transmission, the host is required to pad the CNG frames with zeros when writing the frame to the CT8021 in order to make the CNG frame size match the size of the normal speech frame. The CNG frame should be in the form:

0000 0003H to avoid confusion with the special FFFF FFFFH frame erasure frame.

When reading or writing speech frames, to or from the CT8021, the number of physical words transferred for each individual frame is fixed during each record/playback session. For example, if the record/playback session is started in 6.3 mode, then all frames will contain 12 words. Frames which use less than this number of words for actual data will have the unused portion of the frame set to zero. This approach simplifies the coding of low level driver software such that it does not need to deal with variable sized data frames. This is particularly significant when dealing with DMA type transfer.

9.8.9 Reading the Playback signal Level Value

The current Playback signal average level value can be read using the command:

Command CF8FH

1100 1111 1000 1	111
(15-3)	(2-0)

Status:

Average Playback Level Value
(15-0)

The level value is calculated using the following algorithm:

LEVEL = 0

FOR each of 240 OUTPUT samples per frame
 IF OUTPUT < 0 THEN OUTPUT = -1 * OUTPUT
 LEVEL = LEVEL + OUTPUT

LEVEL = LEVEL / 256

Note that this is an average signal level (not r.m.s.). The division by 256 instead of 240 leads to a slight understatement of the true average level. The fractional error is $240/256 = 0.9375$, or 6.25%. Dividing by 256 instead of 240 requires less processing by the CT8021 device.

When the CT8021 is operated with a base frame size that is not set to 240, the level calculation is adjusted to report the same level as if the base frame size was set to 240. However since the CT8021 will update and report the level for each frame processed, with a smaller base frame size the CT8021 will report the level measured over a shorter period. This may result in more frame-by-frame variation in the signal level measured. When measuring the level of a signal with lower frequency components, or with a low frequency modulation, more “ripple” may be observed in the reported signal level. One example of this would be reporting the signal level of a 350+440 Hz dial tone.

Level Scale

The level scale is relative to the maximum positive full scale value of 7FFFH (decimal 32767). This corresponds to an input signal level of approximately +3dBm0 from the external mu-law codec.

The precise input signal level scale can be calculated thus:

$$\text{Output Level in dBm0} = +3.17 - 20 \log(\text{base } 10) \left(\left(\frac{\text{output level}}{32767} \right) * \frac{256}{240} \right)$$

so the default threshold level of 0100H = 256 decimal, corresponds to -38.4 dBm0.

To calculate the actual average output signal level, read the LEVEL from the device and then perform the following calculation:

$$\text{CORRECTED LEVEL} = (\text{LEVEL} * 256) / 240$$

$$\text{output level in dBm0} = 3.17 - 20 * \log_{10}(\text{CORRECTED LEVEL} / 32767)$$

9.9 Stop Record/Playback

Playback and Record can be stopped using the IDLE command. However, this causes both Playback and Record to be stopped. To stop Playback without stopping Record or to stop Record without stopping Playback, the STOP command may be used. There are three stop commands, two for Playback and one for Record:

STOP RECORD:

Command: 5120H

Status: 5120H

STOP PLAYBACK:

Command: 5220H

Status: 5220H

STOP PLAYBACK and flush codec output buffer (immediate stop) RECOMMENDED

Command: 5221H

Status: 5221H

In PLAYBACK mode, STOP PLAYBACK causes the CT8021 to enter IDLE mode. In FULL DUPLEX speech mode, this command changes the CT8021 state to RECORD.

In RECORD mode, STOP RECORD causes the CT8021 to enter IDLE mode. In FULL DUPLEX speech mode, this command changes the CT8021 state to PLAYBACK.

Use the STOP PLAYBACK with flush option to terminate playback if the record channel remains active after playback has completed.

When operating playback only or record only, it is recommended that the IDLE command be used to stop playback/record.

9.10 PLAYBACK & RECORD VOLUME CONTROL

9.10.1 Read Record/Playback Volume Control Command:

Command: CF00H/CF01H

1100 1111 (15-8)	0000 00 (7-2)	0 (1)	INPUT (0)
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Status:

RECORD/PLAYBACK VOLUME

INPUT: Set to 0, reads current playback (output) volume
 Set to 1, reads current record (input) volume

9.10.2 Set Record/Playback Volume Control

Command (C1): CF02H/CF03H

1100 1111 (15-8)	0000 00 (7-2)	1 (1)	INPUT (0)
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Status (S1):

1100 1111 (15-8)	0000 00 (7-2)	1 (1)	INPUT (0)
---------------------	------------------	----------	--------------

INPUT: Set to 0, writes playback (output) volume
 Set to 1, writes record (input) volume

Command (C2):

RECORD/PLAYBACK VOLUME

Status (S2):

RECORD/PLAYBACK VOLUME

The Record/Playback volume is a “Q8” number which is used to scale the input (record) signal or output (playback) signal (see notes below) .

The scaling performed is :

input or output signal multiplied by (RECORD or PLAYBACK VOLUME)/100H.

So that a record/playback volume setting of:

100H scales the signal by 1.0
200H scales the signal by 2.0
80H scales the signal by 0.5

No protection is given against over-scaling, if the multiplication factor times the input signal exceeds 16 bits, overflow (distortion) will occur.

With an external G.711 A-law or Mu-law codec, the CT8021 provides bit-transparent transfer of data between the codec and the host when operated in A-law/Mu-law record/playback mode with a volume setting of 1.0 (0100H).

For best audio performance, optimal scaling of the record signal should be performed by external hardware before A-to-D conversion by the codec, such that the full dynamic range of the codec is utilized. Failure to use appropriate external gain will result in increased quantization noise being introduced into the recording. Note however, that excessive external gain may cause problems with correct operation of the Acoustic Echo Canceller. Also if the AEC feature is used, then the external gains must be fixed.

Do not attempt to use the CT8021 AEC feature with an external hardware AGC or ALC circuit.

Note that the 8 bit PCM mu-law codes produced by the codec, when converted to a linear representation occupy 14 bits (including the sign bit). The CT8021 handles these 14 bit codes using a 16 bit word representation such that the 14 mu-law bits are “left-justified” within the 16 bits. The lower (least significant) two bits are padded with zeros. This ensures compatibility with digital recordings originating from external systems where a full 16 bits of significant input are encoded or compressed. When the CT8021 is used to decode or de-compress such recordings for output to a mu-law codec, the least significant two bits of the de-compressed 16 bit signal are discarded before conversion to mu-law encoding (and output to the codec) takes place. For 8 bit PCM A-law, the linear representation occupies 13 bits. This is also left-justified within 16 bits and the lower bits filled with zeros.

“Q8” format is a notation used to represent fractional numbers. In normal integer arithmetic, it is not possible to represent numbers less than zero or any fractional number. To overcome this limitation multiplication by fractional numbers is performed by scaling up the multiplier by a fixed base number and then scaling down the result by the same base number. The base number provides an effective decimal point placed at the base number. That is, when the multiplier equals the base number, the effective multiplication is by 1.0 (multiply by base number, then divide by base number). To keep the scaling down process simple, the number base chosen is always a multiple of 2. This allows scaling down (dividing) by the base to be performed as a simple binary bit-shift. In “Q8” arithmetic, the “decimal point” is placed between bits 8 and 7 of the 16 bit binary multiplier word (l.s. bit is bit 0, ms bit is bit 15). The base number used is 2 to the power 8 or $256 = 100H$. So to multiply by 0.5, the Q8 number 80H ($= 0.5 * 100H$) is used thus: $400H * 0.5$ is performed as $400H * 80H / 100H$. The divide by 100H is performed not as a division but as a binary 8 bit right shift.

The record and playback volume controls described here do not affect the operation of the CT8021 when in standalone speakerphone mode (analog phone line speakerphone). See Speakerphone mode description for volume control in this mode.

9.10.3 Automatic Gain/Level Control (AGC or ALC)

The CT8021 includes an AGC feature that can be used in record mode for automatic level adjustment when recording messages.

For optimal Acoustic Echo Cancellor (AEC) performance, the Host should make sure that the AGC is switched off before activating the AEC.

The purpose of the Acoustic Echo Cancellor (AEC) is to attenuate, as far as possible, acoustic echoes from the speaker that are picked up by the microphone. The purpose of the Automatic Gain Control (AGC) is to amplify low level signals that are picked up by the microphone. As a result of this, the AGC and AEC tend to operate in conflict with each other.

However the CT8021 incorporates an additional feature that reduces the amount of conflict that exists between the AEC and AGC. The AGC includes a feature that reduces that rate of gain increase allowed when the AEC status indicates that the active speech direction is from the far-end talker. If the user requires that the AGC and AEC must operate concurrently, then the following recommendations should be followed:

1. Limit the maximum AGC gain to the lowest value that produces acceptable results.
2. Make use of the hold time feature of the AGC to minimize gain increases during short conversation pauses.
3. Increase the amount of Echo Suppression (loop attenuation) used by the echo cancellor to compensate for the additional gain introduced by the AGC.
4. Activate the “advanced” AEC features.

Activating the AGC, disables the Record Volume Control. The Record Volume Control has no effect when the AGC is active, however, Record Volume Control setting changes are retained by the CT8021 device and will become effective when the AGC is de-activated.

The AGC provides the following parameters which are programmable by the host:

Max Gain:	The maximum gain (max amplification) provided by the AGC
Min Gain:	The minimum gain (max attenuation) provided by the AGC
Attack Threshold:	The AGC output signal level above which attack occurs (gain reduction) in the attack phase and below which decay occurs when in the decay phase (gain increase).
Attack time-constant:	This controls the rate at which the gain decreases during the attack phase
Decay time-constant:	This controls the rate at which the gain increases during the decay phase.
Hold Time:	The length of time that the AGC gain is held constant following the end of the attack phase and before the beginning of the decay phase
Start Gain:	The initial gain setting to use when the AGC is activated. This may be set anywhere between the min and max gain values.

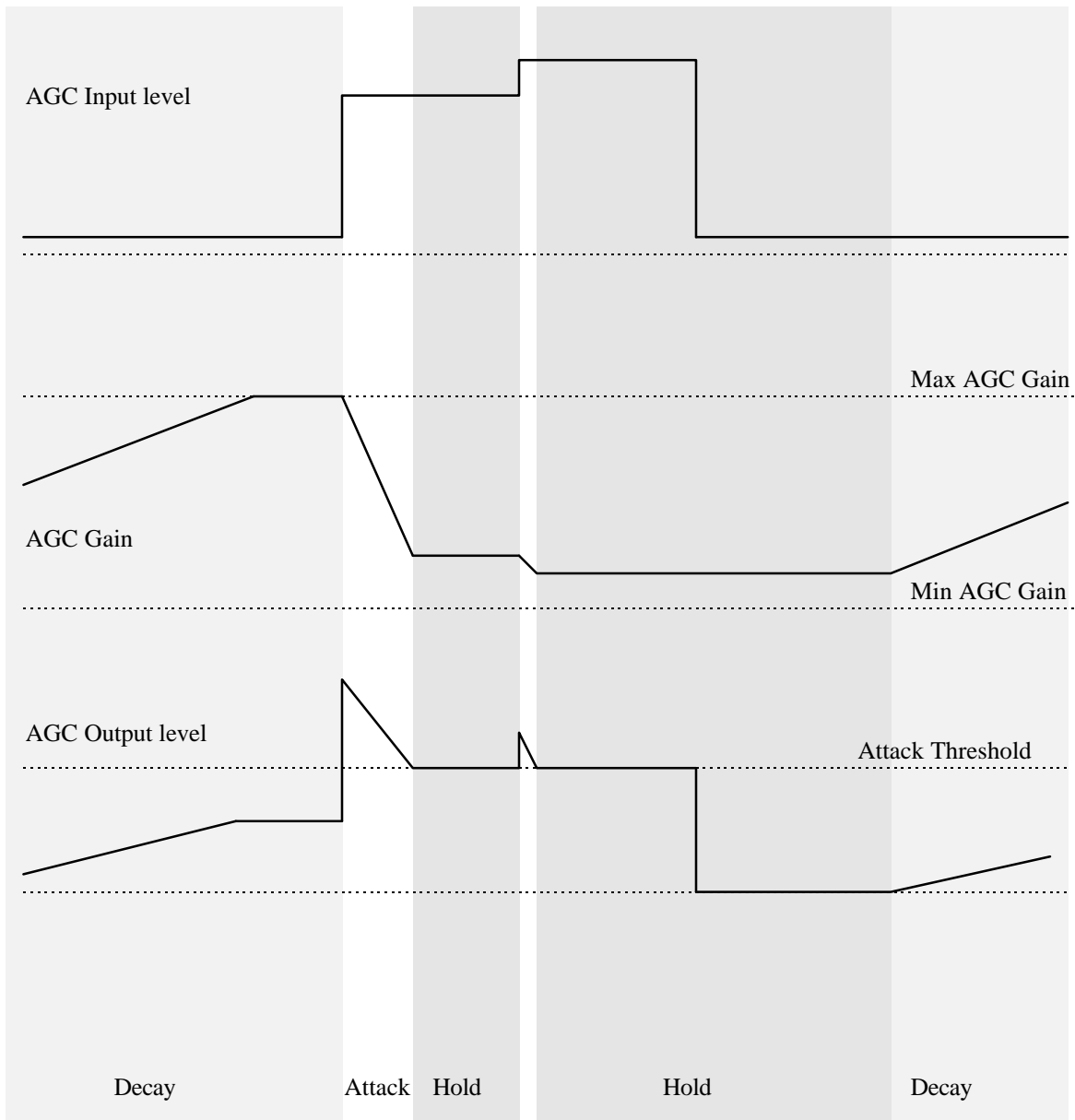


Figure 9-1 AGC Attack Hold & Decay

The figure above, shows the AGC Input signal level (or signal “envelope”), the AGC Gain and the AGC Output signal level.

During the initial Decay phase shown, the input signal is constant at a low level. Since the AGC output level is below the Attack Threshold, the AGC gain increases which in turn increases the output signal level. In this example the gain increases until the AGC Max Gain value is reached, at which point no further increase in gain is possible.

In the next phase, the input signal level suddenly increases, causing the output signal level to exceed the Attack Threshold. At this point the AGC enters the Attack Phase, and the AGC gain is decreased until the output signal level is below the Attack Threshold.

Once the output level is below the Attack Threshold once more, the AGC enters the Hold Phase. In this phase the output level remains at, or below the Attack Threshold. On entering the Hold phase, the AGC sets an internal Hold timer, which then starts to decrement.

Before, the Hold timer can expire, the input signal level increases again, resulting in a second attack phase. After the output level is reduced below the Attack Threshold again, the AGC re-starts the Hold phase.

Next the input signal level is suddenly reduced. Since the AGC is still in the Hold phase, no change in the AGC gain occurs, and the AGC output follows the reduction in input signal level.

When the Hold timer expires, the AGC enters the Decay phase again, and the AGC gain starts to increase. The rate at which the gain increases during the Decay phase (decay time-constant) and the rate in which the gain reduces during the Attack phase (attack time-constant) are programmable. Best results for speech recording are normally achieved by selecting a fast Attack time-constant (allows rapid gain reduction) and a slow Decay time-constant (allows only slow gain increase). This configuration causes the AGC to primarily respond to the peak speech level, thus providing relative attenuation of any background noise audible during gaps in the speech. If the Hold time is set to 2-3 seconds, the background noise will remain attenuated even during short pauses in the input speech.

The AGC is automatically reset to the Start Gain value every time a new record session is started. (That is: every time a Record C1 command is issued)

9.10.3.1 Programming the AGC

Setting AGC parameters (CF9xH) :

Command C1:

1100 1111 1001 0 (15-3)	SELECTOR (2-0)
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Status S1:

1100 1111 1001 0 (15-3)	SELECTOR (2-0)
----------------------------	-------------------

Command C2:

Parameter Value (15-0)

Status S2:

Parameter Value (15-0)

Where SELECTOR is defined:

SELECTOR	Parameter Description
000	AGC Min Gain. The 16 bit parameter value used to set the AGC Min Gain is treated as a “Q8” value. That is, the decimal point used to construct the value is considered to be to the right of bit-8. So in Q8 notation, the value 1.00 is represented by the hex number 0100H (decimal 256). The default value for this parameter is 0020H, this represents the fraction 0.125 in Q8 format. So the default min gain is a multiplier factor of 0.125 or one-eighth or approximately -18 dB. The min gain value can also be regarded as the maximum attenuation of the AGC. In this case, the maximum attenuation is by a factor of 8 or approximately +18 dB.
001	AGC Max Gain. The 16 bit parameter value used to set the AGC Min Gain is treated as a “Q8” value (see min gain). The default value for this parameter is 1000H, this represents the fraction 16.0 in Q8 format. So the default max gain is a multiplier factor of 16 or approximately 24 dB. The default max and min gain, give the AGC gain a range from -18 dB to +24 dB which is a range of 42 dB.
010	AGC Start Gain. The 16 bit parameter value used to set the AGC Start Gain is treated as a “Q8” value (see min gain). The default value for this parameter is 0800H, this represents the fraction 8.0 in Q8 format. So the default max gain is a multiplier factor of 8 or approximately +18 dB. Since the default decay time constant is much larger than the default attack time constant, the default start gain is set high. This is because the time taken to reduce the gain (attack) is much less than the time taken to increase the gain (decay). For example if the initial speech level is very high, the gain will be reduced quickly (attack) to adjust to the high input level. If the initial speech level is low, since the decay time constant is large (slow), increasing the gain to amplify the low speech level may take several seconds. Hence starting with a high gain that can be quickly reduced provides the best initial performance.
011	AGC Hold Time. The 16 bit parameter value used to set the AGC Hold time has units of 250 microseconds. The default value of this parameter is 1F40H (8000 decimal) which corresponds to a Hold Time of 2.00 seconds.
100	AGC Attack Time Constant. See time constant table. The default value of this parameter is 0005H.
101	AGC Decay Time Constant. See time constant table. The default value of this parameter is 000DH (14 decimal).
110	AGC Attack Threshold. The AGC Attack Threshold is the AGC output value above which the AGC enters the Attack phase (gain reduction). The default value of this parameter is 2000H. The value can be set anywhere in the range 0000H to 7FFFH. Note that 7FFFH is the maximum internal (digital) signal level. So a value of 2000H selects AGC attack when the peak signal amplitude is above 0.25 full scale.
111	AGC On/Off. Setting this parameter to 0001H enables AGC operation, setting the parameter to 0000H disables AGC operation. By default the parameter is 0000H with the AGC in the Off state.

Reading AGC parameters (CF8xH):

The AGC parameters programmed can be read back using the command below:

Command:

1100 1111 1000 0 (15-3)	SELECTOR (2-0)
----------------------------	-------------------

Status :

Parameter Value (15-0)

Time Constant Table

Time Constant Parameter	attack/decay fraction per 125 microseconds = $1/2^{(n+1)}$ $U(n+1) = U(n) - U(n) * \text{fraction}$	Time Constant in ms (assuming 8 KHz sample rate)
0	0.5	250 us
1	0.25	512 us
2	0.125	1 ms
3	0.0625	2 ms
4	0.03125	4 ms
5	0.015625	8 ms
6	0.0078125	16 ms
7	0.00390625	32 ms
8	0.001953125	64 ms
9	0.0009765625	128 ms
10	0.00048828125	256 ms
11	0.000244140625	512 ms
12	0.0001220703125	1024 ms
13	0.00006103515625	2048 ms
14	0.00003051757813	4096 ms
15	0.00001525878906	8192 ms

In the Attack phase (gain decreases) , the AGC gain is recalculated every codec sample (125 us) such that

$$\text{AGC gain} = \text{AGC gain} - ((\text{AGC gain} - \text{min AGC gain}) * 1/2^{(n+1)})$$

This behaves as an approximately exponential decay from the current instantaneous gain value towards the min AGC value.

In the Decay phase (gain increases), the AGC gain is recalculated every codec sample (125 us) such that:

$$\text{AGC gain} = \text{AGC gain} + ((\text{max AGC gain} - \text{AGC gain}) * 1/2^{(n+1)})$$

This behaves as an approximately exponential decay from the current instantaneous gain value towards the max AGC value.

Note that the time constant, is the time that it takes for an initial value to decay to 1/e of its initial value where $e = 2.71828$.

9.11 Pre-Scaling of DTMF and Call Progress Filter Input

The CT8021 provides a pre-scaler multiplier which allows independent adjustment to be made to the input signal level that goes to the tone detection sub-system (DTMF and Call Progress Tone Filters). This pre-scaler (input volume) control is separate from the record and playback volume controls.

The pre-scaler setting can be read using the following command:

Command = CF05H
Status = pre-scaler value (default is 0040H)

The pre-scaler setting can be written using the following command:

C1 Command = CF07H
S1 Status = CF07H

C2 Command = new pre-scaler setting
S2 Status = new pre-scaler setting

The default value for the pre-scaler is 0040H, and it acts as a linear scaler. Changing the pre-scaler setting to 0020H will multiply the signal by 0.5 relative to the normal setting.

9.12 HOST SYNC MODES

The playback and record sync modes can be set independently :

RECORD SYNC:

Set CODEC sync mode : 5100H
Status: 5100H

Set DATA sync mode: 5101H
Status: 5101H

Set POLL sync mode: 5102H
Status: 5102H

Set HOST sync mode: 5103H
Status: 5103H

PLAYBACK SYNC:

Set CODEC sync mode : 5200H
Status: 5200H

Set DATA sync mode: 5201H
Status: 5201H

Set POLL sync mode: 5202H
Status: 5202H

Set HOST sync mode: 5203H
Status: 5203H

Notes

1. The sync modes should normally only be changed when the device is in IDLE mode.
2. CODEC sync mode is the default following reset
3. DATA sync mode is selected automatically during Host-to-Host compression or decompression. This sync mode will remain in effect until a new sync mode command is issued. Once Host-to-Host compression has been started (after the C1 command), it is possible to issue another sync mode command in order to change to a different the sync mode.
4. Select POLL sync mode if TFR_MODE = [binary] 11. Use this mode for DMA and if performing data transfers via the Host Transmit/Receive Data Buffer Access Ports.

9.13 PLAYBACK & RECORD BUFFER CONTROL

9.13.1 Monitoring Buffer Depth

The buffer monitoring commands for the record and playback channels return the number of frames currently present in the respective receive and transmit queues. Each frame corresponds to 30 ms of speech (assuming the base frame size is set to 240). The host uses these commands for buffer management purposes. The frame count does not include data held in the host interface transmit or receive data buffer registers.

RECORD:

Read Buffer Depth Command: 5112H
 Status: the number of record frames currently in the receive buffer.

PLAYBACK:

Read Buffer Depth Command: 5212H
 Status: the number of playback frames currently in the transmit buffer.

9.13.2 Speech Frame Create

The frame create command creates an additional speech frame in the transmit or receive buffer provided that the buffer is not full. The frame is created by copying the most recent frame in the buffer.

RECORD:

Create Record Frame: 5117H
 Status: 0 or 1 The number of frames created. If the buffer is full the status reply will be 0. If the buffer is not full, and the frame can be created, the status reply will be 1.

PLAYBACK:

Create Playback Frame: 5217H
 Status: 0 or 1 The number of frames created. If the buffer is full the status reply will be 0. If the buffer is not full, and the frame can be created, the status reply will be 1.

Note that for playback the G.723.1 / TrueSpeech 6.3, 5.3, 4.8 and 4.1 rates include a built-in frame creation mechanism that is activated using a specially encoded speech playback frame. This mechanism allows replacement of a dropped or corrupted speech frame by extrapolation of speech data from previous frames. See section on Frame Erasure.

9.13.3 Speech Frame Delete

The frame delete command deletes a speech frame from the transmit or receive buffers provided that the buffer is not empty. The status reply indicates the number of frames deleted.

RECORD:

Delete Record Frame: 5116H
 Status: 0 or 1 The number of frames deleted. If the buffer is empty at the time this command is called, there will be no frames to delete and the status reply will be 0. If the buffer is not empty, and a frame is deleted, the status reply will be 1.

PLAYBACK:

Delete Playback Frame: 5216H
 Status: 0 or 1 The number of frames deleted. If the buffer is empty at the time this command is called, there will be no frames to delete and the status reply will be 0. If the buffer is not empty, and a frame is deleted, the status reply will be 1.

9.13.4 Playback & Record Buffer Freeze (Pause)

The buffer freeze command stops output from the playback buffer in the playback channel and input to the record buffer in the record channel. The buffer freeze command can be used to implement a pause feature. When the playback channel is paused, it outputs silence to the external codec. Freezing the playback channel can also be used at the start of playback to pre-load the channel with buffers before the actual start of playback. This creates a reservoir of speech frames within the CT8021 which can be used for smoothing out fluctuations in the rate at which speech frames arrive for playback. The state of the CT8021 internal freeze flag bits can be modified in any mode of operation : IDLE, PLAYBACK, RECORD or FULL DUPLEX.

RECORD:

Freeze Record Channel: 5113H
 Status: 5113H

Un-Freeze Record Channel: 5114H
 Status: 5114H

Read Record Freeze State: 5115H
 Status: 0 or 1 The status returns 1 if the channel is currently in the freeze state, 0 otherwise.

PLAYBACK:

Freeze Playback Channel: 5213H
 Status: 5213H

Un-Freeze Playback Channel: 5214H
 Status: 5214H

Read Playback Freeze State: 5215H
 Status: 0 or 1 The status returns 1 if the channel is currently in the freeze state, 0 otherwise.

Compatibility Note: Revision 0112 onwards, of the CT8020 firmware (CT8020A11BQC) fixed a minor bug which caused incorrect operation of the Buffer Freeze command for Playback when the CT8020's internal playback buffer was completely empty. In earlier versions of the CT8020 firmware, the CT8020 would not output silence when Buffer Freeze was activated if the Playback buffer was empty. In this situation, the CT8020 would continuously re-output the most recent two speech frames. Please also refer to the Playback Auto-Repeat feature described later in this document.

9.13.5 Buffer Depth Limit

The buffer depth limit command can be used to limit the number of speech frames held in the playback or record channel buffers. This allows the host to strictly limit the amount of delay and latency that the buffering of speech frames in the CT8021 can add to an end-to-end speech link. At the same time, the CT8021's buffering capability helps to reduce the load on the host processor by permitting more relaxed real-time deadlines in moving data into or out of the CT8021. The CT8021 has 480 bytes available for storage space for each channel (physically located within the external data sram). This is sufficient for up to:

- 1 uncompressed (480 byte) 16-bit linear frame
- 2 uncompressed (240 byte) 8-bit/A-law/mu-law frames
- 15 TrueSpeech 8.5 frames
- 20 TrueSpeech 6.3, 5.3, 4.8 or 4.1 frames
- 5 G.728 frames using 48 word (30ms) frame size
- 7 G.728 frames using 32 word (20ms) frame size
- 15 G.728 frames using 16 word (10ms) frame size
- 30 G.728 frames using 8 word (5 ms) frame size
- 60 G.728 frames using 4 word (2.5 ms) frame size

The CT8021 does not check the buffer limit set by the host, but simply takes as the actual limit the minimum value of the physical buffer capacity and the host programmed limit. For example, in mu-law mode, if the host programs a limit of 10 frames, the limit used will actually be only 2, the physical capacity of the device. In TrueSpeech 6.3 mode, if the host programs a limit of 10 frames, the limit used will be 10 since this is within the physical capacity of the device.

RECORD:

Set Record Channel Limit: 5180H + X where X is in the range 0 to 60 (decimal)
 Status: 5180H + X

Read Record Channel Limit: 5118H
 Status: 00XXH where XX is the limit programmed

PLAYBACK:

Set Playback Channel Limit: 5280H + X where X is in the range 0 to 60 (decimal)
 Status: 5280H + X

Read Playback Channel Limit: 5218H
 Status: 00XXH where XX is the limit programmed

9.13.6 Playback Auto-Repeat

If data under-run occurs during playback, the transmit buffer becomes empty and the CT8021 will default to repeatedly outputting the most recent speech frame up to a (default) limit of 8 times. This behavior is appropriate to “mask” a short 1 or 2 frame period in which playback data is not available. Repeating the most recent frame 1 or 2 times causes the least perceptible break in the continuity of playback. However, if the data under-run condition persists for a longer period of time, the repeated playback of a single frame can cause annoyance to the user. The auto-repeat limit can be programmed by the host in the range 1-15. Programming the auto-repeat limit to zero, will cause auto-repeat to operate continuously. When the auto-repeat limit is exceeded the CT8021 will output silence until more playback data becomes available.

Command: 525XH

Status: 525XH

Where X selects the number of repeats in the range 1 - 15. If X is set to zero, the CT8021 will continuously auto-repeat the most recent frame.

9.14 FULL DUPLEX SPEECH MODE

Full duplex (DSVD) speech mode is the simple combination of PLAYBACK and RECORD operating concurrently (and independently). Full duplex mode is entered by starting both Playback and Record in either order.

If the Host wishes to take advantage of the fact that the record and playback data streams are synchronous within the CT8021, in order to use a single interrupt source for generating an interrupt for both Playback and Record, then the RX READY interrupt should be used. Internally the CT8021 process the Playback data stream before processing the Record data stream. So under normal circumstances, the RX READY (record) indication will be asserted after TX READY (playback). Because of the Playback-then-Record order, at the time the RX READY condition is asserted, the TX READY will normally also be set. At this time the Host can perform data transfers for both TX and RX directions.

The only disadvantage to this approach is the restriction imposed on the Host response time when the CT8021's internal playback buffer is **empty** and the internal DSP is operating at close to 100% load (Full Duplex G.728 plus concurrent Acoustic Echo Canceller). In a minimum delay environment, the host may wish to attempt to keep the CT8021's internal buffers in the empty state for as much of the time as possible. Reducing the amount of speech buffered within the CT8021 will keep the end-to-end delay to a minimum. Under these circumstances, the response time of the Host using the RX READY assertion to completion of the Playback data transfer to the CT8021 must be shorter, otherwise Playback data under-run will occur.

The diagram below shows typical CT8021 internal processing activity during one 30 ms frame period.

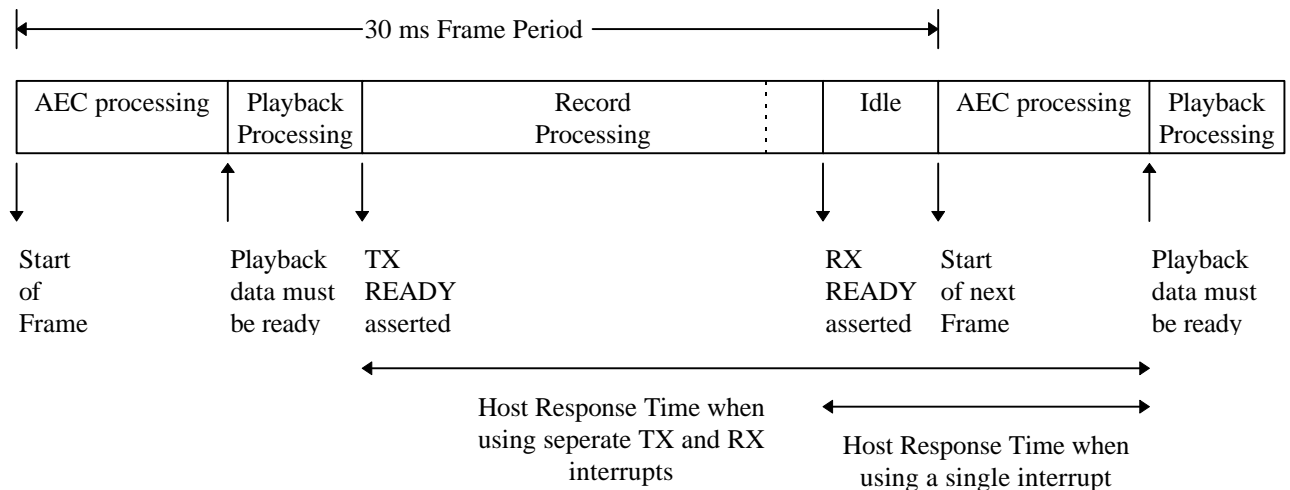


Figure 9-2 CT8021 internal processing activity

The Host Response Time is only an issue if the Host is attempting to implement a **minimum** delay system, where the CT8021's internal buffers are not used to provide additional latency (buffering or de-jitter). If minimum delay is not required, the CT8021's playback buffer can be used to provide extra buffering such that there is always at least 1 speech frame ready and waiting in the playback buffer. In this situation, the Host always has at least a 1 frame period latency in which to respond to the TX READY condition.

9.15 ACOUSTIC ECHO CANCELLOR IN DSVD MODE

The Acoustic Echo Canceller (AEC) may be used to cancel echoes present in the input signal during full duplex (DSVD) speech mode. It can also be used when only playback is in operation to remove playback echoes from the signal input to the DTMF and Call Progress Tone Detectors. When used to cancel playback only echoes, it is recommended that the Echo Suppressor (loop attenuation) feature and Advanced AEC feature is disabled. For the AEC to function, it requires both an outgoing (playback) and incoming (for record or tone detection) audio data stream for the AEC to operate upon. When the AEC is operating, the Automatic Gain Control (AGC) feature (available in conjunction with record mode) should be de-activated. However, the playback and record level controls are still available when the AEC is active. Care must be taken when setting the record and playback levels so as not to compromise operation of the AEC. The purpose of the AEC is to reduce as much as possible, the acoustic echo of the outgoing speech (playback) reflected from the speaker into the microphone (record speech stream). Increasing the playback or record volumes, will increase the level of audible echo. Note that the record level setting does not affect the signal level input to the tone detectors.

The AEC is switched on or off using the following commands.

Command: B001H AEC on
Status B001H

Command: B002H AEC off
Status B002H

The above commands can be issued at any time, but the AEC will only be active when playback is active (otherwise there is no echo to cancel).

The commands:

Get Speakerphone Status

Set Speakerphone Parameters (speaker and line volume and loop attenuation)

Speakerphone Configuration (loop adjustment mode)

can be used to control the AEC. See the section on (standalone) Speakerphone mode for detailed descriptions of these commands.

The speakerphone commands listed above only work when record and/or playback are already active. Sending these commands when the CT8021 is in IDLE mode will cause (improper) entry to standalone speakerphone mode.

Attempting to train the Electrical Echo Canceller (EEC) in DSVD speakerphone mode is not permitted. The EEC is used **only** in standalone speakerphone mode to cancel the stationary (fixed, non-varying) electrical signal echo from the telephone line (via the second, optional, codec0).

Use the following sequence of commands (from idle mode)

Start Playback and/or Start Record

AEC on

Set Speakerphone Parameters

optional - Speakerphone Configuration

optional - Get Speakerphone Status

See Appendix A for additional information.

9.15.1 Concurrent AEC and Speech Operation

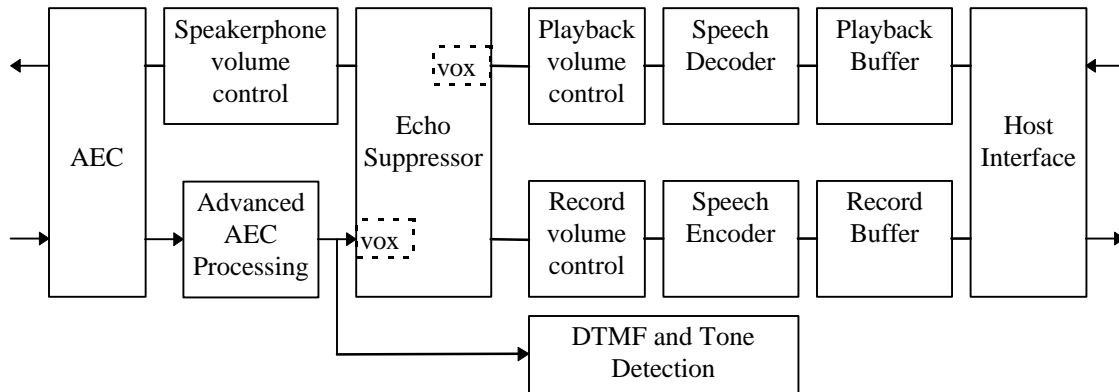


Figure 9-3 AEC and Speech Compression

This diagram shows the relationship of the Speakerphone Acoustic Echo Canceller (AEC) and Echo Suppressor components to the record/playback channels when DSVD speakerphone mode is used.

Note that the speech “direction” decision for the Echo Suppressor is made by the “vox” detectors at the inputs of the Echo Suppressor. Changes to the playback level made using the playback volume control (recommended) will be seen by the “vox” and affect the direction decision. Changes to the playback level made using the Speakerphone volume control (Set Speakerphone Parameters command) will not be seen by the “vox” and will not affect the direction decision.

When using the AEC to remove playback echoes at the Tone Detector inputs only (and speech compression/record is not used), it is recommended that the Echo Suppressor (loop attenuation) is set to 0dB, the automatic loop adjust mode is disabled and the Advanced AEC feature is off:

```

enable AEC
disable Advanced AEC
enable DTMF and/or Tone Detectors
Start playback
set loop attenuation to 0dB
switch automatic loop adjustment off
set loop attenuation to 0dB (to force removal of any attenuation added by the auto loop adjustmet mode)
  
```

Once this combination of features has been set up, you may stop and restart playback without needing to reprogram the AEC.

9.15.2 AEC Tail Length

The CT8021 supports programmable length AEC Tail-Length up to 30ms. The maximum available Tail-Length depends on which other features of the CT8021 are active as show in the table below:

CT8021 Operating Mode	Maximum AEC Tail-Length	
	milliseconds	AEC Filter Taps
Full Duplex TrueSpeech 6.3, 5.3, 4.8, 4.1 (G.723.1) with DTMF Detection and Call Progress Tone Detection	30	240
Full Duplex G.728 only (30ms frame size)	20	160
Full Duplex G.728 with DTMF Detection and Call Progress Tone Detection (30 ms frame size)	10	80
Full Duplex G.728 only (10ms frame size)	10	80
TrueSpeech 8.5 (external) with DTMF Detection and Call Progress Tone Detection	30	240
Standalone Speakerphone Mode	30	240

Each AEC Filter Tap is equivalent to 125 microseconds of Tail-Length (at 8.0 KHz).

The AEC Tail-Length is programmable in units of 2ms or 16 Taps.

The AEC Tail-Length may only be changed when the CT8021 is in IDLE mode and the AEC is switched off.

To program the AEC Tail-Length use the command:

```
Command    B0X0H
Status     B0X0H
```

Where X0H is the AEC Tail-Length in Taps:

```
B050H      Selects 10 ms , 80 Taps
B0A0H      Selects 20 ms , 160 Taps
B0F0H      Selects 30 ms , 240 Taps
```

The default AEC Tail-Length is 10ms, 80 Taps

9.15.3 Controlling the AEC adaptation (training) rate

The host may limit the AEC adaptation (training) rate so as to limit the CT8021 processing resources consumed by the AEC. This also reduces the adaptation rate of the AEC (increases the time taken to train or converge the AEC). Reducing the adaptation rate increases the time taken for the AEC to train, but it also increases the time taken for the AEC to diverge (un-train). Reducing the **adaptation** rate increases the **stability** of the AEC.

The AEC adaptation rate may be controlled in real-time by the host so that the host may enable rapid adaptation in the initial stages of a call to quickly train the AEC and then select slower adaptation in the later stages of a call to increase the AEC stability making the AEC more immune to “double-talk” conditions which can de-stabilize the AEC.

The processing load (mips) consumed by the AEC feature is the sum of three main components:

- control overhead
- actual cancellation
- adaptation (training)

The control overhead is approximately constant.

The cancellation load is proportional to the AEC Tail-Length and does not vary frame by frame.

The peak adaptation load is proportional to the AEC Tail-Length, but switches on or off on a sample by sample basis as determined by the internal control function (speech direction).

For a given frame size and tail-length, the processing load due to the first two factors is fixed.

The adaptation load varies on a frame-by-frame basis and depends on the speech direction which is evaluated on a sample-by-sample basis within the speech frame.

To prevent the host from over-committing the processing resources of the CT8021 (by configuring an inappropriate combination of Tail-Length, Speech Coder and Frame Size) , the AEC function monitors the CT8021 processing load in real-time. If the AEC senses that the CT8021 DSP processor is over-committed, it can automatically throttle (limit) the AEC adaptation process to reduce the amount of processing resource consumed by the AEC adaptation.

The AEC adaptation throttle limits the number of adaptation cycles that may run during the processing of each speech frame. By default the maximum number of adaptation cycles is set to the number of samples in the speech frame. For example: a 30ms speech frame contains 240 samples, so the AEC may run the AEC adaptation operation up to a maximum of 240 times during the speech frame. The number of adaptation cycles actually run depends on the measured speech direction for each speech sample within the frame. Adaptation of the AEC is enabled only when the speech direction is in the playback (decompression) direction. If the speech direction is sensed to be in the record (compression) direction, no adaptation (training) takes place. The speech direction is sensed on a sample-by-sample basis, so the number of adaptation cycles that run in a particular speech frame can vary from a minimum of zero, to a maximum equal to the speech frame length. For a 30ms speech frame the number of adaptation cycles can vary from 0 to 240.

The AEC adaptation cycle limit is set equal to the speech frame size at the beginning of each full duplex speech session. For each speech frame that the AEC senses the CT8021 running out of processing resources, the AEC adaptation cycle limit is reduced (in steps of 16).

The host may over-ride the maximum and minimum AEC adaptation cycle limits by programming a maximum and minimum value for the AEC adaptation cycle limit. If the maximum and minimum value are programmed such that they are equal, then the AEC adaptation cycles operate with a fixed limit.

Whenever the AEC cycle adaptation limit is set to a value lower than the current frame size, the peak processing load due to the AEC adaptation (training) process is reduced. The side effect of this is that the AEC adaptation (training) time is correspondingly increased.

If the speech frame is 240 and the AEC adaptation limit is set to 120, the peak adaptation processing load is reduced by 50% and the AEC adaptation (training) time is doubled. If the limit were set to 60, then the load would be reduced by 25% and the adaptation (training) time would be quadrupled.

The following command sequences can be used to program the AEC adaptation limits:

To set the maximum AEC adaptation limit:

```
C1    E02CH
S1    E02CH
C2    00XXH
S2    00XXH
```

Where XX is the adaptation limit

To set the minimum AEC adaptation limit:

```
C1    E02DH
S1    E02DH
C2    00XXH
S2    00XXH
```

Where XX is the adaptation limit

9.15.3.1 Reading the CT8021 DSP Processor Load

The host can check if the CT8021 DSP Processor is being over-committed by polling the DSP processor load for the current speech frame.

```
Command:    511BH
Status      load-value
```

When the load value returned is 0, the DSP processor is 100% loaded.

The load value is measured by counting the number of internal codec interrupts that the DSP receives when it is in an internal idle loop waiting for availability of the next speech frame to process. When this value reaches zero, it indicates that as soon as the DSP code reaches the idle loop, it finds that the next speech frame is immediately available for processing.

The maximum value read (lowest amount of load) depends upon the speech frame size and the 8/16 bit codec configuration. In 8 bit mode, the codec generates a single interrupt for every two 8-bit codec samples. With a 8.0 KHz codec fsync rate, a codec interrupt is generated every 250 us. In 16-bit mode, the codec interrupt is generated for each sample, once every 125 us.

In 8-bit codec mode, with a 30ms speech frame, there will be 120 codec interrupts per speech frame, so that the DSP processor load can be calculated as:

$$(1 - \text{load-value}/120) * 100\%$$

In 16-bit mode with a 10ms speech frame there will be 80 codec interrupts per speech frame, so that the DSP load is:

$$(1 - \text{load-value}/80) * 100\%$$

In both cases, a load-value of zero indicates the DSP processor is 100% loaded. At this load level, the DSP may start to drop speech frames.

The load level is only meaningful in host-to-codec mode. It is not valid in host-to-host mode.

9.15.4 Adding Additional Echo Suppression

Additional echo attenuation can be achieved by increasing the level of echo **suppression** used by the CT8021. Echo suppression is different from echo cancellation. In echo cancellation, only the echo component of the input signal is affected, any input signal that originates at the microphone input is not affected. With echo suppression the entire input signal is affected, both the component due to echo and the component due to the “real” microphone input.

Echo suppression was the technique used in traditional half-duplex speakerphones. In this type of phone, only one audio signal direction is enabled at any given time. This takes advantage of the fact that in a normal conversation only one user is talking at any moment. This allows a decision to be made to attenuate one speech direction based on which speech direction is the “in-active” direction. The audio signal for the inactive speech direction is always completely muted. To completely mute one direction of speech, half-duplex speakerphones introduce greater than 60 dB of attenuation into the inactive speech path.. In a half duplex speakerphone, since only one speech direction is enabled at any time, there is no possibility of hearing an echo. The problem with this solution comes when both users attempt to talk at the same time. When this happens the direction decision making logic in the half-duplex speakerphone is unable to function correctly and the result is often that both speech directions are attenuated.

The CT8021 implements a scaleable form of Echo Suppression where the amount of attenuation added to the inactive direction can be programmed in the range 0 to -28 dB. Using 0 dB of echo suppression effectively de-activates the echo suppresser. Providing a scaleable echo suppresser attenuation allows a gradual compromise between full-duplex operation (0 dB suppression) and half-duplex operation (with -28 dB suppression). With a scaleable echo suppresser, even when a speech direction is attenuated, it is not completely muted, so it is possible to tell that speech is present.

By adding a small amount of echo suppression to the CT8021 (around 16 dB), it is possible to ensure that any residual echo not fully canceled by the AEC is removed.

The echo suppression attenuation (loop attenuation) can be programmed only after the AEC is activated and after playback and record has been started. Use the “Set Speakerphone” command described in the speakerphone section of the CT8021 data sheet.

9.15.4.1 Set Echo Suppressor Attenuation

This is the “Set Speakerphone Parameters” command used in standalone speakerphone mode. It may be used in DSVD Speakerphone Mode to set the echo suppresser attenuation. This command should be sent only AFTER starting playback and record.

Command:

1110	001	Line Volume	Speaker Volume	Loop Attenuation
(15-12)	(11-9)	(8-7)	(6-3)	(2-0)

Status:

1110	001	Line Volume	Speaker Volume	Loop Attenuation
(15-12)	(11-9)	(8-7)	(6-3)	(2-0)

Line Volume: This controls the volume transmitted to the telephone line

- 00 +8 dB
- 01 +4 dB
- 10 +0 dB (normal setting)
- 11 Mute Microphone (also resets AEC)

Speaker Volume: This controls the volume to the speakerphone loudspeaker.

- 0000 +14 dB
- 0001 +12 dB
- 0010 +10 dB
- 0011 +8 dB
- 0100 +6 dB
- 0101 +4 dB
- 0110 +2 dB
- 0111 0 dB (normal setting)
- 1000 -2 dB
- 1001 -4 dB
- 1010 -6 dB
- 1011 -8 dB
- 1100 -10 dB
- 1101 -12 dB
- 1110 -14 dB
- 1111 Speaker Mute

Loop Attenuation: Total attenuation used by receive and transmit attenuators
(Echo Suppression)

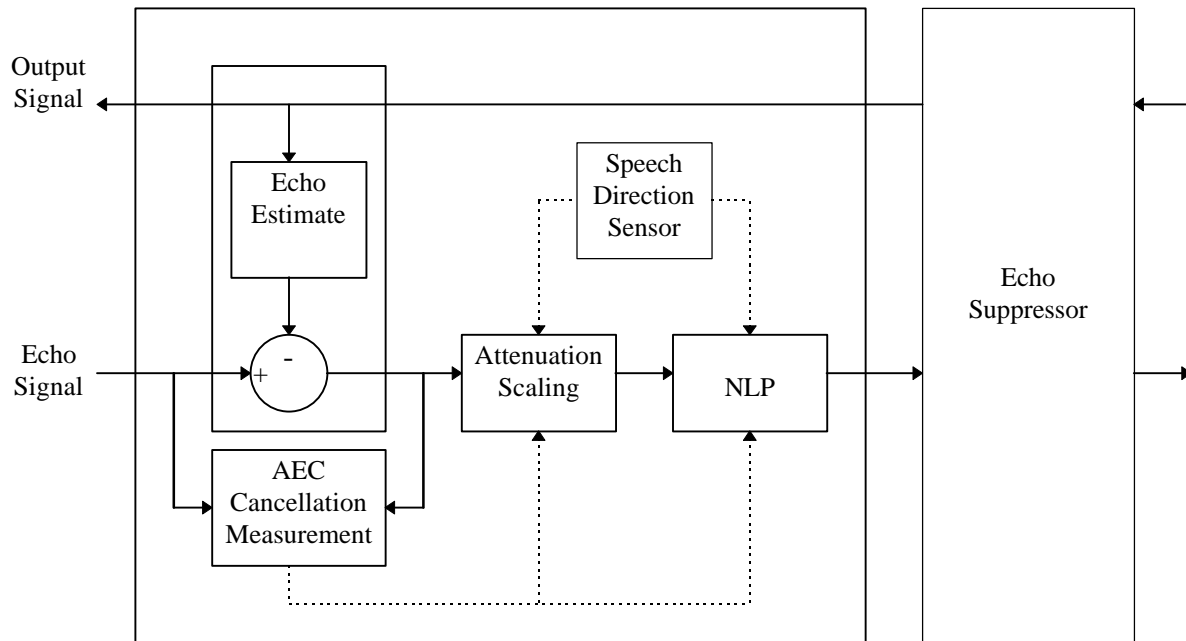
Full Duplex Operation

- 000 0 dB
- 001 -4 dB
- 010 -8 dB (default for DSVD mode)

Half Duplex Operation

- 011 -12 dB
- 100 -16 dB
- 101 -20 dB
- 110 -24 dB
- 111 -28 dB

9.15.5 Activating the Advanced Echo Canceller features in the CT8021



To increase the amount of echo attenuation provided by the AEC, the host may enable the “Advanced” AEC features of the CT8021. The Advanced-AEC is actually comprised of two independent mechanisms that can be used to reduce the residual echo left after actual cancellation.

- Non-Linear Processing (NLP)
- Attenuation Scaling or Boosting (AS/AB)

The NLP and AS/AB features are available both in DSVD and Standalone Speakerphone Mode.

9.15.5.1 Non-Linear Processing (NLP)

NLP introduces deliberate distortion into the signal so that low level signals are distorted to a greater extent than high level signals. The CT8021 offers two NLP modes: center-clipping and center-subtraction.

Center-clipping uses the following algorithm:

if input is $< +\text{Threshold}$ and input $> -\text{Threshold}$, then input is forced to zero.

Center-subtraction uses the following algorithm:

if input is $> +\text{Threshold}$ then input = input - Threshold
 else if input $< -\text{Threshold}$ the input = input + Threshold
 else input is forced to zero.

Center-clipping produces a quantization distortion effect for small amplitude signals. Center-subtraction produces an effect similar to “cross-over” distortion in a push-pull amplifier.

The NLP threshold is set dynamically relative to the output signal amplitude (note that in the CT8020 devices, the NLP threshold was set relative to the input (echo) signal). The minimum NLP threshold is controlled by setting the NLP threshold floor:

NLP threshold = output signal * scale factor (1/16)
 if NLP threshold $<$ NLP threshold floor, then NLP threshold = NLP threshold floor.

The NLP threshold floor is set using the following command:

```
C1    E011H
S1    E011H
C2    XXXXH
S2    XXXXH
```

Where XXXX is the NLP threshold floor. Recommended values are 0080H or 0100H.

The output signal scale factor can be adjusted using the additional control bits in the AEC train and lock command.

This command may be sent at any time.

To disable NLP use the following command

```
C1    E011H
S1    E011H
C2    FFFFH
S2    FFFFH
```

9.15.5.2 Attenuation Scaling (AS) & Attenuation Boost (AB)

Attenuation Scaling is a technique that allows the CT8021 to increase the amount of echo attenuation achieved when dealing with acoustic echoes in environments which prevent the AEC cancellor from fully training. One example of this is environments with high background noise levels or significant non-linear distortion in the echo path..

The dynamic Attenuation Scaling (AS) mechanism measures the amount of cancellation attenuation (in dB) produced by the AEC cancellor, scales this attenuation then adds attenuation (suppression) equal to the scaled attenuation to the cancelled echo signal.

For example, if the AEC measures 6dB of attenuation due to the cancellor, and the scale factor is set to 2, then the Attenuation scaling adds $6*2 = 12$ dB of attenuation (suppression).

The Attenuation Scaling mechanism can also be operated in fixed Attenuation Boost (AB) mode where a fixed amount of extra attenuation (suppression) is added.

To set the Attenuation Scaling mode use the following command:

```
C1    E013H
S1    E013H
C2    XXYYH
S2    XXYYH
```

For fixed scaling set XX=fixed attenuation, set YY=00. Example 0600H selects 6dB of fixed Attenuation Boost. Valid values for XX are 0,6,12,18 etc in steps of 6dB

For dynamic Attenuation Scaling set YY= attenuation scaling factor and XX=00. Example 0002H sets attenuation scaling with a scale factor of 2. Valid range for YY is 0-10.

This command may be sent at any time.

To disable AS/AB:

```
C1    E013H
S1    E013H
C2    0000H
S2    0000H
```

The NLP/AS/AB commands may be sent at any time, either before or after the Acoustic Echo Canceller is activated. These commands need only be sent once. They do not need to be sent every time the AEC is started. However, sending these commands multiple times does not cause any problems.

9.15.5.3 Dynamic Activation of NLP and Attenuation Scaling (AS)

The NLP and AS/AB features are designed to be activated only in the presence of far-end speech or echo. The CT8021 offers two mechanisms for activating the NLP/AS/AB features:

- NLP/AS/AB activation controlled by speech direction
- NLP/AS/AB activation when the measured AEC cancellation exceeds the attenuation lock threshold

See command “Setting the Lock Threshold” to set the cancellation threshold used for activating NLP/AS/AB.

See command “AEC Train and Lock” for selecting the NLP/AS/AB activation mode

9.15.5.4 Example Advanced AEC Configurations

The CT8021 offers flexibility in controlling the “advanced” AEC features (NLP/AS/AB) to allow customization of the AEC performance to meet the requirements of different echo environments.

Acoustic Echo Environments

Acoustic Echo environments tend to show the following typical properties:

- High levels of background noise
- Very short direct (microphone-speaker) echoes with short durations (< 5ms) with high amplitude echos.
- Significantly extended multi-path echo durations (> 50-100 ms) producing very low amplitude echoes
- Increased non-linearity in the echo path
- Low input user signal levels relative to speaker output levels (user is far from microphone, but speaker and microphone are close together)
- Echo path varies in real-time requiring rapid AEC training

Telephone Hybrid Echoes

Telephone Hybrid (2-to-4 wire interface) echo environments tend to show the following typical properties:

- Low noise levels
- Short echo durations (< 10 ms)
- Very little non-linear distortion in the echo path
- Input (microphone) and Output (Speaker) signal levels are approximately equal
- Echo path is constant for a single call

Large duration (> 100ms) multi-path echoes can be handled using echo cancellors with very long filter lengths (512-1024 tap filters) or using multiple sub-band (frequency division) techniques. This approach tends to require significant computation and processing resources resulting in a relatively high cost solution.

In cost sensitive applications Acoustic type echoes are more appropriately handled using techniques like NLP and AS/AB together with some degree of Echo Suppression.

High background noise levels and non-linear echo paths tend to reduce the amount of echo cancellation achieved thus requiring increased reliance on the NLP/AS/AB features to compensate.

Example Acoustic Configuration (large amplitude echoes):

C1	E011H	Set NLP threshold floor
S1	E011H	
C2	0100H	Use higher threshold floor value
S2	0100H	
C1	E013H	Set Attenuation Scaling
S1	E013H	
C2	0002H	Set attenuation scale factor to x2
S2	0002H	
C1	E012H	Train & Lock Command
S1	E012H	
C2	0023H	NLP threshold = Speaker Output >> 3 (divided by 8) NLP Mode = 1, activate NLP/AS based on measured AEC attenuation
S2	0023H	
C1	E014H	Set lock threshold (used to activate NLP/AS)
S1	E014H	
C2	0003H	Set threshold to 3dB measured attenuation
S2	0003H	

Example Telephone Echo environment (local 2-to-4 wire hybrid provides 6-12 dB or cancellation - large amplitude echoes)

C1	E011H	Set NLP threshold floor
S1	E011H	
C2	0080H	Use lower threshold floor value
S2	0080H	
C1	E013H	Set Attenuation Scaling
S1	E013H	
C2	0002H	Set attenuation scale factor to x2
S2	0002H	
C1	E012H	Train & Lock Command
S1	E012H	
C2	0024H	NLP threshold = Speaker Output >> 4 (divided by 16) NLP Mode = 1, activate NLP/AS based on measured AEC attenuation
S2	0024H	
C1	E014H	Set lock threshold (used to activate NLP/AS)
S1	E014H	
C2	0006H	Set threshold to 6dB measured attenuation
S2	0006H	

Example Telephone Echo environment (local 2-to-4 wire hybrid provides 20+ dB or cancellation - small amplitude echoes)

C1	E011H	Set NLP threshold floor
S1	E011H	
C2	0100H	Use higher threshold floor value
S2	0100H	
C1	E013H	Set Attenuation Scaling/Boost
S1	E013H	
C2	0600H	Set attenuation boost to 6dB
S2	0600H	
C1	E012H	Train & Lock Command
S1	E012H	
C2	0004H	NLP threshold = Speaker Output >> 4 (divided by 16)
		NLP Mode = 0, activate NLP/AS based on measured speech direction
S2	0004H	
C1	E014H	Set lock threshold (used to activate NLP/AS)
S1	E014H	
C2	0003H	Set threshold to 3dB measured attenuation
S2	0003H	

For low level echoes it is recommend that the NLP activation mode be set to 0, since for low level input echoes the AEC may not be able to accurately measure the achieved AEC cancellation. Activating the NLP/AS/AB using the sensed speech direction will often produce better results.

Increased echo attenuation can often be produced by increasing the NLP threshold floor and reducing the NLP Threshold Scaling. Also increase the AS or AB values.

Use of NLP activation mode 1 (use measured AEC attenuation to enable NLP/AS/AB) will tend to produce better full-duplex operation at the expense of reduced echo attenuation.

9.15.6 AEC Train and Lock

The CT8021 AEC includes features that allow it to be used to cancel fixed non-acoustic echoes. For example, the AEC can be used to cancel electrical echoes created by a 2-to-4 wire telephone hybrid.

When the AEC is used to cancel acoustic echoes, it must cope with real-time changes in the echo path of the acoustic signal. These changes may be caused by movement of the user in front of the speaker and microphone, or by actual physical movement of the speaker or microphone. As a result of this, the AEC must continuously attempt to adapt and search for the optimum echo cancellation setting. The constant search process inevitably means that the AEC may not always be operating at an optimum point.

If the AEC is required to cancel a fixed path electrical echo, it is possible to train the AEC against the fixed path and then lock the AEC training control.

The AEC can be instructed to perform an automatic train-and-lock sequence using the following command:

	Command	Status Response
C1	E012	E012
C2	4004	4004

If the AEC is already in a locked state when this command is set, the AEC is unlocked allowing re-training.

To de-activate the train and lock feature, use the following command sequence:

	Command	Status Response
C1	E012	E012
C2	0004	0004

To force the AEC into a locked state, use the following sequence:

	Command	Status Response
C1	E012	E012
C2	C004	C004

See also Detailed Train & Lock Command Description.

9.15.6.1 Detailed Train & Lock Command Description

C1 E012

S1 E012

C2/S2: (The default C2 parameter command value for this command is 0004H)

Force Lock State	Enable Lock Mechanism	MicroPhone Input Pre-Scale	0	Mic Vox Sensitivity	Far-End Vox Sensitivity
(15)	(14)	(13-12)	(11)	(10)	(9)

0	NLP/AS/AB Control	AEC Train Speed	AEC NLP Mode	Force NLP	NLP Threshold Scaling
(8)	(7)	(6)	(5)	(4)	(3-0)

Force Lock State	Set this bit to 1 to force the AEC into the locked state
Enable Lock Mechanism	Set this bit to 1 to enable the AEC train and lock mechanism
Microphone Input Pre-Scale	00: Sets normal microphone input scaling 01: Sets x2 (+6dB) microphone input scaling 10: Sets x4 (+12dB) microphone input scaling 11: Sets x8 (+18dB) microphone input scaling Sets the microphone signal input scaling, this modifies the microphone input signal prior to all other signal processing
Mic Vox Sensitivity	0: Set normal microphone VOX sensitivity 1: Reduce microphone VOX sensitivity by 12 dB This affects the speech direction decision algorithm
Far-End Vox Sensitivity	0: Set normal far-end (line-side) VOX sensitivity 1: Reduce far-end VOX sensitivity by 12 dB This affects the speech direction decision algorithm
NLP/AS/AB Control	0: Activate NLP/AS/AB is speech direction detected is from far-end OR if measured AEC cancellation exceeds threshold programmed for LOCK 1: Activate NLP/AS/AB ONLY if measured AEC cancellation exceeds threshold programmed for LOCK
AEC Train Speed	0: Adapt AEC at normal rate 1: Adapt AEC at half-rate This reduces the AEC processing load at the expense of increasing the training time. This control acts independently of the mechanism for limiting the number of AEC training cycles per speech frame. The AEC Train Speed control affects both DSVD speakerphone mode and Standalone Speakerphone mode.
Force NLP	Force NLP processing to be always activated. This over-rides the NLP/AS/AB control.
NLP Threshold Scaling	This field sets the scaling factor (SF) used to calculate the NLP center clipping threshold: Threshold = Speaker Output signal >> SF The default SF is 4, so the threshold is calculated by right-shift-by-4 (division by 16)

9.15.6.2 AEC Locked State

In the “locked” state, the AEC will not undergo further updates of its internal coefficients unless the amount of instantaneous attenuation measured is greater than the Lock Threshold. This places a significant restriction on the AEC adaptation control algorithm. It is designed to prevent a “hunting” effect where the AEC attempts to adapt to low level noise instead of a real echo signal.

Use of this feature is not recommended for echo paths that can vary in real-time. If the echo path changes while the AEC is in the locked state, the amount of attenuation achieved may never exceed the Lock Threshold. In this situation, the AEC will never update its internal coefficients.

9.15.6.3 Setting the Lock Threshold

The attenuation threshold at which the AEC enters the Locked state is programmable. Use the following command sequence:

	Command	Status Response
C1	E014	E014
C2	XXXX	XXXX

where XXXX is the attenuation threshold in dB.

For example the default threshold is -12 dB and so XXXX = FFF4 (-12 in hexadecimal form)

9.15.6.4 Setting the Un-Lock Threshold

The AEC can be programmed to automatically un-lock if the instantaneous measured attenuation indicates that the AEC is not cancelling the echo. This may happen if the echo path is changed.

If the echo path is changed significantly, it is possible that the echo estimate calculated by the AEC is so badly in error that the AEC creates an effect where it INCREASES the echo signal level instead of reducing it. In this situation, the AEC can be programmed to automatically un-lock when amplification of the echo signal is detected:

	Command	Status Response
C1	E01B	E01B
C2	XXXX	XXXX

where XXXX is the attenuation threshold in dB.

For example, the default threshold is +3dB and so XXXX=0003.

Note that when measuring the AEC attenuation, a negative dB value represents a reduction in the output signal level (attenuation), whereas a positive dB value represents an increase in the output signal level (amplification).

9.15.6.5 Freezing the AEC

In the Locked state, the AEC coefficients are updated if the measured attenuation exceeds the Lock Threshold. To cause an complete freeze of the AEC, where no updating of the AEC coefficients take place, simply program the Lock and Un-Lock Thresholds to large values and force the AEC into the locked state.

For example:

	Command	Status Response	
C1	E01B	E01B	program the un-lock threshold to a large positive value
C2	7FFF	7FFF	
C1	E014	E014	program the lock threshold to a large negative value
C2	FF00	FF00	
C1	E012	E012	Force the AEC into the locked state
C2	C004	C004	

9.15.7 Controlling the Automatic Loop Adjustment Attenuation

When the AEC is operating under normal conditions, it constantly monitors the effect of the echo cancellation process. The intent of the AEC is to reduce the amplitude of any echo signals passing through the AEC. When the AEC passes through a signal that originates at the near-end (a signal that is due to the local user talking, not due to an echo of the speaker output), the signal should pass through without any change in amplitude. Under both these sets of conditions, the output signal level from the AEC block should never be larger than the input signal level.

If the AEC detects that the output level of the AEC is greater than the input level of the AEC, this indicates a fault condition. This state may arise if the acoustic echo path is rapidly changed, for example by suddenly moving the microphone to a position immediately in front of the speaker. Under these circumstances, the AEC by default will increase the amount of echo suppression in use to attempt to prevent positive feedback in the audio loop (howling).

9.15.7.1 Programming the Loop Adjustment Threshold

The Loop Adjustment Threshold at which the additional loop attenuation is activated can be programmed by the host:

	Command	Status Response
C1	E01A	E01A
C2	XXXX	XXXX

where XXXX is the Loop Adjustment Threshold in dB.

For example, the default threshold is +1 dB, so XXXX=1dB

The automatic loop adjustment feature can be disabled by programming the Loop Adjustment Threshold to a large positive value, e.g. XXXX=7FFFH or by programming the amount of attenuation added to zero (see below).

Also the automatic loop adjustment feature can be disabled by using the Speakerphone Configuration Command described in the Standalone Speakerphone section.

Note that when using the AEC concurrently with speech compression modes, this particular command can only be sent AFTER enabling the AEC and entering the playback and record states.

9.15.7.2 Programming the Loop Adjustment Attenuation

When the AEC detects error-state amplification in excess of the Loop Adjustment Threshold, by default it adds 12dB of echo suppression in addition to the echo suppression (loop attenuation) selected by the Set Echo Suppressor Attenuation command. The maximum amount of total echo suppression in operation is limited to 28dB.

The amount of additional echo suppression introduced can be programmed by the host using the following command:

	Command	Status Response
C1	E015	E015
C2	000X	000X

Where X is between 0 and 7 and represents units of 4dB steps.

For example the default attenuation added is 12 dB, so X=3

When the AEC detects that the error conditions that caused the additional echo suppression to be added no longer exist, the additional echo suppression (loop attenuation) is removed automatically.

9.15.8 Reading the instantaneous AEC attenuation

The host can read the approximate instantaneous attenuation as measured internally by the AEC using the following command:

Command E00A
 Status response XXXX

Where XXXX is the approximate attenuation in dB. This can be a negative value which indicates the current instantaneous echo signal is being attenuated, or a small positive value indicating that either the AEC is not trained, or that no input or echo signal is present. If the attenuation value read is a positive value larger than +6dB, this indicates that the AEC is operating under adverse or error conditons.

The value read can only show a significant negative value (indicating a significant amount of attenuation) when a speaker output signal is present which causes a large input echo. When no input signal is present, the amount of attenuation achieved is indeterminate.

Note that the reported attenuation is the difference in signal level between the microphone input signal and the echo-cancelled microphone input signal. It is NOT the difference between the speaker output and the echo-cancelled microphone input signal. Any attenuation provided by the external echo path is NOT included in the reported attenuation. Example: if the echo path returns an echo -20 dB relative to the output signal, and the AEC succeeds in cancelling a further 6dB of echo, then the reported cancellation is -6dB (not -26dB which is the overall cancellation attenuation). Also note that the reported AEC cancellation does NOT include the effect of the NLP/AS/AB feature). The reported cancellation is the cancellation effect due to the CT8021 AEC only.

9.15.9 Saving and Restoring the AEC and EEC coefficients

The current AEC and EEC echo estimation filter coefficients can be saved and restored into the external data sram attached to the CT8021 device. Coefficients saved in the external srams are never lost except in the case of a power on-off cycle. The coefficients saved are not affected by a device reset. The coefficients will be lost if the external sram self test command is used.

The working filter coefficients of the AEC and EEC are reset each time the echo canceller is restarted. In the case of the AEC, the filter coefficients are reset by a AEC-off, AEC-on sequence.

9.15.9.1 Save AEC coefficients

Command	E016
Status	E016

9.15.9.2 Restore AEC Coefficients

Command	E017
Status	E017

9.15.9.3 Save EEC coefficients

Command	E018
Status	E018

9.15.9.4 Restore EEC coefficients

Command	E019
Status	E019

9.15.10 Evaluating Echo Canceller Performance

The CT8021 AEC uses a pair of VOX detectors to detect the presence of speech and control the operation of the echo canceller and echo suppressor. These detectors respond to short term rate of change in signal energy. This enables the detectors to distinguish between real voice activity and back ground noise. As a result of this, it is not appropriate to test the CT8021 AEC performance using continuous sine waves. The CT8021 should only be tested using real speech or alternatively using pulsed (dual) sinewaves or pulsed white noise. The pulse duration should typically be less than 500 ms.

9.16 SPEECH FRAME INTERRUPT

The CT8021 can provide a speech frame-by-frame interrupt from the TX READY and RX READY signals of the Hardware Status Register. For added flexibility two additional mechanism are provided for generating interrupts derived from the frame rate.

9.16.1 Frame Interrupt via the FR pin

For compatibility with the CT8015, the CT8021 can provide a Frame Interrupt signal via the FR/ pin GPIO5. For the CT8021, by default, this pin is not enabled, and GPIO5 is configured as an input, see section on GPIO pins for instructions to enable the FR pin functionality.

If the FR pin functionality is enabled, then when record, playback or full-duplex speech mode is active, the FR/ pin is asserted (driven low) every frame period. This makes the FR/ signal useful as a frame interrupt to the host. Since FR/ is asserted every frame period, the host is guaranteed that a new record frame will be available in the receive frame buffer, or a space will be available for a new playback frame in the transmit frame buffer, each time the FR/ is asserted. The host can clear the FR/ signal using the CLEAR FR command:

The FR/ pin is implemented using the GPIO 5 pin. This is an optional signal provided for CT8015 TrueSpeech DSVD Co-Processor compatibility. If this pin is not required, the GPIO 5 pin is available for general purpose use. Instead of using the FR/ pin, the Host can use the IRQN interrupt pin and select that it be driven from the transmit/receive data buffer TX or RX READY signal by programming the Hardware Control Register. This will then provide an interrupt each time a compressed data frame is ready (30 ms). The interrupt is cleared by filling or emptying the data buffer as appropriate.

The following commands are available for controlling the FR/ pin: Enable, Disable, Clear and Read.

Enable FR:

command: 5141H
 status: 5141H

by default, the FR pin is enabled.

Disable FR

command: 5142H
 status: 5142H

Clear FR

command (5140H)

0101 0001 0100 0000

(15-0)

status:

XXXX XXXX XX	FR	X XXXX
--------------	----	--------

(15-6)

(5)

(4-0)

The state of the FR/ pin (before it is cleared) is returned in bit-5. If the FR pin was asserted prior to being cleared, FR = 0, otherwise FR = 1. The FR/ is active low (when asserted, the pin is low).

Read FR

command (5143H)

0101 0001 0100 0011

(15-0)

Status = YYXXH

YYYY YYYY	GPIO-7	GPIO-6	FRN	X	GPIO-3	GPIO-2	GPIO-1	DATA FLAG
(15-8)	(7)	(6)	(5)	(4)	(3)	(2)	(1)	(0)

The state of the FR/ pin is returned in bit-5. If the FR pin is asserted, FR = 0, otherwise FR = 1. The FR/ is active low (when asserted, the pin is low). The state of the FR/ pin is not affected by the Read FR command. See also section on reading and writing the GPIO pins.

9.16.2 Frame Interrupt via Aux Software Status Register

The CT8021 can also be programmed to generate a frame interrupt every N frames via the Aux Software Status Register (ASSR).

Set ASSR update rate:

Command C1 = 5145H
Status S1 = 5145H

Command C2 = N (update ASSR every N frames)
Status S2 = N

If RECORD or PLAYBACK is active, the CT8021 will write its internal frame count to the ASSR every N frames. The Host can then program the CT8021 Hardware Control Register to generate an interrupt on the Aux Status Ready condition. The Host clears the interrupt by reading the ASSR. Setting N to zero, disables update of the ASSR. This is the default setting.

The Host can also read the current value of the internal frame count at any time using the command:

Read Frame Count:

Command = 5144H
Status = current frame count.

This frame count is incremented every frame period (30ms) if either RECORD or PLAYBACK is active. The frame counter is cleared on entry to either RECORD or PLAYBACK from IDLE mode.

9.17 DEVICE SELF-TEST

9.17.1 Check Internal Program ROM Integrity

Command = 3000H

0011	00	00 0000 000	0
(15-12)	(11-10)	(9-1)	(0)

Status

0011	X	DATA ROM STATUS	PROGRAM STATUS	ROM	X XXXX XXXX
(15-12)	(11)	(10)	(9)		(8-0)

PROGRAM ROM STATUS: Set to 0 to indicate internal program rom checksum is valid.
 Set to 1 to indicate internal program rom failure.

DATA ROM STATUS: Set to 0 to indicate internal data rom checksum is valid.
 Set to 1 to indicate internal data rom failure.

X: Reserved bits

ERRATA: The CT8021A11AQC contains a bug which causes the device to always report an internal prom failure.

9.17.2 Test External Data SRAM

Command = 3001-3004H

0011	00	00 0000 0	size
(15-12)	(11-10)	(9-3)	(2-0)

Status

0011	SRAM TEST	XXX XXXX XXXX
(15-12)	(11)	(10-0)

SRAM TEST Set to 0 to indicate external data SRAM test passed OK
 Set to 1 to indicate external data SRAM failure.

X: Reserved bits

size 1 tests 8K bytes / 4K words
 2 tests 32K bytes / 16K words
 3 tests 64K bytes / 32K words
 4 tests 16K bytes / 8K words

After running the external SRAM test, the CT8021 device should be reset. This will re-initialize the external SRAM contents to their normal operational values. Running the SRAM test, destroys all the data values held in the external SRAM. SRAM tests start at external memory address 4000H.

9.17.3 External Data SRAM Configuration

After reset, the CT8021 automatically probes the external memory to detect if the memory configuration is 8 or 16 bits wide. For normal operation of the CT8021, the external memory must be 16 bits wide. The memory width setting detected can be over-riden with the following commands.

Command = 0821H to select a 8-bit wide memory (15 ns)

Command = 0801H to select a 16-bit wide memory (15 ns)

This command does not generate a status response. When this command is received, it halts all current operations and performs an internal reset. After issuing this command, the host should wait for a period of 10 ms and then write the IDLE command (0000H) to the CT8021. Once this has been completed the Host should then re-configure the codec configuration before proceeding with further operations.

9.18 DEVICE IDENTIFICATION

9.18.1 Get Device Identification code

Command = 3400H

0011	01	00 0000 0000
(15-12)	(11-10)	(9-0)

Status

1000	0000	0010	0000
(15-12)	(11-8)	(8-4)	(3-0)

The status reply is the device code as a hexadecimal number i.e. CT8021 = 8021H.

9.18.2 Get Device Version (revision) code

Command = 3800H

0011	10	00 0000 0000
(15-12)	(11-10)	(9-0)

Status

XXXX	XXXX	XXXX	XXXX
(15-12)	(11-8)	(8-4)	(3-0)

XXXX = 4 bit hexadecimal code.

The status reply is the device version code as a hexadecimal number i.e. version 1.15 = 0115H.

9.19 TONE GENERATION in IDLE PLAYBACK & RECORD MODES

This command is used to generate tones in IDLE, PLAYBACK and RECORD modes only. To generate DTMF tones during Standalone Speakerphone mode, please refer to the Speakerphone commands. When tone generation is completed in IDLE mode, the Host should use the IDLE command to return the CT8021 to the IDLE state.

Generate Tone Command:

0110	000	MASTER GAIN	TONE INDEX
(15-12)	(11-9)	(8-5)	(4-0)

TONE INDEX: Selects the tone to be generated.

MASTER GAIN:

Modifies (reduces) the level of the tone to be generated with respect to the level programmed into the tone table. The level value in this field is added to the level value programmed into the tone table. The value resulting is used to select the output level (see table below). For example, if the tone table contains the level 5, and the master gain is set to level 8, the tone is actually output with level value of $5 + 8 = 13$. The result of the addition is limited ≤ 15 , so if the tone level is 8, and the master gain is 9, the result $8 + 9 = 17$, is limited to 15 and the tone is output using level value 15. An alternate view of this control is note that the output level is reduced by $MASTER\ GAIN \times 2dB$, with a minimum output level of -24 dB.

Status:

0110	0000 000	MASTER GAIN	TONE INDEX
(15-12)	(11-5)	(4)	(3-0)

In IDLE mode, with default output codec routing (= 00) tone is output via codec 0 only (telephone line side). To output tone to codec1, change the output codec routing in the codec configuration command.

In Playback or Record mode, tone is output to the default codec (normally codec 1) unless over-riden by the codec configuration command

Tone generation is possible in two mode: synchronous or asynchronous.

9.19.1 Stop Tone Generation

To terminate tone generation in IDLE mode, use the IDLE command 0000H.

To terminate tone generation during playback/record mode use the following command:

Stop Tone:

command: 6E0AH
status: 6E0AH

Note that when generating tones during playback/record, it is recommended that tone generation be configured for asynchronous mode. If tones are generated during playback, the tone output replaces the normal output audio data generated by the playback data. Playback data is consumed at the normal rate even when tone generation is active.

9.19.2 Synchronous mode

In this mode (default) the duration of the tone is determined by the number of frame periods for which the command is repeated. Synchronous mode is the default mode and is compatible with the tone generation of the CT8005 TrueSpeech Messaging Co-Processor.

The CT8021 creates the output waveform in frames of 30 ms each (240 samples at 8.0 KHz). The CT8021 will check for a host command at the end of each frame after updating the status response and asserting Status Ready in the Hardware Status Register. If the host repeats the original command, the status reply is repeated, and tone generation is continuous. If the host changes the tone parameters, the waveform output is changed accordingly. If the command is the IDLE command, tone generation is terminated. Note that the CT8021 requires that the IDLE command is used to correctly terminate tone generation. If the host does not issue any command, the CT8021 continues to check for a command every 30 ms and continues to generate the tone specified by the last command.

To generate a tone of 90 ms duration in this mode, the host performs the following sequence:

```
write tone command 60XXH
wait for Status Ready    (30 ms delay)
repeat tone command
wait for Status Ready    (30 ms delay)
repeat tone command
wait for Status Ready    (30 ms delay)
write idle command (0000H)
```

To generate a silence gap of 90 ms, use the above sequence with the tone command 6000H. This generates a special “silence” tone.

9.19.3 Asynchronous mode

In this mode, the on and off period for the tone, is pre-programmed by the host, and then complete generation of the tone (on and off periods) is executed with a single command. The host may poll the CT8021 during execution of the command to determine when generation of the complete tone has finished.

Additional Tone Mode Commands

SET SYNCHRONOUS (default, CT8005 compatible) TONE MODE

command: 6E00H
status: 6E00H

SET ASYNCHRONOUS TONE MODE

command: 6E01H
status: 6E01H

GET TONE MODE

command: 6E02H

if in synchronous mode
status: 0000H

if in asynchronous mode
status: 00001H

This command can be used to determine the current tone mode.

The following group of commands are only effective in ASYNCHRONOUS tone mode:

SET TONE ON PERIOD

command C1: 6E04H
status S1: 6E04H
command C2: XXXXH = data value
status S2: XXXXH = data value echoed

SET TONE OFF PERIOD

command C1: 6E05H
status S1: 6E05H
command C2: XXXXH = data value
status S2: XXXXH = data value echoed

GET TONE ON PERIOD

command: 6E06H
 status: XXXXH = data value

GET TONE OFF PERIOD

command: 6E07H
 status: XXXXH = data value

The units used for setting and reading the tone on and off period are 250 microseconds. By default the tone on and tone off periods are both set to 0192H (400 decimal) which is equivalent to a period of 100 ms, (i.e. $400 * 250$ microseconds = 100 milliseconds).

GET TONE STATE

command: 6E03H

if tone generation is still in progress then

status: 0001H

otherwise

status: 0000H

9.19.4 Tone Generation in Asynchronous mode:

To generate a tone on = 90 ms followed by a tone off = 90 ms use the following sequence:

write SET ASYNCHRONOUS TONE mode command
 wait for Status Ready in Hardware Status Register and read status response.
 (this needs to be done only once)

write SET TONE ON PERIOD command
 wait for Status Ready and read status response.
 write tone on period value command = 0168H (0168H = 360 decimal, $360 * 250$ us = 90 ms)
 wait for Status Ready and read status response.
 write SET TONE OFF PERIOD command
 wait for Status Ready and read status response.
 write tone off period value command = 0168H
 wait for Status Ready and read status response.
 (this needs to be done only once, unless the on/off period requires further change)

write tone command 60XXH
 wait for Status Ready and read response

poll device using GET TONE STATE command
 wait for Status Ready and check status reply to determine if tone has completed yet
 (repeat this last step as often as required).

Once tone generation has been completed, the Host should issue the IDLE command.

9.19.5 Tone Level Table

Level Value (index)	Codec output level (dB relative to 0.707 Vrms)	Digital output level dBm0
0	+6	+1.5
1	+4	-0.5
2	+2	-2.5
3	0	-4.5
4	-2	-6.5
5	-4	-8.5
6	-6	-10.5
7	-8	-12.5
8	-10	-14.5
9	-12	-16.5
10	-14	-18.5
11	-16	-20.5
12	-18	-22.5
13	-20	-24.5
14	-22	-26.5
15	-24	-28.5

This table shows the actual tone output level as a function of the level value, calculated by adding the level value contained in the tone table to the level value specified in the MASTER GAIN field of the Generate Tone Command.

The codec output levels given are those produced by the VFR0 output pin of a Texas Instruments TP3054 mu-law codec when connected to the CT8021 (600 ohm load). Note that the TP3054 output is driven by a power amplifier with approximately 4 dB of gain. A digital signal at 0 dBm0 (0.775 Vrms) will produce an output signal of 1.3 Vrms (+4.5 dBm-600 ohms) at the codec output.

9.19.6 New Tone Initialization

Command (C1):

0110	10	00000	TONE INDEX
(15-12)	(11-10)	(9-5)	(4-0)

Status (S1):

0110	10	00000	TONE INDEX
(15-12)	(11-10)	(9-5)	(4-0)

Command (C2):

0110	0000	Gain 1	Gain 0
(15-12)	(11-8)	(7-4)	(3-0)

Status (S2):

0110	0000	Gain 1	Gain 0
(15-12)	(11-8)	(7-4)	(3-0)

Command (C3):

Frequency Factor for Tone 0
(15-0)

Status (S3)

Frequency Factor for Tone 0
(15-0)

Command (C4):

Frequency Factor for Tone 1
(15-0)

Status (S4)

Frequency Factor for Tone 1
(15-0)

TONE INDEX: index in range 1 to 31, index 0 is reserved for a “silent” tone.

Gain1,Gain0: level value from tone level table

Frequency Factor: $32767 \times \cosine(2 \times \pi \times F/8000)$
 where $\pi = 3.14159$
 F is the desired frequency
 8000 is the sample rate at which the codec is operating
 Note : single frequencies can be generated by setting either frequency factor to 7FFFH

9.19.7 Default (Power-On) Tone Table Contents

Tone Index	Freq. 0	Gain 0	Freq. 1	Gain 1	DTMF Digit
0	No Tone				
1	697	1	1209	0	1
2	697	1	1336	0	2
3	697	1	1477	0	3
4	770	1	1209	0	4
5	770	1	1336	0	5
6	770	1	1477	0	6
7	852	1	1209	0	7
8	852	1	1336	0	8
9	852	1	1477	0	9
10	941	1	1209	0	* (Star)
11	941	1	1336	0	0
12	941	1	1477	0	# (Pound)
13	800	1			
14	1000	1			
15	1250	1			
16	950	1			
17	1100	1			
18	1400	1			
19	1500	1			
20	1600	1			
21	1800	1			
22	2100	1			
23	1300	1			
24	2450	1			
25	350	1	440	0	
26	440	1	480	0	
27	480	1	620	0	
28	697	1	1633	0	A
29	770	1	1633	0	B
30	852	1	1633	0	C
31	941	1	1633	0	D

Note that index zero is reserved for a silent tone. This can be used to generate gaps or pauses between tone, for example when generating a series of DTMF digits.

See Tone Level Table for gain value equivalent levels.

9.20 CT8021 Line Monitor Commands

The CT8021 Line Monitor command can be used to access 5 detectors (1 DTMF and 4 Call Progress). These detectors can be individually enabled or disabled. When enabled, the detectors run whenever the CT8021 is in any of the following modes: Line Monitor, Record, Playback or Full-Duplex Speech (record+playback) mode. By default all the detectors are disabled.

9.20.1 Line Monitor Command

Command (7000H)

0111	0000	0000	0000
(15-12)	(11-8)	(7-4)	(3-0)

Synchronous Mode Status:

0111	F3	F2	F2	F0	0	PEAK	DTMF VALID	CPF VALID	DTMF DIGIT
(15-12)	(11)	(10)	(9)	(8)	(7)	(6)	(5)	(4)	(3-0)

Asynchronous Mode Status

Is 4-bits Frame Count	F3	F2	F2	F0	0	PEAK	DTMF VALID	CPF VALID	DTMF DIGIT
(15-12)	(11)	(10)	(9)	(8)	(7)	(6)	(5)	(4)	(3-0)

This command activates Line Monitor Mode. To terminate Line Monitor Mode use the IDLE command (0000H). If asynchronous mode is selected (see below) it can also be used during record/playback to poll the state of the detectors.

DTMF DIGIT	The DTMF digit value (only valid if DTMF VALID is set) : 1-9 1-9 0 0BH * 0AH # 0CH A 0DH B 0EH C 0FH D 0
CPF VALID	Indicates that at least 1 of the 4 detector outputs F0-F3 is set (logical OR of F0-F3)
DTMF VALID	Set when a valid DTMF digit is detected
PEAK	Only valid when RECORD is active. This bit has the same meaning as the PEAK bit in the Record S2 status response
F0-F3	Individual filter output valid bits for filters 0 -3
Is 4-bits Frame Count	Only present when asynchronous mode is selected. This field can be used by the host to track the number of frames processed by the CT8021 since the last Line Monitor Status.

9.20.2 Synchronous and Asynchronous Monitor Mode

By default the Line Monitor Command operates in synchronous mode (CT8005 compatible). In this mode the Line Monitor Status response is delayed until completion of processing of the next 30 ms frame. In this mode the Line Monitor Status response can be used as a “frame clock” since the status response is synchronized to the frame period. For continuous monitoring the Host should issue a new Line Monitor Command each frame period (each time the previous Line Monitor Status response is received). Note that in Synchronous Mode, after issuing the Line Monitor command, the Host must wait for the (delayed) status response before it can issue another command.

In Asynchronous Mode, the status response to the Line Monitor Command is immediate. The status reported represents the most recent 30 ms frame processed. To assist the Host in correlating the time of the response with other system events, the ms 4-bits of the response contain the ls 4-bits of the CT8021's internal frame counter. This frame counter increments once every 30 ms frame period. This count value can be used by the Host to measure the duration of a valid filter output.

Asynchronous mode should be selected if the Host needs to poll the detectors during record/playback.

to select Synchronous mode, use the command:

Set Synchronous Mode	7E00H
Status Response	7E00H

to select Asynchronous mode, use the command:

Set Asynchronous Mode	7E01H
Status Response	7E01H

to read the current mode, use the command:

Read Sync Mode	7E02H
Status Response	0000H = Synchronous Mode 0001H = Asynchronous Mode

9.20.3 Enabling and Disabling the DTMF Detector

To Enable the DTMF Detector, use the command

Enable DTMF Detector	5151H
Status Response	5151H

To Disable the DTMF Detector, use the command

Disable DTMF Detector	5150H
Status Response	5150H

9.20.4 Controlling the Call Progress Tone Filters F0-F3

The CT8021 includes 4 Call Progress Filters. Each filter can be addressed using the select filter command to set the current value of the CT8021's filter ID parameter (CPFID):

Select Filter Command (5154H - 5157H)

0101	0001	0101	01	SELECT
(15-12)	(11-8)	(7-4)	(3-2)	(1-0)

Status Response (5154H - 5157H)

0101	0001	0101	01	SELECT
(15-12)	(11-8)	(7-4)	(3-2)	(1-0)

SELECT = CPFID

00	Filter F0
01	Filter F1
10	Filter F2
11	Filter F3

Once Selected, the following commands can be used to control the filter:

Filter Enable (5153H)

Filter Disable (5152H)

Status response:

0000	000	DTMF Detector Enabled	0000	F3	F2	F1	F0
(15-12)	(11-9)	(8)	(7-4)	(3)	(2)	(1)	(0)

The F3-F0 and DTMF Detector Enabled bits are set to show which filters are enabled.

To enable all 4 filters the following command sequence can be used

Command	Status Response	Action
5154H	5154H	Select Filter F0
5153H	0001H	Filter F0 is enabled
5155H	5155H	Select Filter F1
5153H	0003H	Filters F1+F0 are enabled
5156H	5156H	Select Filter F2
5153H	0007H	Filters F2+F1+F0 are enabled
5157H	5157H	Select Filter F3
5153H	000FH	Filters F3+F2+F1+F0 are enabled

9.20.5 Reading the Filter Energy Output

The actual Filter Energy output values can be read using the following command (516XH):

0101	0001	0110	FILTER	INDEX
(15-12)	(11-8)	(7-4)	(3-2)	(1-0)

FILTER

00	F0
01	F1
10	F2
11	F3

INDEX

00	Broadband Energy ls word (bits 15-0)
01	Broadband Energy ms word (bits 31-16)
10	In Band Energy ls word (bits 15-0)
11	In Band Energy ms word (bits 31-16)

The status response contains the 16-bit value selected by INDEX.

Each filter provides two 32-bit energy parameters:

The broadband energy of the un-filtered input signal.

The in-band energy at the filter output.

The energy value is calculated over a 240 sample (30 ms) input frame:

$$\text{Energy} = 1/32 * \sum (x/4) * (x/4)$$

If more than one filter is enabled, the broadband energy from each enabled filter will be the same.

9.20.6 Selecting the Filter Characteristics

The CT8021 includes Filter Parameters Sets for 8 different call progress tone (CPT) fixed filters plus an additional set of 4 Filter Parameters that may be programmed by the host. Each of the four filters F0-F1 can be mapped onto any of the 12 available filter parameter sets. The first 4 filter parameter sets are stored in the CT8021s internal RAM and may be re-programmed by the host.

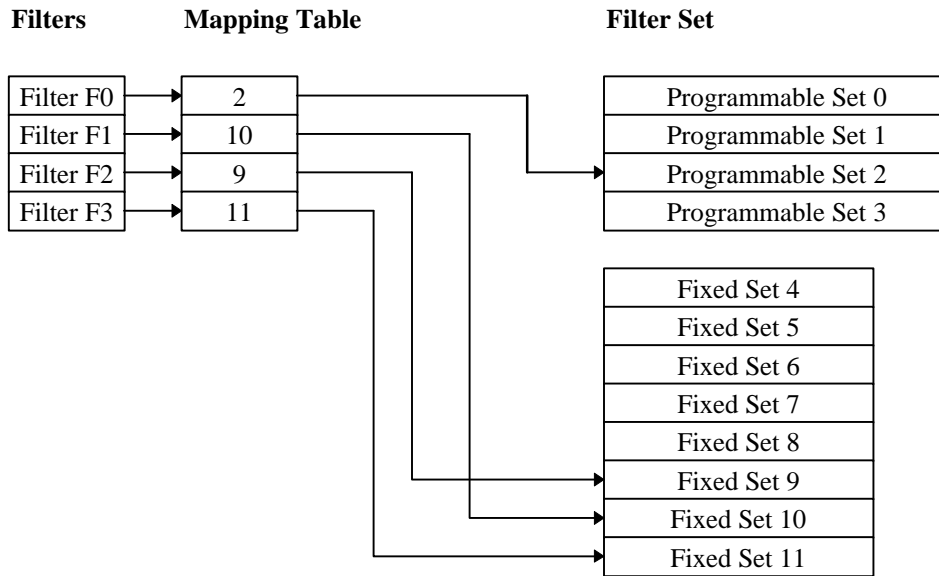


Figure 9-4 Filter Mapping

The following table shows the default contents of the Filter Parameter Sets

Filter Parameter Set	
0	Programmable Filter Set 0 (by default this is loaded with Fixed Set 4)
1	Programmable Filter Set 1 (by default this is loaded with Fixed Set 5)
2	Programmable Filter Set 2 (by default this is loaded with Fixed Set 6)
3	Programmable Filter Set 3 (by default this is loaded with Fixed Set 7)
4	Fixed Filter Set 4 (wide CPT filter 300Hz - 640 Hz)
5	Fixed Filter Set 5 (narrow CPT filter 300Hz - 500 Hz)
6	Fixed Filter Set 6 (fax calling tone CNG 1100Hz +/- 5%)
7	Fixed Filter Set 7 (350 Hz +/- 5%)
8	Fixed Filter Set 8 (400 Hz +/- 5%)
9	Fixed Filter Set 9 (480 Hz +/- 5%)
10	Fixed Filter Set 10 (440 Hz +/- 5%)
11	Fixed Filter Set 11 (620 Hz +/- 5%)

The Filter Parameter Set used by each of the Filters F0-F3 may be programmed by the Host. The filter set used by each filter is controlled by the CT8021's Filter Parameter Set Mapping Table. The default Filter Set selection is shown below:

Default Filter Parameter Set Mapping Table

FILTER	Table Value	Filter Parameter Set used by Filter
F0	0	Programmable Filter Set 0 (equivalent to Fixed Set 4 after reset)
F1	1	Programmable Filter Set 1 (equivalent to Fixed Set 5 after reset)
F2	2	Programmable Filter Set 2 (equivalent to Fixed Set 6 after reset)
F3	3	Programmable Filter Set 3 (equivalent to Fixed Set 7 after reset)

So after reset the following filter parameters are selected

FILTER	Filter Parameter Set used by Filter
F0	wide CPT filter 300Hz - 640 Hz
F1	narrow CPT filter 300 Hz - 500 Hz
F2	fax calling tone CNG filter 1100 Hz +/- 5%
F3	350 Hz +/- 5%

At reset, the filter parameters in Programmable Filter Sets 0-3 are loaded from the parameters in Fixed Filter Sets 4-7.

After reset, the Programmable Filter Sets may be re-loaded from Fixed Filter Sets using the following command:

Restore Initial Filter Parameters Command 515DH
 Status Response 515DH

9.20.7 Changing the Filter Parameter Set Mapping Table

The contents of the Filter Parameter Set Mapping Table can be changed using the following:

Command (517XH):

0101	0001	0111	MAPPING VALUE
(15-12)	(11-8)	(7-4)	(3-0)

Status Response:

0101	0001	0111	MAPPING VALUE
(15-12)	(11-8)	(7-4)	(3-0)

Where MAPPING VALUE is in the range 0-11 (decimal) corresponding to Filter Parameter Sets 0-11.

To change a Filter Parameter Set Mapping Table entry:

1. First select the entry to be changed (F0-F3) using the Select Filter Command 5144H-5147H
2. Change the table entry using the command 517XH.

The following Mapping Table can be created using the command sequence below:

FILTER	Table Value	Filter Parameter Set used by Filter
F0	7	Set 7 = 350 Hz
F1	10	Set 9 = 440 Hz
F2	9	Set 10 = 480 Hz
F3	11	Set 11 = 620 Hz

Command Sequence:

```

5154H    Select Filter F0
5177H    Select Filter Set 7
5155H    Select Filter F1
517AH    Select Filter Set 10
5156H    Select Filter F2
5179H    Select Filter Set 9
5157H    Select Filter F3
517BH    Select Filter Set 11

```

9.20.8 Filter Detection Algorithm

The Filter Detection Algorithm has three stages:

1. The In-Band filter energy for each 15 ms half-frame (120 samples) is compared against a minimum threshold. By default, this is set to approximately -37 dBm0. If the energy is less than the threshold a “no detect” decision is made for the half-frame.
2. The In-Band filter energy for each 15 ms half-frame (120 samples) is compared against the Broad-Band energy. If the In-Band energy is greater than $21/32 * \text{Broad-Band energy}$ then a “detect” decision is made for the half-frame.
3. A filter detect history is constructed every frame period which covers the last 4 frames, or 8 half-frames. If a detect decision was made in at least 5 of the last 8 half-frames, then the corresponding Filter Output Valid bit is set in the Line Monitor Status response.

Note: dBm0 scale - For a mu-law or a-law codec, the full scale input signal range of the codec corresponds to approximately 3.14 dBm (0 dBm is 1 mW in 600 ohms). For signal level compatibility across digital systems this signal level is referred to as 3.14 dBm0. When using a 16-bit linear codec, the full scale range of this codec is also assumed to be 3.14 dBm0.

9.20.9 Reading the Filter History Register

The Filter History Register for each Filter F0-F3 can be read using the command:

Command (515CH)

0101	0001	0101	1100
(15-12)	(11-8)	(7-4)	(3-0)

Status:

...	n-12	n-11	n-10	n-9	n-8	n-7	n-6	n-5	n-4	n-3	n-2	n-1	n
(15-13)	(12)	(11)	(10)	(9)	(8)	(7)	(6)	(5)	(4)	(3)	(2)	(1)	(0)

The status response shows the detection decision for the most recent 15 ms half-frame “n” and up to the last 15 half-frames “n-1” to “n-15”. By default, the CT8021 only checks from n to n-7, with n-8 to n-15 ignored. The default detection algorithm looks for five “1” bits in the range “n” to “n-7”.

To read the History Register for Filter F1, use the following command sequence:

Command	Status Response	Action
5155H	5155H	Select Filter F1
515CH	XXXXH	Filter F1 History Register is read

9.20.10 Reading the Frame Counter

The CT8021’s internal 16-bit frame counter can be read using the command

Command (5144H):

0101	0001	0100	0100
(15-12)	(11-8)	(7-4)	(3-0)

Status:

16-bit Frame Counter
(15-0)

9.20.11 Programming the CT8021 Filters

Filter Parameter Sets 0-3 can be programmed by the Host. The Filter Parameter Set to be programmed is selected by the CPFID parameter set using the Filter Select command 5154H-5157H.

Each Filter Parameter Set contains 19 words of parameter data.

To start programming of the Filter Parameter Set, use the command:

Command (**515BH**):

0101	0001	0101	1011
(15-12)	(11-8)	(7-4)	(3-0)

Status (515BH)

0101	0001	0101	1011
(15-12)	(11-8)	(7-4)	(3-0)

After writing this command, the next **19** words written to the Software Control Register (command register) will be written to the Filter Set selected.

Command (XXXXH):

Filter Parameter Set Data Word
(15-0)

Status (XXXXH):

Filter Parameter Set Data Word
(15-0)

CT8020 Compatability Note: Note that the earlier **CT8020A11BQC** (firmware revision 0112) only allowed downloading of the first **15** words of the filter parameter set. This was implemented using the **515EH** command. The new **515BH** command implemented in the CT8021 and the **CT8020D11AQC** (firmware revision 0114) allows downloading of all **19** parameters in the filter parameter set. The CT8021 continues to support the **515EH** command for purposes of backwards compatability with the older CT8020A11BQC version.

The filter data consists of 15 words of filter coefficients for the sixth order filter (3 x 5 words), plus 4 additional control parameters.

Each filter is a sixth order filter constructed from 3 cascaded second order stages.

The following shows the parameter order and example values for Fixed Filter Set 4 (300-640 Hz) :

Word	Example Value	Description
0	28967	A1 (Q14) = 1.768028
1	-28798	A2 (Q15) = -0.878854
2	-1988	B2 (Q15) = -0.060682
3	0	B1 (Q14) = 0
4	1988	B0 (Q15) = 0.060682
5	28017	A1 (Q14) = 1.710051
6	-30963	A2 (Q15) = -0.944905
7	12605	B2 (Q15) = 0.384666
8	0	B1 (Q14) = -0.620271
9	12607	B0 (Q15) = 0.384720
10	31398	A1 (Q14) = 1.916387
11	-31859	A2 (Q15) = -0.972276
12	30541	B2 (Q15) = 0.932033
13	-30020	B1 (Q14) = -1.832294
14	30542	B0 (Q15) = 0.932074
15	7	Internal Filter numeric scaling
16	159	Minimum in-band energy threshold
17	21	21/32 in-band to broad-band ratio parameter
18	0FF5H	shift-mask = 0FFH, bit count=5 for History test

Q15 format uses 32768 to represent 1.000

Q14 format uses 16384 to represent 1.000

The Internal filter numeric scaling is set to compensate for the internal gain within the filter in order to prevent numeric overflow. The higher the internal gain (Q factor), the lower the numeric scaling value required. For a narrow band pass filter (high Q) this should be set to around 5, for a wide band pass filter this should be set to around 7. The value used is best determined by experimentation, by looking for non linearity in the filter output value as a function of input signal frequency and level.

The Minimum in-band energy is squared by the CT8021 and used to compare against the in-band filter energy.

The in-band to broad-band ratio parameter is used to create a scaling ratio by dividing by 32 and using the resulting fractional multiplier to scale the broad-band energy for comparison with the in-band energy.

The final parameter is split into two parts. The ms 12-bits create an AND mask to control how many History Register bits are counted. The ls 4 bits provide the bit count threshold 0-15 for counting the "1" bits in the History Register.

The filter coefficients themselves have to be created using special Digital Filter Design Software. This software is available from a number of sources specializing in DSP design software. Filter design packages vary in the nomenclature used to label the A and B parameters as well the sign value +/- of the constants. The best approach to take when designing a new filter, is to first create a design for the 300-640 Hz filter defined above. The filter coefficients generated can usually be approximately compared against the example above to determine if any of the parameters need to be re-ordered or have their sign +/- inverted.

The example filter above (300-640 Hz) was designed with the following parameters:

type	6th order elliptic
lower stop-band cut-off	250 Hz
lower pass-band cut-off	300 Hz
upper pass-band cut-off	640 Hz
upper stop-band cut-off	800 Hz
sample rate	8000 Hz
pass band ripple	0.1 (1 dB)
stop band ripple	0.05 (-26 dB stop band attenuation)

9.20.12 Generating Interrupts from Line Monitor Events

The CT8021 can generate line monitor event interrupts to the host controller via the auxillary software status register (ASSR). The host can set an interrupt mask corresponding to the status response bits for the normal line monitor status response.

Set line monitor status interrupt mask:

C1: 5148H
 S1: 5148H

C2/S2 interrupt mask:

000X	F3	F2	F2	F0	0	0	DTMF VALID	CPF VALID	DTMF DIGIT
(15-12)	(11)	(10)	(9)	(8)	(7)	(6)	(5)	(4)	(3-0)

Setting a mask bit to 1 will cause the ASSR to be updated with the current line monitor status if the corresponding bit in the line monitor status changes state. If the X bit is set to 1, the ASSR will be automatically updated every speech frame period even if there is no change to the line monitor status value.

By default the line monitor status interrupt mask is set to zero.

To generate a host interrupt from the line monitor status, the host should set the Aux Status Update IE bit in the Hardware Control Register. The interrupt may be cleared by reading the ASSR.

For interrupt driven line monitor status updates, the line monitor feature should be operated in asynchronous mode.

For example: To generate an ASSR update on detection of any DTMF digit use the following command sequence:

C1 5148H
 S1 5148H
 C2 0020H
 S2 0020H

9.20.13 Operating the Line Monitor detectors on the outgoing playback audio stream

By default, the Line Monitor detectors operate on the incoming (record) audio stream. However the CT8021 can be configured so that the detectors operate on the outgoing (playback) audio stream. This enables the detector to detect tones and DTMF digits in the decompressed (playback) audio.

To set the Line Monitor detector direction use the following command.

```
C1: 5149H
S1: 5149H
C2: direction
S2: direction
```

Set direction=0 for normal operation

Set direction=1 for tone detection on the outgoing (playback) audio stream

9.21 STANDALONE SPEAKERPHONE

9.21.1 Enter Standalone Speakerphone Mode Command:

This command is not available in DSVD Speakerphone mode.

1110	01	01	Line Sensitivity	Microphone Sensitivity	Priority	0
(15-12)	(11-10)	(9-8)	(7-5)	(4-2)	(1)	(0)

Status:

1110	01	01	Line Sensitivity	Microphone Sensitivity	Priority	0
(15-12)	(11-10)	(9-8)	(7-5)	(4-2)	(1)	(0)

Line Sensitivity:

000	2 dB
001	4 dB
010	6 dB
011	8 dB (normal setting)
100	10 dB
101	12 dB
110	14 dB
111	16 dB

Microphone Sensitivity:

000	2 dB
001	4 dB
010	6 dB (normal setting)
011	8 dB
100	10 dB
101	12 dB
110	14 dB
111	16 dB

The line and microphone sensitivity determine the signal-to-noise ratio (SNR) at which the voice detectors for the transmit and receive directions will be activated. The voice detectors are used in half-duplex mode to determine which direction is talking, and which direction should be attenuated.

Priority:

0	both directions have equal priority
1	microphone side (near end) has priority

The priority control is used to determine the behavior of the system when both directions are silent. In the equal priority (or no priority) mode, the loop attenuation is distributed equally between the receive and transmit directions. If the priority bit is set, the line side is given all the attenuation and the microphone (near end) as priority.

Before entering Standalone Speakerphone mode, the Codec Configuration must be programmed with Output Codec set to 00 and Input Codec set to 0.

9.21.2 Get Speakerphone Status Command:

This command may be used in DSVD Speakerphone mode.

1110	000	0 0000 0000
(15-12)	(11-9)	(8-0)

Status:

1110	000	0	Direction	0	Current Loop Attenuation
(15-12)	(11-9)	(8)	(7-4)	(3)	(2-0)

Current Loop Attenuation: This indicates the amount of loop attenuation currently in use. See Set Speakerphone Parameters command below for the format of this field.

Direction: This indicates which direction (microphone/speaker) is considered active by the CT8021 Speakerphone control logic. This is a 4 bit field used to represent the number range from +7 to -8. The most positive values indicated that the signal from the microphone side is dominant (microphone speaking). The most negative values indicate that the signal from the line side is dominant (telephone line speaking). Values in the center of the range can either indicate a transition from one side to the other, or if the value stays at approximately the zero position, indicates that neither side is talking.

+7	0111	
+6	0110	Microphone side is speaking.
+5	0101	
+4	0100	
+3	0011	Transition to/from microphone active
+2	0010	
+1	0001	
0	0000	Both sides silent (constant energy) or transition
-1	1111	
-2	1110	
-3	1101	
-4	1100	Transition to/from line side active
-5	1011	
-6	1010	
-7	1001	Line side is speaking
-8	1000	

9.21.3 Set Speakerphone Parameters Command:

This command may be used in DSVD Speakerphone Mode

1110	001	Line Volume	Speaker Volume	Loop Attenuation
(15-12)	(11-9)	(8-7)	(6-3)	(2-0)

Status:

1110	001	Line Volume	Speaker Volume	Loop Attenuation
(15-12)	(11-9)	(8-7)	(6-3)	(2-0)

Line Volume: This controls the volume transmitted to the telephone line

- 00 +8 dB
- 01 +4 dB
- 10 +0 dB (normal setting)
- 11 Mute Microphone

Speaker Volume: This controls the volume to the speakerphone loudspeaker.

- 0000 +14 dB
- 0001 +12 dB
- 0010 +10 dB
- 0011 +8 dB
- 0100 +6 dB
- 0101 +4 dB
- 0110 +2 dB
- 0111 0 dB (normal setting)
- 1000 -2 dB
- 1001 -4 dB
- 1010 -6 dB
- 1011 -8 dB
- 1100 -10 dB
- 1101 -12 dB
- 1110 -14 dB
- 1111 Speaker Mute

Loop Attenuation: Total attenuation used by receive and transmit attenuators

Full Duplex Operation

- 000 0 dB
- 001 -4 dB
- 010 -8 dB

Half Duplex Operation

- 011 -12 dB
- 100 -16 dB (startup condition)
- 101 -20 dB

Abnormal Conditions

- 110 -24 dB
- 111 -28 dB

9.21.4 SpeakerPhone Configuration Command:

This command may be used in DSVD Speakerphone mode provided that Line Training = 00.

1110	010	0 0000	Line Training	Automatic Loop Adjustment
(15-12)	(11-9)	(8-4)	(3-2)	(1-0)

Status:

1110	010	0 0000	Line Training	Automatic Loop Adjustment
(15-12)	(11-9)	(8-4)	(3-2)	(1-0)

Automatic Loop Adjustment:

- 00: Loop Attenuation is not automatically adjusted
- 01: Increase loop attenuation by 12 dB on detection of abnormal conditions. Maintain modified attenuation until a new command is received from host.
- 10: Increase loop attenuation by 12 dB on detection of abnormal conditions. Attempt to restore normal loop attenuation when possible. This is the preferred (normal) setting.
- 11: Reserved

Line Training:

- 00: Do not modify line-side (electrical) echo canceller settings.
- 01: Send training pulse to line to set (electrical) echo canceller coefficients.
- 10: Reserved
- 11: Reserved

When the Acoustic Echo Canceller (AEC) is active in DSVD speakerphone mode, the Line Training field should always be set to 00.

9.21.5 Generate Tone (in speakerphone mode):

This command is not available in DSVD Speakerphone mode.

1110	100	Master Gain	Tone Index
(15-12)	(11-9)	(8-5)	(4-0)

Status

1110	100	Master Gain	Tone Index
(15-12)	(11-9)	(8-5)	(4-0)

See Generate Tone command (mode 6) for usage.
 Each command/status interaction generates tone for 30 ms.
 This command is not available in DSVD speakerphone mode.

Use Tone Index #0 to generate silence between dialed digits. During tone generation, the speech path between codec 0 and codec 1 is disabled. To re-enable this path at the end of DTMF dialing, the host should issue command E000H (Get SpeakerPhone Status). This is used to inform the CT8021 that dialing is complete.

9.21.6 Get Electrical Echo Canceller Quality Factor Command:

This command is not available in DSVD Speakerphone mode.

1110	101	0 0000 000	HIWORD
(15-12)	(11-9)	(8-1)	(0)

Status:

Electrical Echo Canceller Quality Factor
--

This command returns either the low (HIWORD = 0) order or high order (HIWORD = 1) 16 bits of the electrical (line-side) echo canceled 32-bit quality factor. The quality factor is a measure of how well the training pulse succeeded in configuring the electrical echo canceller coefficients.

During training, the echo canceller sends the training pulse and receives an echo of the training pulse back from the line. The echo canceller coefficients are adjusted to achieve maximum cancellation of the training pulse echo.

If no echo occurs (for example, if a 4-wire telephone connection is used), all the echo canceller coefficients are set to zero, no cancellation is required (“perfect” cancellation is achieved - there is no reflected signal), and the quality factor is zero in all 32 bits.

If the training pulse is sent while there is a high level incoming signal (dial tone for example), the echo canceller is unable to cancel the incoming signal (training pulse echo + dial tone) and the quality factor becomes very large (> 100000H). This indicates that training failed and that full duplex operation of the speakerphone can not be activated (loop attenuation must be greater than or equal to 12 dB).

If the training pulse echo is successfully canceled, the quality factor should be less than 10000H (all high order bits zero). Under these circumstances, full duplex speakerphone operation may be selected (loop attenuation less than 12 dB).

If the training pulse has not been sent, the quality factor will be set to FFFFFFFFH.

The numbers given here are approximate, and depend to a great extent on the design of the external hardware. The user is expected to experiment and discover suitable threshold values appropriate to the users hardware environment.

Note that this command reports the current value of the Electrical Echo Canceller (EEC) Quality Factor. Since training of the EEC takes several milliseconds, it is possible to read this value before training has completed. In this case the Quality Factor from the state prior to training will be read. The host should allow around 50 ms to elapse between sending the command to train the EEC and attempting to read the new Quality Factor. During the training period, the host should not issue any additional commands to the CT8021, as this will delay or disrupt the training process.

See Appendix A for additional information.

9.22 HOST-TO-HOST DATA COMPRESSION AND DECOMPRESSION

The CT8021 can be operated as a full or half-duplex TrueSpeech compression engine in a codec-less configuration. In this situation, the CT8021 routes the uncompressed data stream to the host instead of to the codec.

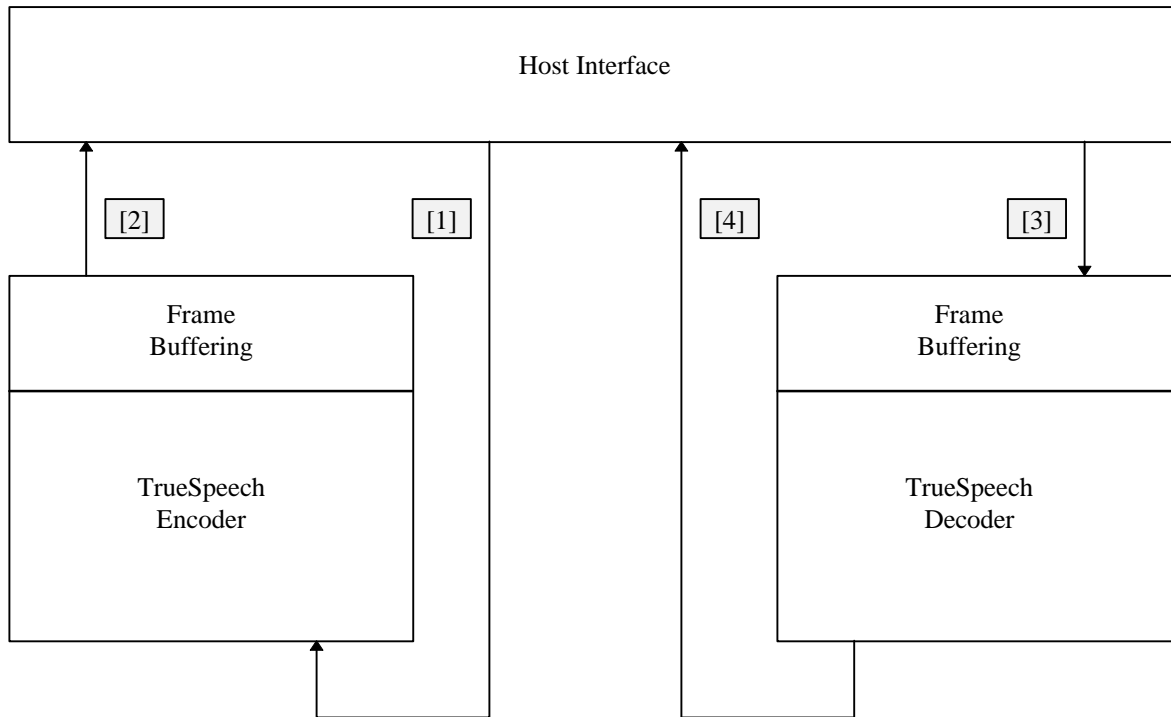


Figure 9-5 Host-to-Host Operation

The Host-to-Host compression/decompression protocol is structured so that the four data exchange operations are independent. The four data exchange operations that take place are:

1. Un-compressed 16-bit linear data transfer from host to CT8021, denoted using [1] in the description below.
2. Compressed TrueSpeech data transfer from CT8021 to host, denoted using [2] in the description below.
3. Compressed TrueSpeech data transfer from host to CT8021, denoted using [3] in the description below.
4. Un-Compressed 16-bit linear data transfer from CT8021 to host, denoted using [4] in the description below.

The compressed data exchanges, [2] and [3] to and from the CT8021 internal Frame Buffers take place using the same transfer protocol as used for record and playback (TFR Mode = 00). The Host should ensure that the AEC (Acoustic Echo Canceller) state is set to OFF for Host-to-Host operation.

9.22.1 Host-to-Host compression

Host-to-Host compression from the Host viewpoint is similar to the record process. The protocol used to transfer compressed data frames from the CT8021 to the host [2] is the same as that used for recording in DATA sync mode with TFR Mode = 00. However, instead of using the Record C1 command to start the process, the Host-to-Host compression C1 command is used. The compression C1 command is also used to transfer the actual uncompressed data frames to the CT8021 [1].

The following command is used to start Host-to-Host compression:

Compress Command C1 (= FXXXH):

1111 01	RECMODE	000 0	TFR_MODE	DATA
(15-10)	(9-7)	(6-3)	(2-1)	(0)

Status Response S1:

1111 01	RECMODE	000 0	TFR_MODE	READY
(15-10)	(9-7)	(6-3)	(2-1)	(0)

9.22.1.1 Recording (Compression) Modes

RECMODE	Recording Modes
000	TrueSpeech at 8.5, 6.3, 5.3, 4.8 or 4.1 Kbits/sec
001	Reserved
010	Reserved
011	Reserved
100	64 Kbit/sec A-law / Mu-law PCM (120 words per 30ms data frame)
101	128 Kbit/sec 16-bit linear (240 words per 30ms data frame)
110	64 Kbits/sec 8-bit signed linear (120 words per 30 ms data frame)
111	64 Kbits/sec WSS 8-bit unsigned linear (120 words per 30 ms data frame)

See Record section for more details on RECMODE and TFR_MODE.

DATA

In the first C1 command (used to start compression) the DATA flag is set by the host to indicate whether the host wishes to immediately transfer the first un-compressed speech frame [1] to the CT8021 for compression.

In subsequent compression C1 commands, the DATA flag is ignored by the CT8021. In this case the CT8021 always assumes that the host will be ready to transfer uncompressed data [1].

READY

If the READY flag is set in the status response, the CT8021 expects that immediately following the S1 response, the host will transfer 240 words (480 bytes) of uncompressed data. The READY flag will always be set in the first S1 response if the host set the DATA flag in the first C1 command. When the Host receives a status response with the READY flag set, it MUST immediately commence the transfer of uncompressed speech data.

Host-to-Host compression data transfer [1] and [2]

Host starts compression process [1] by issuing compression C1 command, and waits for Status Ready in the Hardware Status Register to be asserted.

CT8021 acknowledges start of compression with compression S1 response.

IF the READY flag is set in status response S1:

Host transfers uncompressed data by writing the (240) uncompressed data words to the Software Control register and waiting for STATUS READY to be set after each word written. Each time STATUS READY is set, the host must read the Software Status Register in order to clear STATUS READY. The actual value read from the Software Status Register is the same as (echo of) the last data word written.

The compression C1, and S1 status response - READY flag are used to control subsequent un-compressed speech data transfers[1] to the CT8021.

Once compression has been started, the Host transfers compressed data from the CT8021 [2] using the normal record data transfer protocol (TFR Mode = 00 or 11).

The record protocol used to transfer compressed speech data frames from the CT8021 also supports the BUFFER MONITORING, FRAME CREATE, FRAME DELETE, and RECORD STOP commands.

Host-to-Host compression can be terminated either using the RECORD STOP command or the IDLE command. One of these commands should be issued during the command-status sequence in place of the compression C1 command.

Compression READY polling

Instead of using the compression S1 status READY response to determine if the CT8021 is ready to accept more uncompressed speech data, the Host can poll the READY state using the following command:

```
command:      5119H
status:       0000H  if CT8021 is ready to accept more uncompressed data.
              0001H  if CT8021 is NOT ready to accept more uncompressed data.
```

This command is only valid if compression mode is active.

The advantage this command offers over simple use of the compression C1-S1 command-status, is that it allows the host to query the READY state without immediately committing to performing the data transfer. With the compression C1-S1 command-status, if the READY bit in the S1 status response is set, the host MUST immediately transfer data.

9.22.2 Host-to-Host decompression

Host-to-Host decompression is from the Host viewpoint is similar to the playback process. The protocol used to transfer compressed data frames from the Host to the CT8021 [3] is the same as that used for playback in DATA sync mode. However, instead of using the Playback C1 command to start the process, the Host-to-Host decompression C1 command is used. Note that in the case of decompression, the host must still send the playback C2 and C3 commands after sending the decompression C1 command to start the decompression process. The decompression C1 command replaces the playback C1 command, but the playback C2 and C3 commands are still required when starting decompression. The decompression C1 command is also used to transfer the actual uncompressed data frames from the CT8021 [1].

The following command is used to start Host-to-Host decompression:

Decompress Command C1:

1111 11	PLAYMODE	000 0	TFR_MODE	0
(15-10)	(9-7)	(6-3)	(2-1)	(0)

Status Response S1:

1111 11	PLAYMODE	000 0	TFR_MODE	READY
(15-10)	(9-7)	(6-3)	(2-1)	(0)

When starting decompression this command must then be followed immediately by the playback C2 and C3 commands. Note that when the first decompression C1 command is sent to start decompression, the READY flag in the status response will not be set, since at this point the CT8021 will have no decompressed data available.

To Start Playback:

- Playback C1
- Playback C2
- Playback C3

To Start Decompression:

- Decompress C1
- Playback C2
- Playback C3

9.22.2.1 Playback (Decompression) Modes

PLAYMODE	Playback Modes
000	TrueSpeech at 8.5, 6.3, 5.3, 4.8, 4.1 Kbits/sec
001	Reserved
010	64 Kbit/sec A-law / Mu-law PCM (120 words per 30ms data frame)
011	Reserved
100	64 Kbits/sec 8-bit signed linear (120 words per 30 ms data frame)
101	64 Kbits/sec 8-bit unsigned linear (120 words per 30 ms data frame)
110	128 Kbits/sec 16-bit signed linear (240 words per 30 ms data frame)
111	Reserved

See Playback Mode section for more details on PLAYMODE and TFR_MODE.

READY

If the READY flag is set in the status response, the CT8021 expects that immediately following the S1 response, the host will transfer 240 words (480 bytes) of uncompressed data. When the Host receives a status response with the READY flag set, it MUST immediately commence the transfer of uncompressed speech data.

Host-to-Host decompression data transfer [3] and [4]

Host starts decompression process [3] by issuing decompression C1 command, and waits for Status Ready bit in the Hardware Status Register to be asserted.

CT8021 acknowledges start of decompression with decompression S1 response.

Host reads S1 status, Status Ready is cleared.

(Since no data has been decompressed at this time the READY flag will not be set)

Host sends playback C2 and C3 commands

Once decompression has been started, the Host transfers compressed data to the CT8021 [3] using the normal playback data transfer protocol.

Subsequent de-compressed speech is transferred using the decompression C1 command. The S1 status - READY flag is used to control the transfers :

Host sends decompression C1 command

CT8021 returns S1 status response including READY flag.

IF the READY flag is set:

Host transfers de-compressed data by writing the dummy de-compression command F000H to the Software Control register and then waiting for STATUS READY to be set. Each time STATUS READY is set, the host must read the Software Status Register in order to clear STATUS READY. The data read from the Software Status Register is the de-compressed data. The host writes the F000H command 240 times in order to read 240 data words from the Software Status Register.

The playback protocol used to transfer compressed speech data frames to the CT8021 also supports the BUFFER MONITORING, FRAME CREATE, FRAME DELETE, and PLAYBACK STOP commands.

Host-to-Host decompression can be terminated either using the PLAYBACK STOP command or the IDLE command. One of these commands should be issued during the command-status sequence in place of the playback C2 command.

Decompression READY polling

Instead of using the decompression S1 status READY response to determine if the CT8021 is ready with more uncompressed speech data, the Host can poll the READY state using the following command:

```
command:      5219H
status:       0000H  if CT8021 is ready to provide more uncompressed data.
              0001H  if CT8021 is NOT ready to provide more uncompressed data.
```

This command is only valid if decompression mode is active.

The advantage this command offers over simple use of the decompression C1-S1 command-status, is that it allows the host to query the READY state without immediately committing to performing the data

transfer. With the decompression C1-S1 command-status, if the READY bit in the S1 status response is set, the host MUST immediately transfer data.

9.22.3 Full Duplex Host-to-Host Compression-Decompression

The following command sequence is recommended for performing full duplex Host-to-Host Compression-Decompression. The sequence is described in summary form, for details on the individual steps listed, see the detailed descriptions above.

Start decompression using the decompression C1 command and playback C2 and C3 commands.
Start compression using the C1 command with DATA set to zero.

Perform data transfers using the following command loop:

Check if CT8021 is ready for more playback data using the playback BUFFER MONITORING command.

If the playback buffer is empty:

Transfer compressed speech data to CT8021.

Check if the CT8021 has decompressed data ready using the decompression READY polling command.

If the CT8021 has decompressed data ready (status = 0000H):

Transfer uncompressed speech data from CT8021.

Check if the CT8021 is ready to accept data for compression using the compression READY polling command.

If the CT8021 is ready to accept more data for compression (status = 0000H):

Transfer uncompressed speech data to CT8021.

Check if the CT8021 has record data available using the record BUFFER MONITORING command.

If the record buffer is not empty:

Transfer compressed speech data from CT8021.

Repeat the above loop until all data has been processed.

Terminate the process after reading the last uncompressed playback data and reading the last compressed record data from the CT8021 by issuing the IDLE command

9.23 TEST MODES

The CT8021 includes test modes which are useful in verifying device and system operation.

9.23.1 Test Mode 1: count mode

Command: 511FH
Status: 511FH

In this test mode, data received from the external codec by the record channel is discarded and replaced by an (8-bit) incrementing count. This is intended for use in A-law / Mu-law speech mode where the 8-bit codec receive data is passed directly to the host.

9.23.2 Test Mode 2: digital milliwatt

Command: 511DH
Status: 511DH

In this test mode, data received from the external mu-law codec by the record channel is discarded and replaced by the CCITT G.711 A-law / Mu-law digital milliwatt (0 dBm0) code sequence. This is a sequence of 8 repeating bytes (A-law / Mu-law samples) representing a 1 KHz sinewave at 0 dBm0.

9.23.3 Test Mode 3: loopback

Command: 511CH
Status: 511CH

In this test mode, data received from the external codec by the record channel is discarded and replaced by the codec output data from the transmit (playback) channel. The playback channel is not affected by this mode of operation. The playback speech data is also transmitted to the output codec.

Note that receive codec to transmit codec loopback can be performed by the host with the CT8021 operating in normal mode. To do this, the host simply copies mu-law frames from the record (receive) channel back to the playback (transmit) channel.

9.23.4 Exit Test Mode

Command: 511EH
Status: 511EH

This command is used to exit test mode and return the record and playback channels to their normal operating mode.

9.23.5 Codec Loopback

Independently of the test modes described above, the CT8021 also supports codec-input to codec-output loopback. In this mode, every frame period (30 ms), the CT8021 will loop-back 240 samples from the input codec back to the output codec. To enter this mode, the CT8021 must be in the IDLE state. To activate this mode use the command:

```
Command      = 4000H
Status       = 4000H
```

This loop-back mode can be used in conjunction with Test Mode 2 above to generate a digital milliwatt output signal to the codec.

Also the Codec Configuration command can be used to select the input and output codecs used for the loop-back operation. For example the following codec routing are possible:

Input Signal From	Output Signal To	Codec Configuration Command	
		OUTPUT CODEC	INPUT CODEC
Codec 1	Codec 1	00	0
Codec 1	Codec 0	10	0
Codec 1	Codec 0 & 1	11	0
Codec 0	Codec 0	10	1
Codec 0	Codec 1	00	1
Codec 0	Codec 0 & 1	11	1

The INPUT CODEC field operates to exchange (swap) the input signals from Codec 0 and Codec 1.

Use of this form of loopback operation requires that the CT8021 is in the IDLE state when the command to activate loopback is issued. This loopback configuration can not be used in conjunction with other CT8021 operational modes (e.g. playback and record). To perform loop-back type operations in record or playback, refer to the section on Codec Monitoring.

To exit loopback mode, the Host should issue the IDLE command (0000H).

9.23.6 Codec (Audio) Monitoring

Codec Monitoring provides a mechanism which allows the input signal at one codec to be monitored (echoed) to the output of either or both codecs. This mechanism operates independently of the primary CT8021 operational modes (e.g. record). This allows the Host to implement “call-screening” type functions where a recording being made via the input from codec 0 can be monitored by the user via the output of codec 1.

Command = CE1XH
 Status = CE1XH

where X selects the action from the following table

X	Codec Routing Configuration
0	Normal operation This setting must be selected for standalone speakerphone operation
1	Codec 0 input echoed to Codec 0 output and Codec 1 input echoed to Codec 1 output
2	Codec 0 input echoed to Codec 1 output and Codec 1 input echoed to Codec 0 output
3	Codec 0 input echoed to Codec 0 and Codec 1 output
4	Codec 1 input echoed to Codec 0 and Codec 1 output

The codec routing described here is over-ridden by the values selected for INPUT CODEC and OUTPUT CODEC in the Codec Configuration command. For example, setting CODEC INPUT to ‘1’ will reverse the Codec input entries in the table above.

For any operations involving playback or tone generation, X should be set to zero.

9.23.7 Speech Algorithm Testing

The CT8021 supports the following commands to support compliance verification of certain ITU standards based speech algorithms. Note that these are special test mode commands only. Please refer to the appropriate ITU speech coder standard documentation for further information:

Set Post Filter Control

C1: 5416H
S1: 5416H
C2: 00XXH
C3: 00XXH

XX=00 Post Filter On (normal, default operating state)
XX=01 Post Filter Off for G.728
XX=10H Post Filter Off for G.723.1

Set High Pass Filter

C1: 5417H
S1: 5417H
C2: 00XXH
C3: 00XXH

XX=00 High Pass Filter On (normal, default operating state)
XX=01 High Pass Filter Off (G.728 and G.723.1)

9.24 Power Save Modes

9.24.1 CT8021 Stop Mode

The CT8021 can be placed into Stop Mode using the command:

Command: 0FF0H

Since the CT8021 halts all internal operations in response to this command, there is no status response.

The CT8021 can only be released from Stop mode using the external RESET pin. After reset, the CT8021 performs internal initializations and then asserts the Control Ready bit in the Hardware Status Register. The Host must wait for the Control Ready bit to be set before issuing the first command. Following reset, the Host must re-program the codec configuration before attempting any other operations.

In stop mode, the codec signals FSYNC and SCLK are also stopped.

9.24.2 Stop Codec Mode

Command: 0FF2H

Status: 0FF2H

In Stop Codec mode, the CT8021 FSYNC and SCLK output pins are forced low. Holding FSYNC low and stopping SCLK, will cause some CODECs to enter power down mode. See your codec data sheet to see if this feature is supported.

To achieve maximum system power saving, the codec interface should be placed in stop mode before placing the CT8021 itself in stop mode. Following this sequence will guarantee that FSYNC halts in the low state.

Note that if pull-up/down resistors are connected to the FSYNC and SCLK pins to provide proper input voltages during reset, that use of pull-down resistors will result in less power consumption when the codec is in stop mode. However pull-up resistors maybe used if desired.

9.24.3 Re-Start Codec

The codec can be restarted using the command:

Command: 0FF3H

Status: 0FF3H

9.24.4 CT8021 Slow Down (Power Save) Modes

The CT8021 can be programmed by the host into slow down mode. In slow down mode, the clock rate of the internal DSP core is reduced to allow just sufficient processing power for the operation selected. For example, when the CT8021 is performing Full Duplex TrueSpeech 6.3 operation in conjunction with AEC operation, the DSP core must be run at full speed. But when the CT8021 is being used in pass through mode to playback un-compressed 16-bit linear data, the DSP core can be operated at greatly reduced speed.

To enter slow-down mode use the following command:

Command = 0FEXH where X is the internal DSP clock division factor
 Status = 0FEXH

The internal DSP core clock is divided by (X+1), such that:

- X=0 selects full speed
- X=1 selects half speed
- X=2 selects one third speed
- X=3 selects one quarter speed
- X=F selects one sixteenth speed

The internal DSP clock frequency can be observed on the CLKOUT pin.

The following table gives an approximate guide to the processing speed required for some example operations:

Operation	X value	DSP internal speed factor	Typical Power Consumption
Full Duplex TrueSpeech 6.3-4.1 + AEC	0	1.0	230 mA
Full Duplex TrueSpeech 8.5 + AEC	1	1/2	120 mA
Playback only (all modes)	3	1/5	60 mA
Full Duplex un-compressed speech	3	1/5	60 mA
Line Monitoring (tone detection)	3	1/5	60 mA
IDLE	F	1/16	35 mA

AEC = Acoustic Echo Canceller

9.24.5 Disable CLKOUT

To further conserve power and reduce EMI emissions, the CLKOUT signal of the CT8021 may be disabled with the following command:

Disable CLKOUT command: 0FF6H
 status: 0FF6H

Enable CLKOUT command: 0FF7H
 status: 0FF7H

9.25 General Purpose Input Output pins (GPIO)

The CT8021 provides 8 GPIO pins, GPIO-0 to GPIO-7. Two of these pins may be configured for system use to provide CT8015 compatability as DATAFLAGN amd Frame Interrupt FR.

GPIO-0 may be used to provide the DATAFLAGN output signal
 GPIO-5 may be used to provide the frame interrupt output signal

After reset, all pins are configured as inputs and are available for general purpose use by the host controller.

9.25.1 Configure GPIO system use

GPIO pins are configured for system use by setting the system/user control mask:

Command = 0AXXH

0000 1010	0	0	GPIO5	0	0	0	0	GPIO0
(15-8)	(7)	(6)	(5)	(4)	(3)	(2)	(1)	(0)

Status = 0AXXH

0000 1010	0	0	GPIO5	0	0	0	0	GPIO0
(15-8)	(7)	(6)	(5)	(4)	(3)	(2)	(1)	(0)

Setting GPIO0 to 1 enables GPIO0 as an output to provide the DATAFLAGN signal

Setting GPIO5 to 1 enables GPIO5 as an output to provide the Frame Interrupt pin FR

By default both these control bits are set to zero.

9.25.2 Configure GPIO input/output direction:

Command = 09XXH

0000 1001	GPIO-7	GPIO-6	X	X	GPIO-3	GPIO-2	GPIO-1	X
(15-8)	(7)	(6)	(5)	(4)	(3)	(2)	(1)	(0)

Status = 09XXH

0000 1001	GPIO-7	GPIO-6	1	1	GPIO-3	GPIO-2	GPIO-1	1
(15-8)	(7)	(6)	(5)	(4)	(3)	(2)	(1)	(0)

setting the GPIO bit to '1' configures the pin as an output, setting the bit to '0' configures the pin as an input.

9.25.3 Write GPIO pins

Command = 0BXXH

0000 1011	GPIO-7	GPIO-6	X	X	GPIO-3	GPIO-2	GPIO-1	X
(15-8)	(7)	(6)	(5)	(4)	(3)	(2)	(1)	(0)

Status = 09XXH

0000 1011	GPIO-7	GPIO-6	FRN	X	GPIO-3	GPIO-2	GPIO-1	DATA FLAG
(15-8)	(7)	(6)	(5)	(4)	(3)	(2)	(1)	(0)

The status response echos the GPIO bit values in the 0BXXH commands. Regardless of the GPIO direction configured, the status echos the value echos the value written in the 0BXXH command. If the pin is an input, it is not affected by this operation. Bits 0, 4 and 5 of the status response show the current state of the CT8021 reserved output bits.

9.25.4 Read GPIO pins

Command = 5143H (this is the Read FR - frame interrupt command)

0101	0001	0100	0011
(15-12)	(11-8)	(7-4)	(3-0)

Status = YYXXH

GPIO-7	GPIO-6	GPIO-5	GPIO-4 FRN	GPIO-3	GPIO-2	GPIO-1	GPIO-0 DATA- FLAG
(read)	(read)	(read)	(read)	(read)	(read)	(read)	(read)
(15)	(14)	(13)	(12)	(11)	(10)	(9)	(8)

GPIO-7	GPIO-6	GPIO-5	GPIO-4 FRN	GPIO-3	GPIO-2	GPIO-1	GPIO-0 DATA- FLAG
(readback)	(readback)	(readback)	(readback)	(readback)	(readback)	(readback)	(readback)
(7)	(6)	(5)	(4)	(3)	(2)	(1)	(0)

The m.s. 8-bits “YY” of the status response show the state of the GPIO input pins.

The l.s. 8-bits “XX” of the status response show the last data written to the GPIO pins (readback).

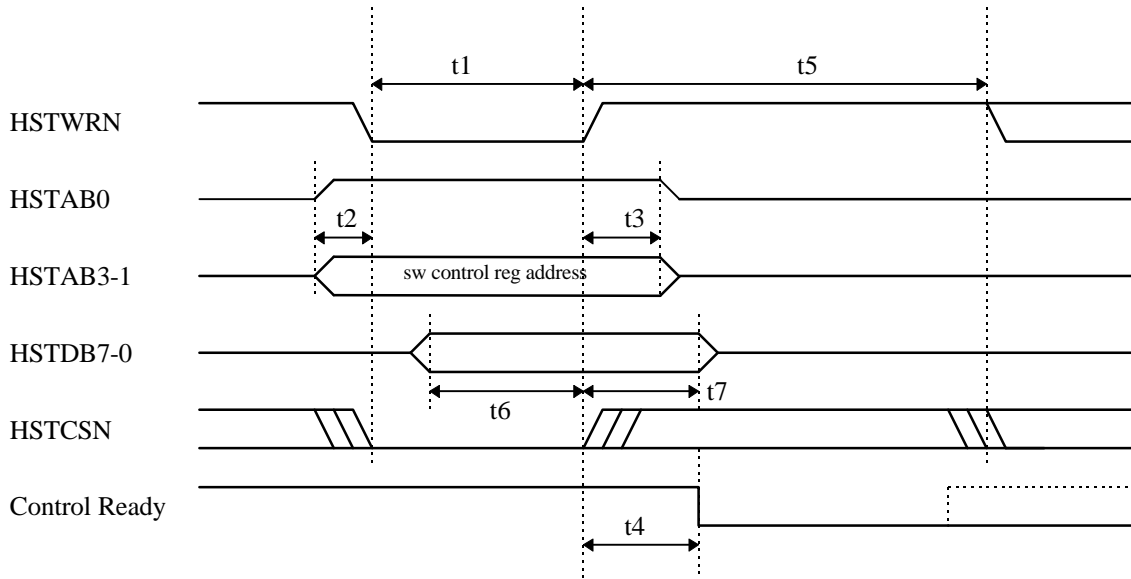
If the GPIO pin is configured as an input, the state of the input pin is reflected in the matching bit in the YY upper byte of the status reply. The matching bit in the XX lower byte shows the last value written to the pin. This “readback” value will control the state of the pin if the direction bit is changed to program the pin as an output. This represents the “potential” state of the pin should it be changed to an output.

If the GPIO pin is configured as an output, the state of the pin as sensed by the GPIO hardware will be reflected in the YY upper byte (reading the state of the actual output pin). The last value actually written to the pin will be shown in the XX lower byte. Under normal circumstances, the bit value in XX and YY will match. Only under abnormal circumstances, for example a short circuit to ground, will the XX and YY bit states be different.

This is also the command shown in the FRAME INTERRUPT section of this data sheet, used for reading the state of the Frame Interrupt pin (FRN).

10. CT8021 Host Interface Timing

10.1 Host Write to Software Control Register m.s. byte

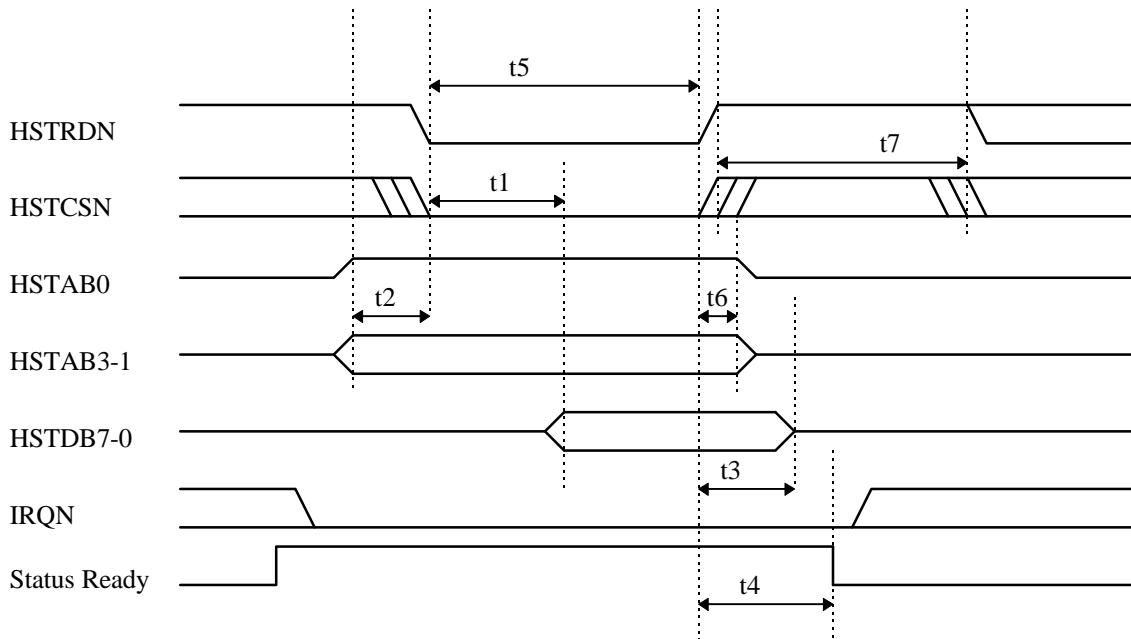


Parameter	Description	Min	Max
t1	HSTWRN pulse width	2 * MAINCLOCKP	
t2	HSTAB3-0 setup time before falling edge of HSTWRN or HSTCSN	5 ns	
t3	HSTAB3-0 hold time after rising edge of HSTWRN or HSTCSN	2 ns	
t4	Delay from rising edge of HSTWRN or HSTCSN to Control Ready cleared		MAINCLOCKP
t5	Recovery time between Host accesses	2 * MAINCLOCKP	
t6	HSTDB7-0 setup time before rising edge of HSTWRN or HSTCSN	20 ns	
t7	HSTDB7-0 hold time after rising edge of HSTWRN or HSTCSN	5 ns	

Notes:

1. Host writes to m.s. byte of Software Control Register
2. Same timings apply to writes to Aux Control Register
3. Control Ready (or Aux Control Ready) cleared by Host Write to Software Control Register (or Aux Software Control Register. Control Ready bit visible to Host in Hardware Status Register. Must be valid in time for read by next host access.
4. MAINCLOCKP period = $2/XIN$ (effective) = 22.2 ns at 45.056 Mips with 4.096 MHz external crystal or 90.112 MHz external clock.
5. t4 may be negative
6. t4 must be less than t5 so that Control Ready is valid for the next host access. The Control Ready signal illustrated is the bit visible to the Host in the Hardware Status Register

10.2 Host Read from Software Status Register m.s. byte

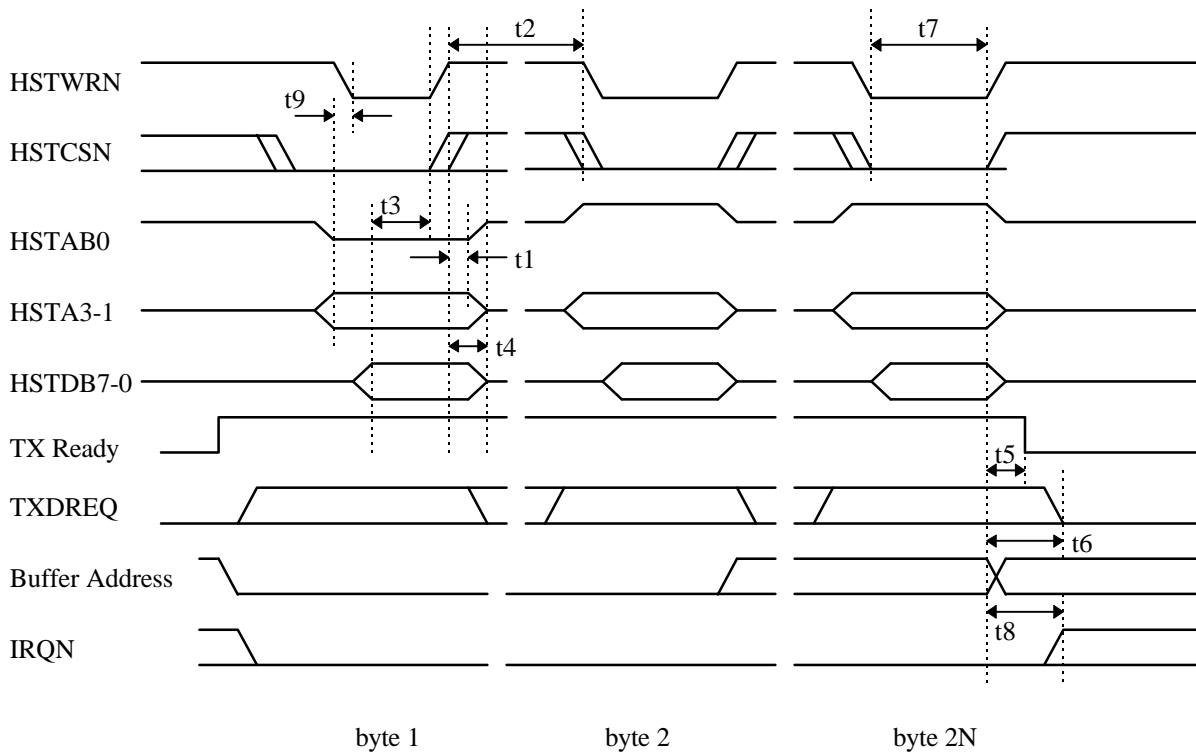


Parameter	Description	Min	Max
t1	HSTRDN or HSTCSN to data out valid		30 ns
t2	HSTAB3-0 setup time before falling edge of HSTRDN or HSTCSN	5 ns	
t3	HSTRDN or HSTCSN inactive to data out tri-state (output disable delay)		30 ns
t4	HSTRDN or HSTCSN positive edge to Status Ready clear		MAINCLOCKP
t5	HSTRDN or HSTCSN width	2 * MAINCLOCKP	
t6	HSTAB3-0 hold time after positive edge of HSTRDN or HSTCSN	2 ns	
t7	Recovery time between Host Accesses	2 * MAINCLOCKP	

Notes:

1. Internal DSP WRITE to Software Status Register (or Aux Software Status Register) sets Status Ready bit in Host Hardware Status Register
2. IRQN asserted to Host assumes that the appropriate IE bit is set in the Host Hardware Control Register
3. Host read of Software Status Register m.s. byte clears Status Ready bit, de-asserts IRQN. The Status Ready bit illustrated is the bit that is visible to the Host in the Hardware Status Register.
4. t4 maybe negative.

10.3 Host Write to Host Transmit Data Buffer Access Port

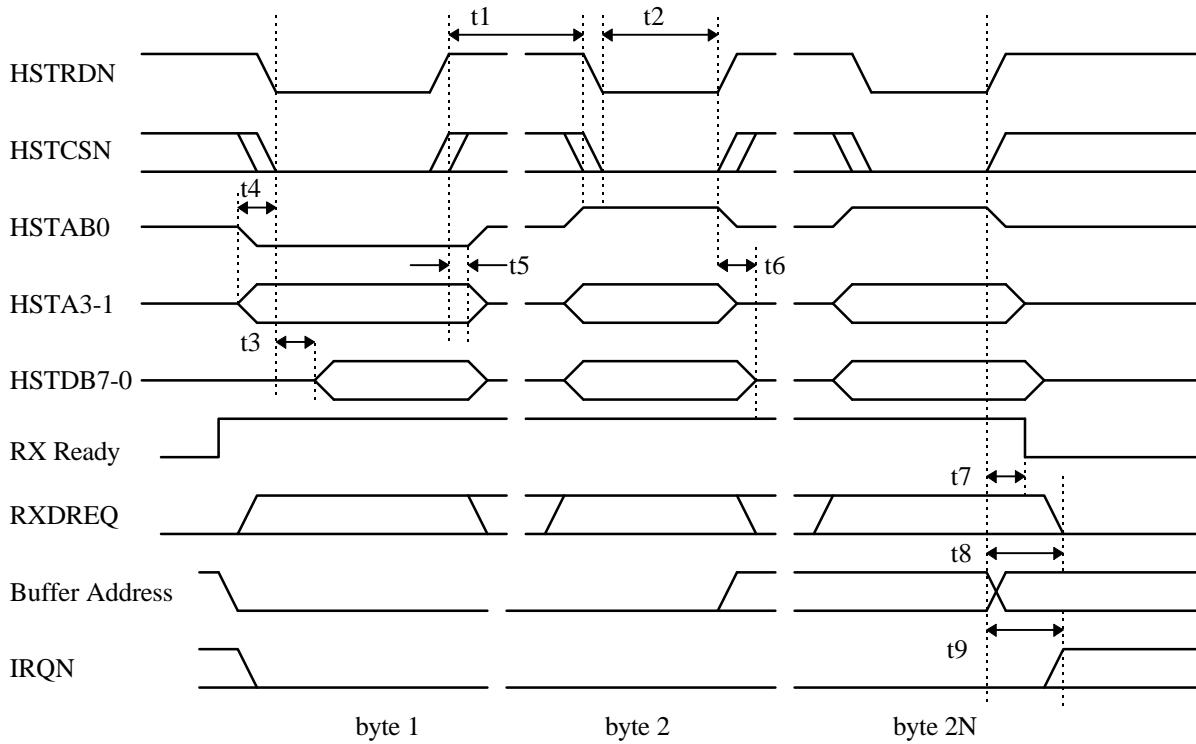


Parameter	Description	Min	Max
t1	HSTAB0-3 hold time after rising edge of HSTWRN or HSTCSN	2 ns	
t2	Recovery time between Host accesses	2 * MAINCLOCKP	
t3	HSTDB7-0 setup time before rising edge of HSTWRN or HSTCSN	20 ns	
t4	HSTDB7-0 hold time after rising edge of HSTWRN or HSTCSN	5 ns	
t5	Final rising edge of HSTWRN to TX Ready cleared		MAINCLOCKP
t6	Final rising edge of HSTWRN to TXDREQ cleared		MAINCLOCKP
t7	HSTWRN width	2 * MAINCLOCKP	
t8	Final rising edge of HSTWRN to IRQN de-asserted		MAINCLOCKP
t9	HSTAB3-0 valid before HSTWRN asserted	5 ns	

Notes

1. Internal Buffer Address cleared by transition of TX Ready bit from 0 to 1
2. Internal Buffer Address increments on rising edge of HSTWRN only if HSTAB0 = 1
3. TX Ready bit clears on final HSTWRN after N access with HSTAB0 =1, where N is the Frame Size value (word count).
4. TXDREQ asserted if TX DMA Enable bit is set in the Hardware Control Register.
5. TXDREQ may operate in Burst Mode or Single Cycle Mode.
6. IRQN driven by TX Ready if TX Ready IE bit set in Host Hardware Control Register.
7. t5, t6 and t8 may be negative

10.4 Host Read From Host Receive Data Buffer Access Port



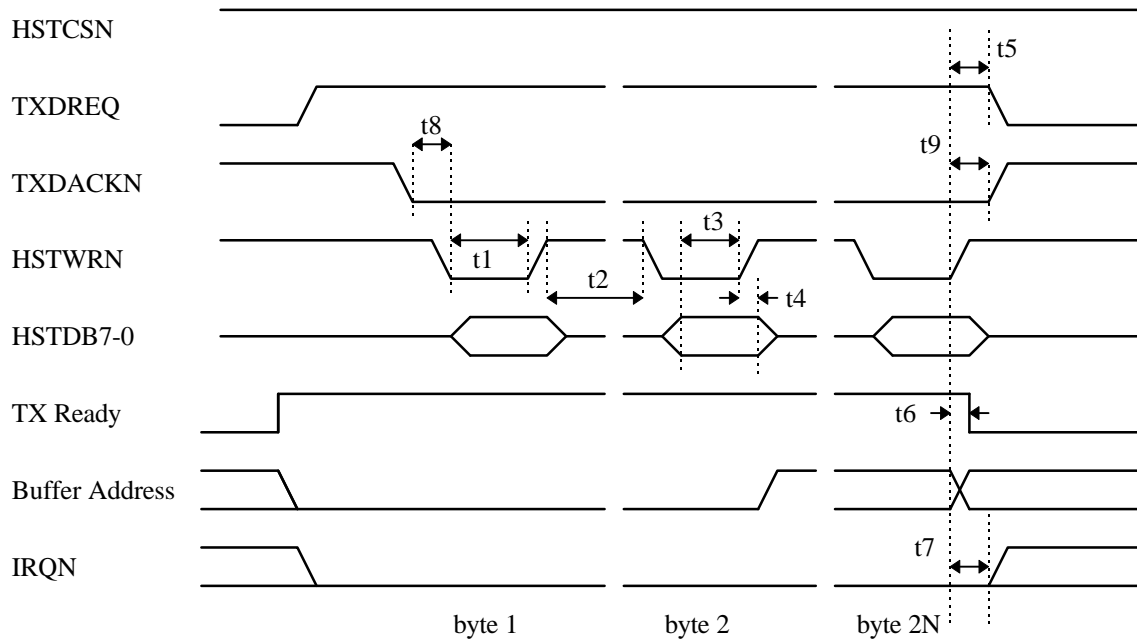
Parameter	Description	Min	Max
t1	Recovery time between Host accesses	2 * MAINCLOCKP	
t2	HSTRDN width	2 * MAINCLOCKP	
t3	HSTRDN or HSTCSN to data out valid		30 ns
t4	HSTAB3-0 setup time prior to falling edge of HSTRDN or HSTCSN	5 ns	
t5	HSTAB3-0 hold time after positive edge of HSTRDN or HSTCSN	2 ns	
t6	HSTRDN or HSTCSN inactive to data out tri-state (output disable delay)		30 ns

t7	Final rising edge of HSTRDN to RX Ready cleared		MAINCLOCKP
t8	Final rising edge of HSTRDN to RXDREQ cleared		MAINCLOCKP
t9	Final rising edge of HSTRDN to IRQN de-asserted		MAINCLOCKP

Notes:

1. Internal Buffer Address cleared by transission of RX Ready bit from 0 to 1.
2. Internal Buffer Address increments on HSTRDN only if HSTAB0 = 1.
3. RX Ready bit clears on final HSTRDN after N accesses with HSTAB0=1, where N is the Frame Size value (word count).
4. RXDREQ asserted if DMA enable bit is set in the Hardware Control Register.
5. RXDREQ may operate in Burst Mode or Single Cycle Mode.
6. IRQN driven by RX Ready if RX Ready IE bit is set in the Host Hardware Control Register.
7. t7, t8, t9 may be negative

10.5 DMA Write to Host Transmit Data Buffer Access Port (Burst Mode)

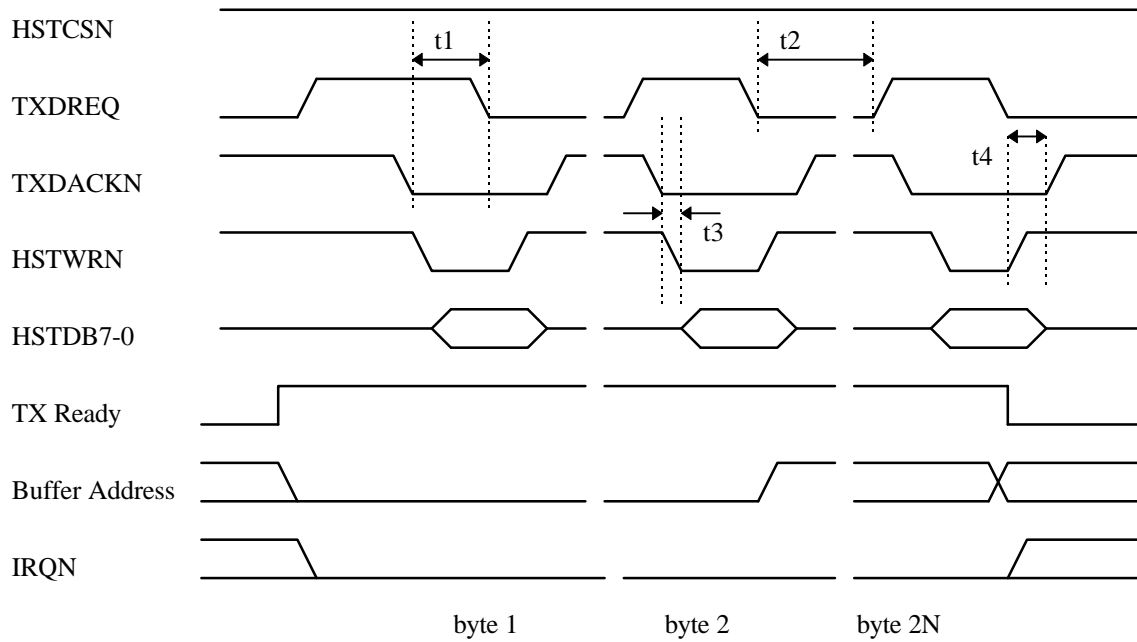


Parameter	Description	Min	Max
t1	HSTWRN width	2 * MAINCLOCKP	
t2	Recovery time between Host accesses	2 * MAINCLOCKP	
t3	HSTDB7-0 setup time before rising edge of HSTWRN	20 ns	
t4	HSTDB7-0 hold time after rising edge of HSTWRN	5 ns	
t5	TXDREQ de-asserted after final rising edge of HSTWRN		MAINCLOCKP
t6	Final rising edge of HSTWRN to TX Ready cleared		MAINCLOCKP
t7	Final rising edge of HSTWRN to IRQN cleared		MAINCLOCKP
t8	TXDACKN asserted before HSTWRN	5 ns	
t9	TXDACKN hold time after rising edge of HSTWRN	2 ns	

Notes:

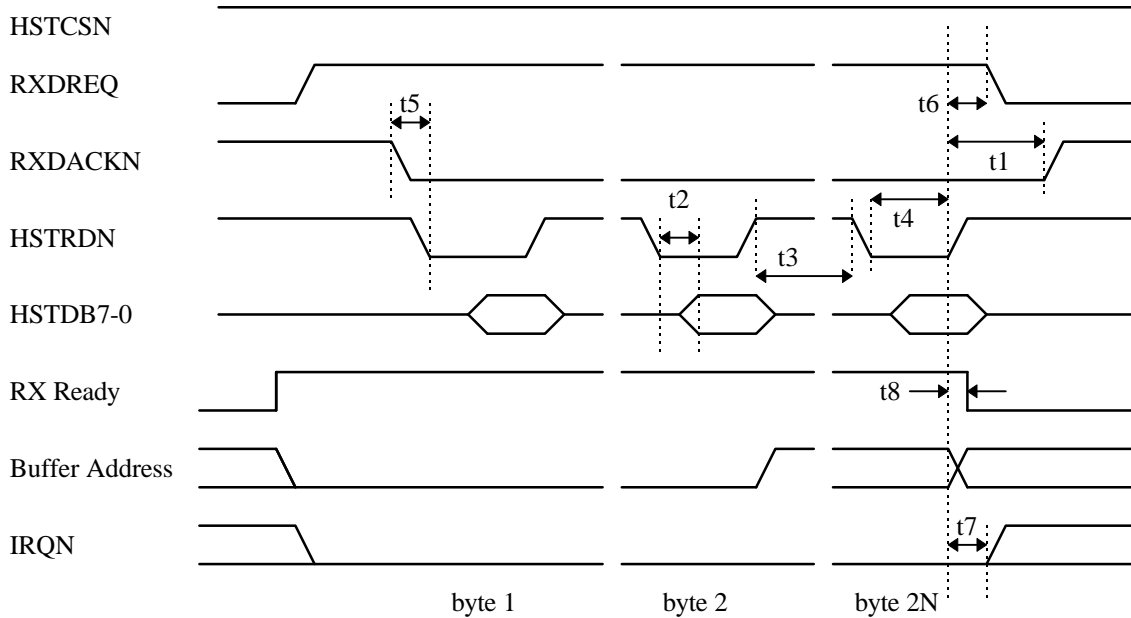
1. t5, t6, t7 may be negative

10.6 DMA Write to Host Transmit Data Buffer Access Port (Single Cycle Mode)



Parameter	Description	Min	Max
t1	TXDACKN asserted to TXDREQ de-asserted response time		3 * MAINCLOCKP
t2	TXDREQ re-assertion delay	16 * MAINCLOCKP	
t3	TXDACKN setup time before falling edge of HSTWRN	5 ns	
t4	TXDACKN hold time after rising edge of HSTWRN	2 ns	

10.7 DMA Read from Host Receive Data Buffer Access Port (Burst Mode)

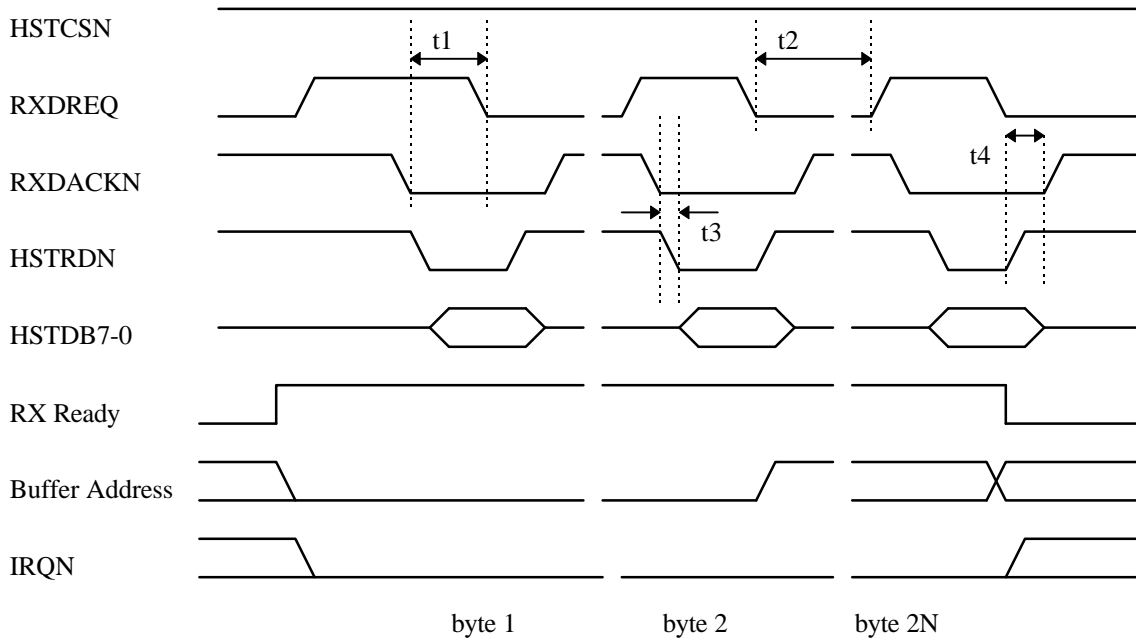


Parameter	Description	Min	Max
t1	RXDACKN hold time after rising edge of HSTRDN	2 ns	
t2	HSTRDN to data valid delay (access time)		30 ns
t3	Recovery time between DMA accesses	2 * MAINCLOCKP	
t4	HSTRDN width	2 * MAINCLOCKP	
t5	RXDACKN setup time before falling edge of HSTRDN	5 ns	
t6	Final rising edge of HSTRDN to RXDREQ deassertion		MAINCLOCKP
t7	Final rising edge of HSTRDN to IRQN deassertion		MAINCLOCKP
t8	Final rising edge of HSTRDN to RX Ready deassertion		MAINCLOCKP

Notes:

1. t6, t7, t8 may be negative

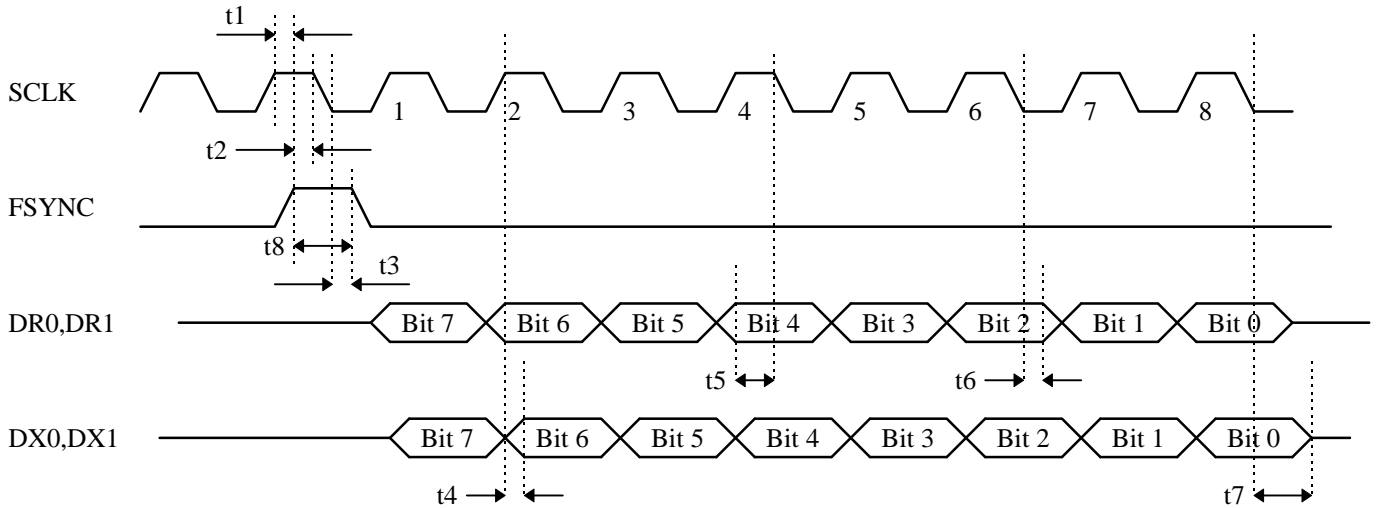
10.8 DMA Read from Host Receive Data Buffer Access Port (Single Cycle Mode)



Parameter	Description	Min	Max
t1	RXDACKN asserted to RXDREQ de-asserted response time		3 * MAINCLOCKP
t2	RXDREQ re-assertion delay	16 * MAINCLOCKP	
t3	RXDACKN setup before falling edge of HSTRDN	5 ns	
t4	RXDACKN hold time after rising edge of HSTRDN	2 ns	

11. CT8021 Codec Interface Timing and AC Specification

11.1 Short Frame Sync

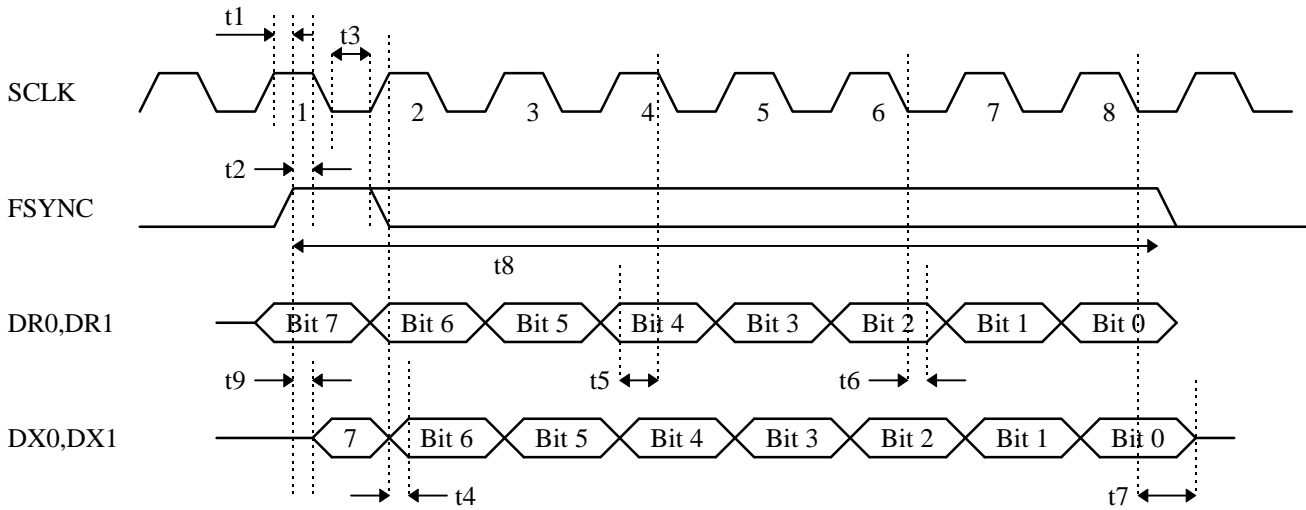


Parameter	Description	Min	Max
t1	Output delay from rising edge of SCLK to rising edge of FSYNC (master mode only)	0 ns	50 ns
t2	FSYNC setup time before falling edge of SCLK (slave mode only)	50 ns	
t3	FSYNC hold time after falling edge of SCLK. (slave mode only)	100 ns	
t4	SCLK rising edge to data output valid	0 ns	30 ns
t5	data in setup prior to falling edge of SCLK	30 ns	
t6	data in hold time after falling edge of SCLK	30 ns	
t7	data out tristate after final falling edge of SCLK	0.25 * SCLKPERIOD (122 ns at 2.048 MHz)	0.5 * SCLKPERIOD (244 ns at 2.048 MHz)
t8	FSYNC width (master mode only)		1 SCLK period (nominal)
	SCLK duty cycle	45%	55%

Notes

- SCLK is 2.048 MHz nominal, SCLKPERIOD is 488 ns nominal
- Timing diagram shows 8-bit mode only. In 16-bit mode, 16 bits of transmit data are shifted into and out of the CT8021. Bit 15 occurs first.

11.2 Long Frame Sync



Parameter	Description	Min	Max
t1	Output delay from rising edge of SCLK to rising edge of FSYNC (master mode only)	0 ns	50 ns
t2	FSYNC setup time before falling edge of SCLK (slave mode only)	50 ns	
t3	FSYNC hold time after falling edge of SCLK. (slave mode only)	100 ns	
t4	SCLK rising edge to data output valid	0 ns	30 ns
t5	data in setup prior to falling edge of SCLK	30 ns	
t6	data in hold time after falling edge of SCLK	30 ns	

t7	data out tristate after final falling edge of SCLK	0.25 * SCLKPERIOD (122 ns at 2.048 MHz)	0.5 * SCLKPERIOD (244 ns at 2.048 MHz)
t8	FSYNC width master mode slave mode	 1 SCLK period	8 SCLK periods in 8-bit mode (nominal). 16 SCLK periods in 16-bit mode (nominal).
t9	ms data bit valid from rising edge of FSYNC. (slave mode only - output enable delay from FSYNC)	0 ns	30 ns
	SCLK duty cycle	45%	55%

Notes

1. SCLK is nominally 2.048 MHz
2. Timing diagram shows 8-bit mode only. In 16-bit mode, 16 bits of data are shifted out of, or into the CT8021. Bit 15 is shifted first.
3. In master mode, FSYNC is 8 SCLK wide (8-bit mode) or 16 SCLK wide (16-bit mode)
4. In slave mode, FSYNC acts as an output enable for DX0,DX1 for the ms data bit during the first SCLK period. (There is no internal synchronization delay on the first bit)
5. In slave mode, once the FSYNC signal has been present for one SCLK falling edge, the internal output enable for DX0 and DX1 is latched and remains active for the appropriate number of SCLK periods. The DX0 and DX1 outputs remain enabled, regardless of the state of FSYNC until following the final SCLK falling edge.
6. In master mode, where FSYNC is an output, it may be used as a tri-state control for gating the receive data. FSYNC will not be de-asserted by the CT8021 until the least significant receive data bit has been latched internally.

12. ELECTRICAL CHARACTERISTICS

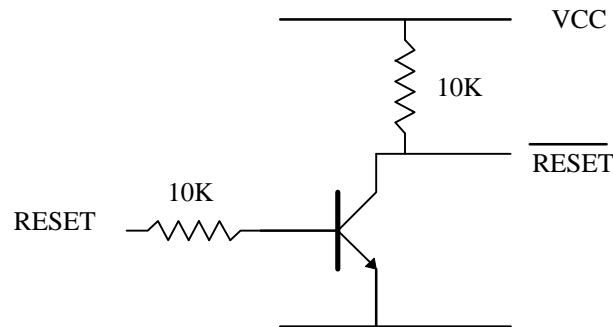
Absolute maximum ratings over specified temperature range

Supply voltage range, VCC	-0.3 V to 6 V
Input voltage range	-0.3 V to 6 V
Operating free-air temperature range	0°C to 70°C
Storage temperature range	-55°C to 150°C

Recommended operating conditions

PARAMETERS	MIN	TYP	MAX	UNIT
V _{CC} supply voltage	4.5	5	5.5	V
V _{SS} supply voltage		0		V
V _{IH} high-level input voltage (all digital input pins except RESET)	2			V
V _{IL} low-level input voltage (all digital input pins except RESET)			0.8	V
VT+ positive going RESET threshold (at VCC = 5.0 v)		3.35	4.6	V
VT- negative going RESET threshold	1.3	1.9		V
I _{OH} high level output current			+4	mA
I _{OL} low level output current			-4	mA
Oscillator crystal (100 ppm) Note the exact 4.096 MHz crystal frequency used to generate the 45.056 mainclock via the x11 PLL is required only if the CT8021 is to operate in codec master mode and generate a SCLK signal at exactly 2.048 MHz.		4.096	4.096	MHz
T _A operating free-air temperature	0		70	°C

The following circuit is recommended for driving the CT8021 RESET pin in order to meet the V_{IH} spec for this pin:



Electrical characteristics over recommended operating free-air temperature range (unless otherwise noted)

PARAMETER	TEST CONDITIONS	MIN	TYP	MAX	UNIT
V _{OH} high level output voltage	VCC = 5V	2.4			V
V _{OL} low level output voltage				0.6	V
I _{OZ} off state output current	VCC = Max			20	μA
I _I input current (pins without pull down resistors)				±20	μA
I _I input current (pins with pull down resistors)				+20 -60	μA
ICC Supply Current	Operating Mode	f = 45.056 Mips			mA
	Stop Device Mode	VCC = 5V	0.5	2	mA
	No Clock 25 °C all inputs GND all outputs NC	VCC = 5V		50	uA
CI Input capacitance				20	pF

DTMF & Tone Generation Characteristics

PARAMETER	MIN	TYP	MAX	UNIT
DTMF twist (high/low tone)		2	±8	dB
Tone generator frequency accuracy	-0.1		+0.1	%fc
Tone generator level*	-24		+6	dB0
Tone generator level accuracy	-0.5		+0.5	dB

* 0 dB0 is 0.707 Vrms programmable with 2 dB resolution. Assumes FSYNC = 8.000 KHz

Acoustic echo cancellor performance

PARAMETER		NOTE
Acoustic Echo Cancellation	30 dB	1
Max echo delay	10 ms	2,3

Notes:

1. The acoustic echo cancellor acheives 30 dB of cancellation for white noise in less than 1.0 seconds with the speaker codec output connected to the microphone codec input via a 2:1 attenuator and an output level equivalent to -10 dBm0.
2. The acoustic echo cancellor can cancell echos up to 10 ms (approximately 3.4 meters or 11 feet in sound propagation).
3. Assumes operation of codec at 8.0 KHz sample rate

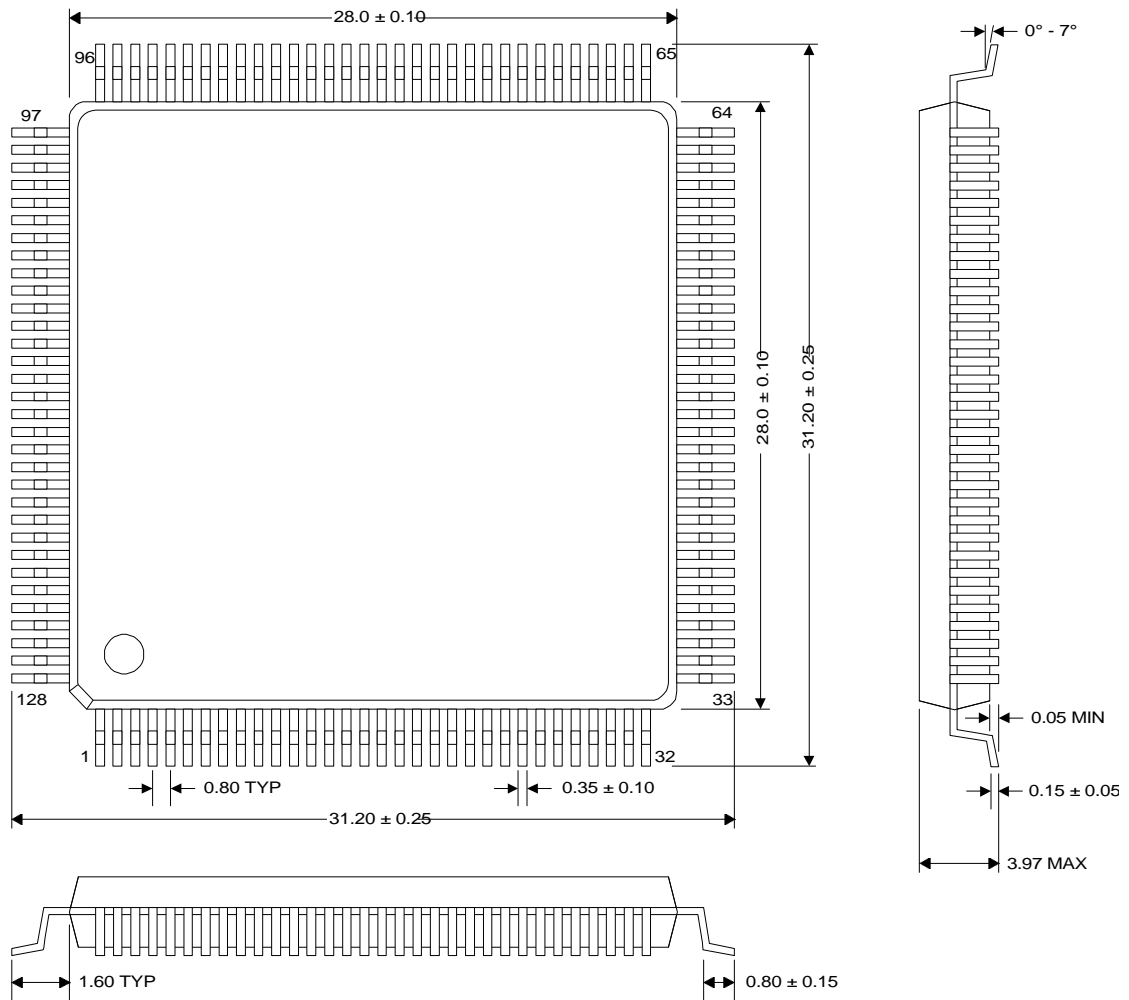
Electrical echo cancellor performance

PARAMETER		NOTE
Training pulse duration	16ms	1
Echo duration	23.25ms	1
Canceler length	7.38 ms	1,2
Training pulse peak amplitude	350 mV	3
Training pulse average peak amplitude	250 mV	3
Electrical echo cancellation	34 dB	4

Notes:

1. Assumes operation of codec at 8.0 KHz sample rate.
2. The electrical echo cancellor will model the impulse response of the attached line circuit for an impulse response up to this duration.
3. Measured at the codec output pin using a TP3054 Texas Instruments mu-law codec.
4. The electrical echo cancellor will achieve 34 dB cancellation for white noise with the line codec output short circuit connected to the line codec input.

13. Mechanical Data - CT8021



Appendix A : Speakerphone Theory of Operation

SPEAKERPHONE

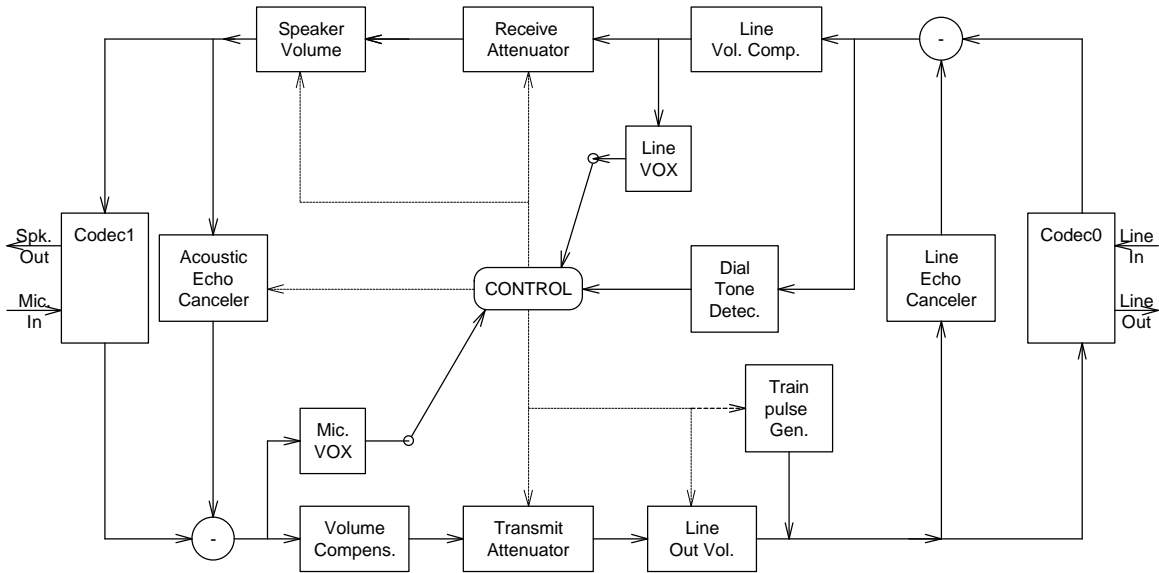


Figure 1.a - Digital part

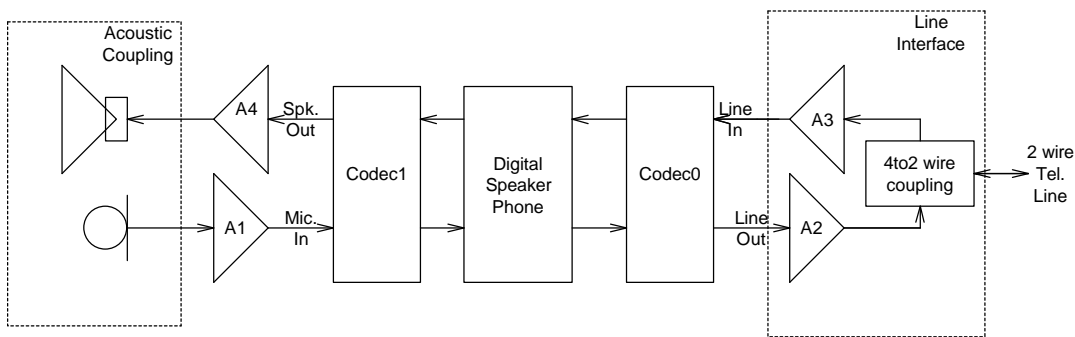


Figure 1.b - Acoustic and analog parts

Introduction

Implementation of a speakerphone system involves three primary sections: external analog circuitry, physical acoustic devices, and the digital canceler (the last, provided by the CT8021). Optimum performance of the speakerphone can only be achieved by correct design and use of all three sections.

Analog Circuitry

The CT8021 interacts with the external physical system via two codecs. Codec0 connects to the telephone line, and Codec1 connects to the microphone and speaker of the speakerphone. The input/output amplifiers (A1-A4) should be configured to make best use of the codecs dynamic range without introducing distortion or clipping of the input/output signals. Care should be exercised in the design of the external circuits to avoid undesirable cross-talk between the various sections. The 4-to-2 wire coupling circuitry at the telephone line interface should be designed to minimize the amount of the outgoing (to line) signal reflected (or echoed) back into the incoming signal (from line). The amount of gain within the loop formed by the amplifiers A1-A4, the 4-to-2 wire coupling, and the microphone-speaker acoustic linkage should be minimized.

The amplifier gains must be selected so that, when the voice signal is coming from the microphone and the telephone line is silent, the signal level at the codec microphone input (codec1) is higher than the reflected signal level at the line codec input (codec0). Also, when the voice signal originates from the telephone line, the input signal at codec0 must be higher than the signal due to the acoustic echo present at the microphone codec1 input. This arrangement allows the CT8021 to determine at any time whether the microphone or line side is speaking. The acoustic echo canceler is only adjusted when the CT8021 determines that the dominant signal is coming from the telephone line side.

Acoustic System

The quality of the microphone and speaker has a direct effect on the level of speakerphone performance attainable. Speech transducers which behave in a non-linear fashion introduce distortions into the signal which can not easily be cancelled out. The microphone and speaker should be operated within their linear ranges. Over-driving of either the microphone or speaker must be avoided. Significantly improved performance can be obtained if any physical coupling between the microphone and speaker is eliminated. Ideally, the speaker and microphone should not be housed within the same physical enclosure. Use of a free-standing speaker or microphone, separate from the main system housing is recommended. In a personal computer application, it is often possible to connect to the PC's existing sound system speakers via a sound card's auxiliary input. Use of a free standing external microphone is often more convenient from the users point of view since it allows more flexibility in positioning. Placing of the microphone and speaker such that they are directly facing each other is not recommended. If microphone and speaker are contained within the same physical housing, care should be taken to minimize the physical coupling between them using acoustic isolation materials. In this situation, it is often the case that the physical coupling caused by the housing creates more feedback than the acoustic path. Physical enclosures often exhibit resonances at particular frequencies, these should be damped as much as possible. It is also possible to reduce the effect of these resonances by electrical filtering within the analog circuitry.

CT8021 Digital Canceler

The CT8021 SpeakerPhone system actually contains two digital cancelers: an adaptive acoustical canceler which cancels the acoustic echo between microphone and speaker, and an electrical canceler which cancels the echo caused by the 4-to-2 wire telephone line connection. The CT8021 can be operated in two distinct modes: full duplex mode in which both cancelers are active, and half duplex mode where only the acoustic canceler is used.

Half-Duplex mode

In this half-duplex mode, the loop attenuation selected must be at least 12 dB. In this mode the SpeakerPhone operates by dynamically distributing the loop attenuation selected between the receive and transmit attenuators (see figure 1a). The CT8021 makes a decision as to which direction is talking, and then rapidly shifts the attenuation to the opposite direction. The total attenuation in the loop (receive attenuation plus transmit attenuation) remains constant. The exception to this is when the CT8021 detects abnormal conditions and loop adjustment mode 01 or 10 is selected. Under these circumstances, the CT8021 can (temporarily) introduce an additional 12 dB of attenuation in order to prevent howling or feedback. Abnormal conditions exist when the CT8021 determines that it is unable to cancel the acoustic echo by adjusting the echo canceler coefficients. If loop adjust mode 10 is selected, the CT8021 will attempt to restore the original loop attenuation selected by reducing the additional attenuation in steps of 4 dB once the abnormal circumstances are removed.

Half-Duplex mode is the default mode selected when the SpeakerPhone operation is started, before the training pulse has been sent to the telephone line activating the electrical echo canceler. The SpeakerPhone should always be started with a loop attenuation of at least 12 dB. The CT8021 operating in half duplex mode, provides for host selection of the loop attenuation. This gives the system designer some flexibility in design of the external analog circuit. It also provides a more natural sounding speaker phone since the non-talking direction is never entirely muted as with some other speaker phone implementations. Using the lower attenuation levels offered by the CT8021 avoids the un-natural abrupt switching between active directions, present with alternative implementations, which causes the user to experience break up of the conversation.

Full-Duplex mode

In full-duplex mode, the electrical (line-side) echo canceler must be activated by the host micro-controller. When instructed, the CT8021 sends a short duration “training” pulse to the telephone line to measure the echo (impulse) response of the line. This pulse is audible to the user, but is of very short duration and is not intrusive. It is important that there is no incoming signal from the telephone line present when the training pulse is sent. The host micro-controller must determine by some algorithm, the correct time to send the pulse. On an incoming call, the pulse should be sent 100-200 ms after answering the call (going off-hook). This delay allows time for the telephone line conditions to stabilize and for the ringing signal to cease. On an outgoing call, the pulse should be ideally sent around 50ms after dialing the second digit. Again this allows time for the line conditions to have stabilized and avoids interfering with digit detection by the local telephone exchange. Sending the training pulse at this time also helps ensure that the line is silent since at this point no ring tone or busy signal is likely to be present. Sending the training pulse at the end of a speed-dial or re-dial sequence also works well, provided it is sent before ringing tone is presented. Sending the pulse after the first digit is not recommended since it is sometimes the case that a secondary dial tone is encountered (as in dialing 9 for an outside line). The host can determine if training was successful by using the Get Electrical Echo Canceler Quality Factor command.

The value of this parameter indicates the degree to which echoes from the telephone line are being canceled. The precise threshold value which indicates success or failure, depends to an extent on the design and amount of gain of the external hardware. Once the electrical echo canceler has been successfully trained, the loop attenuation can be safely reduced to zero or one of the other full-duplex attenuation factors. Since it is difficult to anticipate all circumstances under which the speakerphone may be used, it is recommended that in addition to the automatic training pulse control logic described, that the facility for the user to manually send the pulse is provided. The speakerphone system should provide a manual speakerphone reset push button that the user can press to activate full duplex mode if the automatic control logic fails for any reason.

Volume Control

The CT8021 includes volume control both for the line output level (8 dB range) and speaker output level (28 dB range). The volume should be adjust using the CT8021 rather than by changing external physical gain. In this way the CT8021 is aware of the change in loop gain directly and can internally compensate for it, instead of having to adaptively adjust for the change. The CT8021 also supports speaker and microphone mute.

Dial Tone Detection

The CT8021 includes special handling for constant level tone signals such as dial tone. Signals of this nature are ignored by the voice detectors used for controlling the loop attenuation. When a steady tone is detected for a period of approximately 1 second, priority (or minimum attenuation) is automatically switched to the line side so that the user will hear the dial tone at full volume. Without this special handling, the CT8021 would see that both the receive and transmit direction had constant signal energy and so would distribute the loop attenuation equally between the two directions. Under these circumstances, the dial tone level heard by the user would be attenuated.

Performance

The acoustical echo canceler can handle echo delays up to 10 ms (approximately 3.4 meters or 11 feet in sound propagation). The canceler gives 34 dB cancellation for white noise in less than 0.5 seconds with the speaker codec output short circuit connected to the microphone codec input.

The electrical echo cancellor generates a training pulse of 16 ms duration with an instantaneous peak output voltage of 350 mV. The maximum line echo duration that will be cancelled is 23 ms. With the output codec signal short circuit connected to the input codec signal, the electrical echo cancellor achieves 34 dB of cancellation (measured with white noise). In this test, the echo cancellor performance is primarily limited by the quantization noise associated with the D-to-A and A-to-D translations performed by the mu-law codec.

Example Control Algorithm

This example is provide for illustrative purposes only, system designers are free to construct their own control algorithms as desired.

On detection of an incoming call (off-hook preceeded by ringing signal), the host controller should start the speaker phone in half-duplex mode. The host should then wait around 200 ms following the off hook transition before sending the electrical echo canceller training pulse to allow the line conditions to stabilize. After the training pulse has been sent, the host should interrogate the CT8021 to determine if electrical echo cancellor training was successfull. If not the host should attempt re-training at 500 ms intervals for a total of 3 attempts. If training is not successfull after 3 attempts, the host should continue to let the speaker phone operate in half duplex mode. Training will only fail if there is excessive noise present on the telephone line at the instant that the CT8021 attempts to send the training pulse. As an option, the system designer can provide a manual training or “reset” button to the user, so that the user can decide when the training pulse is sent.

On an outgoing call (e.g. speed dial), the host again starts the the speaker phone in half duplex mode. After dialling of the second digit, the host should wait 50 ms and then attempt to send the training pulse. Once the training pulse has been sent, the host can check the success or otherwise of the training process. If training fails, the host can try to train the electrical echo cancellor after the third and subsequent digits until adequate training is acheived. In the event that training is not possible, the host can allow the speaker phone to continue operation in half duplex mode. Sending the training pulse after the second digit is recommended since in many installations, a secondary dial tone may be presented after dialling of the first digit (for example dialling 9 for an external (public network) line).

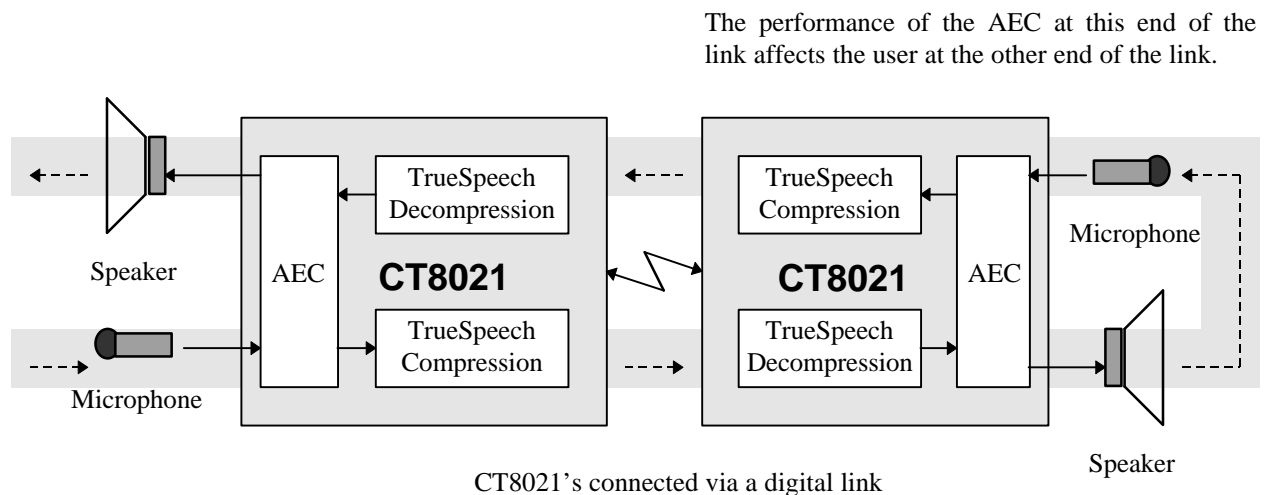
Training needs to be successfully acheived only once in the duration of any call, provided that the physical line conditions do not change. An example of a change that would require re-training is the situation where a second telephone goes off-hook and is operated in parallel with the original telephone.

Note that true full duplex operation will only be acheive after the electrical echo cancellor has successfully trained and the acoustic echo cancellor has had time to self-adjust. The acoustic echo cancellor is only updated when the telephone line side is talking and the microphone side is silent. This may require a few seconds. When operated in loop adjustment mode 10 (the recommended setting), the CT8021 compensates for the initial start-up conditions by inserting additional loop attenuation. Once the acoustic echo cancellor has stabilized, the additional loop attenuation is reduced and then removed.

Note that is an implemenation where the microphone and speaker are physically separate from each other (this is the preferred implementation), the user should be discouraged from moving either microphone or speaker during a call. When the microphone and speaker change relative positions, the acoustic echo cancellor has to work harder to compenstate for the changed acoustic echo path. Under these circumstances, in loop adjustment mode 10, the CT8021 may again introduce temporary extra loop attenuation until the situation re-stablizes. Also if the user desires to change the volume of the loud speaker, this should be done via the CT8021 controls and not by changing the external speaker amplification.

Appendix B :AEC Performance in DSVD Applications

The CT8021 operates the same AEC algorithm in DSVD mode as Speakerphone mode. Since the AEC cancellor operates on the input signal from the microphone, the effect of the AEC is heard at the far-end, as opposed to near-end of the DSVD communication link. The echo path in a symmetrical CT8021 implementation is shown below:



AEC operation in DSVD mode

Under ideal conditions, the AEC can achieve around 34 dB of echo attenuation. With a symmetrical arrangement as above, this can produce up to 68 dB loop (or “round-trip”) attenuation. This is more than sufficient to ensure that acoustic feedback , “howling” or “howl-round” does not occur. However the audible echo (attenuated by up to 34 dB) may be overly annoying to users, due to the significant delay that is often present in this type of configuration. The key problem in this environment is the size of the delay, which may be several hundred milliseconds.

When the AEC is active in **speakerphone** mode, the primary consideration is to prevent “howling”. This condition is met, when the total loop gain in the system is less than 1.0. The round-trip delay in this mode is very short. With a short delay, the human ear is not able to perceive an echo with 34 dB of attenuation except as a minor reverberation sound effect.

When the AEC is active in **DSVD** mode, since the delay is much greater, the **perceptual response to the same level echo is much greater**. The effect of a user hearing their own words echo-ed back to them with a 300-500 ms delay is very distracting, even with 34 dB of attenuation. To overcome this problem, it is necessary to introduce additional attenuation into the loop. To this end, it is recommended that the AEC be operated with at least 12 dB of loop attenuation. While this precludes the possibility of operating in “true” full duplex DSVD speakerphone mode, the additional 12 dB of attenuation enables the CT8021 to achieve “near full duplex” operation.

In this mode, the CT8021 monitors the receive and transmit directions in order to decide which direction is talking. It then attenuates the opposite direction by the loop attenuation amount selected. The loop attenuation setting is programmable by the host in steps of 4 dB from 0 to 28 dB. The active switching of attenuation between the two speech directions is usually not noticeable to the user. The “near full duplex” performance of the CT8021 in this mode, still provides superior performance to comparable systems operating in half-duplex mode, where at any time, one speech direction is completely muted. Also note that, without the AEC active in the illustrated situation, “howling” is extremely likely to occur, making the speech link unusable.

The CT8021 may be operated in DSVD mode without the AEC if a handset is used in place of a speaker-microphone arrangement.

Appendix C Wave File Format (.wav)

WAVE file format is actually a sub-format of RIFF (Resource Interchange File Format) and is as follows:

Example for 8/16 bit linear data (note - the byte offsets given are for illustration only, the byte offsets in a wave file are not absolute):

byte offset	number of bytes	contents
0000H	4	“RIFF” = 52 49 46 46 (hex)
0004H	4	filesize -8 (the size of the waveform chunk)
0008H	4	“WAVE” = 57 41 56 45 (hex)
000CH	4	“fmt “ = 66 6D 74 20 (hex)
0010H	4	format chunk size (16 for linear data)
0014H	2	wFormatTag = 1 (WAVE_FORMAT_PCM)
0016H	2	nChannels (number of channels 1:mono 2:stereo)
0018H	4	nSamplesPerSec (e.g. 8000)
001CH	4	nAvgBytesPerSec (.e.g. 8000, 16000)
0020H	2	nBlockAlign (block size in bytes 8-bit mono:1, 16 bit stereo:4)
0022H	2	wBitsPerSample (e.g. 8 or 16)
0024H	4	“data” = 64 61 74 61 (hex)
0028H	4	waveform data size (bytes)
002CH		start of actual waveform data

Example for TrueSpeech data (note - the byte offsets given are for illustration only, the byte offsets in a wave file are not absolute):

byte offset	number of bytes	contents
0000H	4	“RIFF” = 52 49 46 46 (hex)
0004H	4	filesize -8 (the size of the waveform chunk)
0008H	4	“WAVE” = 57 41 56 45 (hex)
000CH	4	“fmt “ = 66 6D 74 20 (hex)
0010H	4	format chunk size = 32H
0014H	2	wFormatTag = 22H (TrueSpeech tag value)
0016H	2	nChannels = 1
0018H	4	nSamplesPerSec = 8000
001CH	4	nAvgBytesPerSec = 1067 (used for buffer size estimation)
0020H	2	nBlockAlign = 20H
0022H	2	wBitsPerSample = 1
0024H	2	cbSize = 20H (size of extension area following)
0026H	2	wRevision = 1 (For TrueSpeech 8.5)
0028H	2	nSamplesPerBlock = 240
002AH	28	abReserved[28] (reserved area)
0042H	4	“fact” = 66 61 63 74
0046H	4	fact chunk size = 4
004AH	4	time length of data in sample units (e.g. 8000 = 1 sec)

004EH	4	“data” = 64 61 74 61 (hex)
0052H	4	waveform data size (bytes)
0056H		start of actual waveform data

Note that all wave formats except WAVE_FORMAT_PCM are required to include a “fact” chunk which gives the time length of the “data” chunk expressed in samples.

The general format for wave consists of a 4 byte ascii tag for each chunk (“fmt “, “data”) followed by a 4 bytes giving the length of the chunk (not including the 8 byte tag-length header), followed by the actual data for the chunk. A wave file can contain many chunks in any order. An application must search for the chunks it is interested in by skipping over other chunks that it does not understand. For example:

```

“RIFF”
“WAVE”
“test”
16
16 bytes of “test” chunk data
“fmt “
16
16 bytes of “fmt “ data
“data”
1024
1024 bytes of “data”
“more”
32
32 bytes of “more” data

```

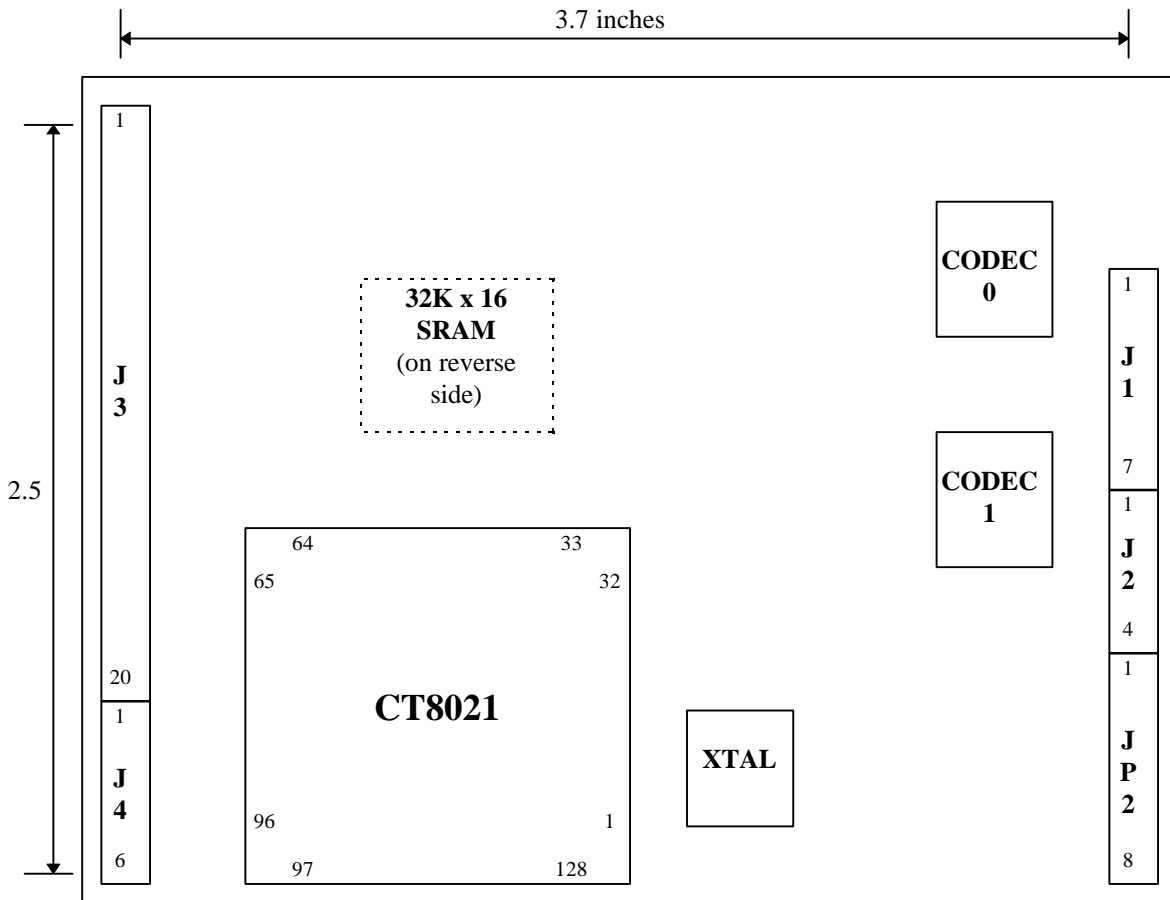
Note that the case of the 4 byte ascii tag is significant.

When an application encounters the tag “test”, it should read the next 4 bytes which contains the length (16) of the “test” chunk, then the application should move forward through the file 16 bytes. This will position the file at the next chunk tag. The application should then read this tag and repeat the above process until it encounters the chunk that it is looking for.

To playback a wave file, the application should first find the “fmt “ chunk and determine if it can handle the wave format it describes. Only then should the application search for the “data” chunk to obtain the actual waveform data.

Appendix D CT8021 Evaluation Board

The diagram below shows the layout of the CT8021. It indicates the positions of the main components and connectors. The actual EVB may contain locations for additional components. These additional components are not required for operation of the CT8021. They are intended to support operation of other devices in the CT8XXX family. Connector pins are on 0.1 inch pitch.



CT8021 Family EVB

The following components that are shown on the **full** EVB schematics were used for development purposes **only** and are not required for normal operation of the CT8021:

- external program SRAMs MCM670R-6
- external EPROM 27010PLCC
- connector JP1

EVB boards normally supplied to customers do not include these additional components. The EVB contains options for several sram configurations. The minimum sram configuration for the CT8021 is 8K x 16, but 32K x 16 is recommend for future compatability.

The jumper pin connection settings for normal operation are TBD

CT8021 EVB Connector Pin Out

J1	
1	AGND
2	Line In
3	AGND
4	Line Out
5	(Line Out-)
6	AGND
7	(Speaker Out-)

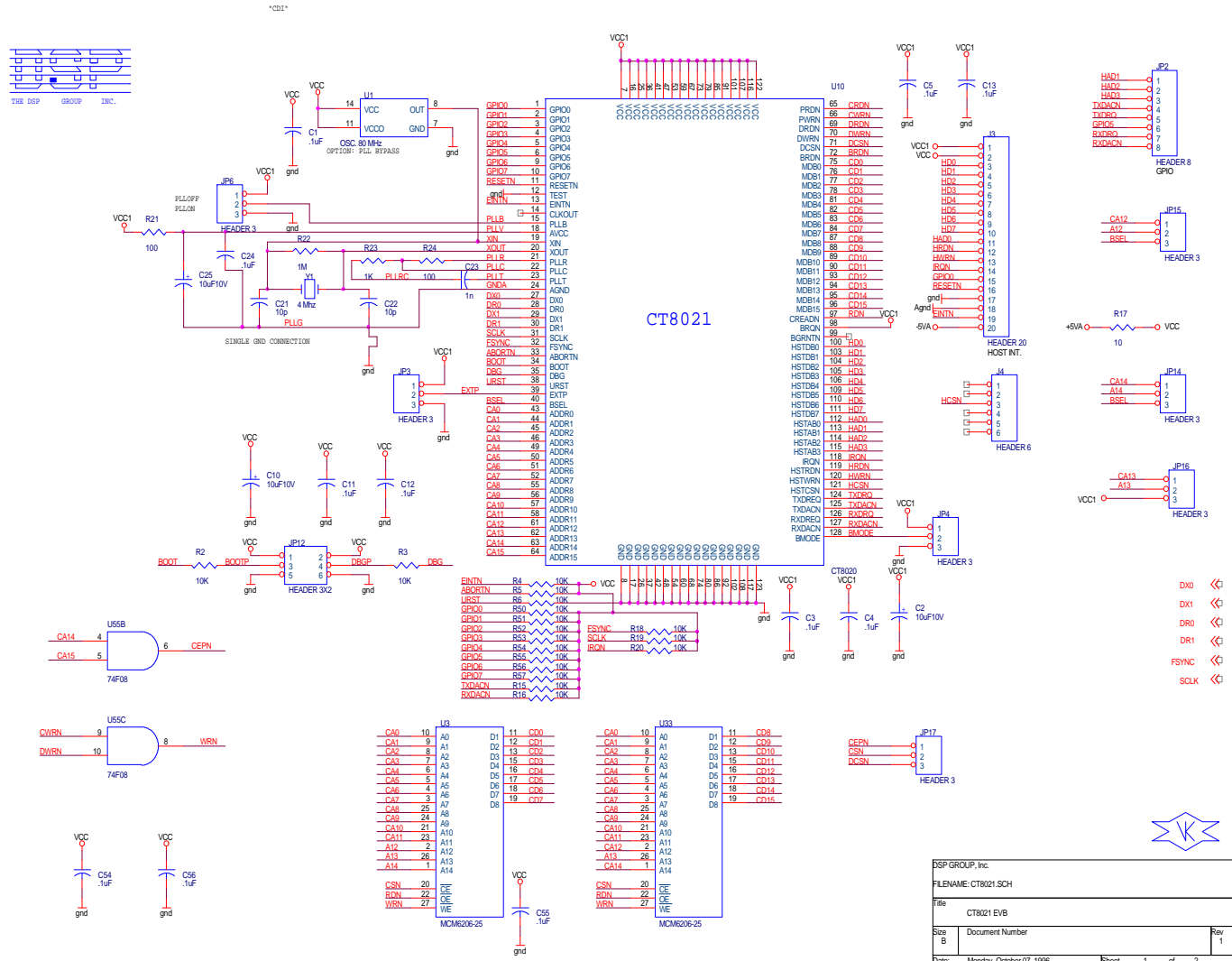
J4	
1	
2	GPIO1
3	HSTCSN
4	
5	
6	

J2	
1	AGND
2	Mic In
3	AGND
4	Speaker Out

JP2	
1	HSTAB1
2	HSTAB2
3	HSTAB3
4	TXDACKN
5	TXDREQ
6	GPI05 / FRN
7	RXDREQ
8	RXDACKN

J3	
1	VCC +5V
2	VCC1 +5V
3	HSTDB0
4	HSTDB1
5	HSTDB2
6	HSTDB3
7	HSTDB4
8	HSTDB5
9	HSTDB6
10	HSTDB7
11	HSTAB0
12	HSTRD
13	HSTWR
14	IRQN
15	GPIO0 / DATAFLAGN
16	RESETN
17	GND
18	AGND
19	EXTINTN
20	-5V

Title (Title)		
Size B	Document Number (Doc)	Rev (Rev/Code)
Date: Monday, October 07, 1996	Sheet 1	of 2

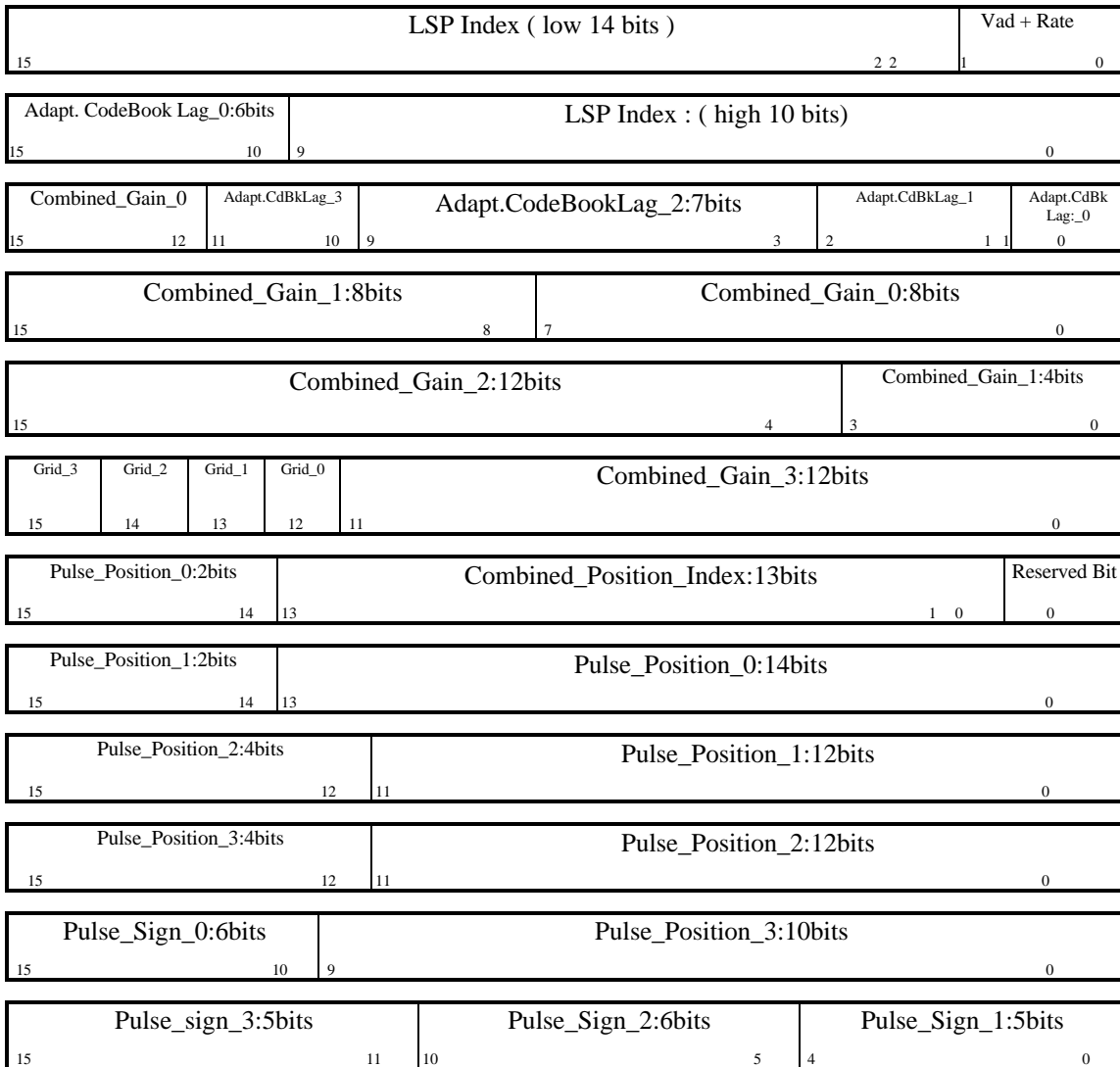


DSP GROUP, Inc.		
FILENAME: CT8021.SCH		
Title CT8021 EVB		
Size B	Document Number	Rev 1
Date: Monday, October 07, 1996	Sheet 1	of 2

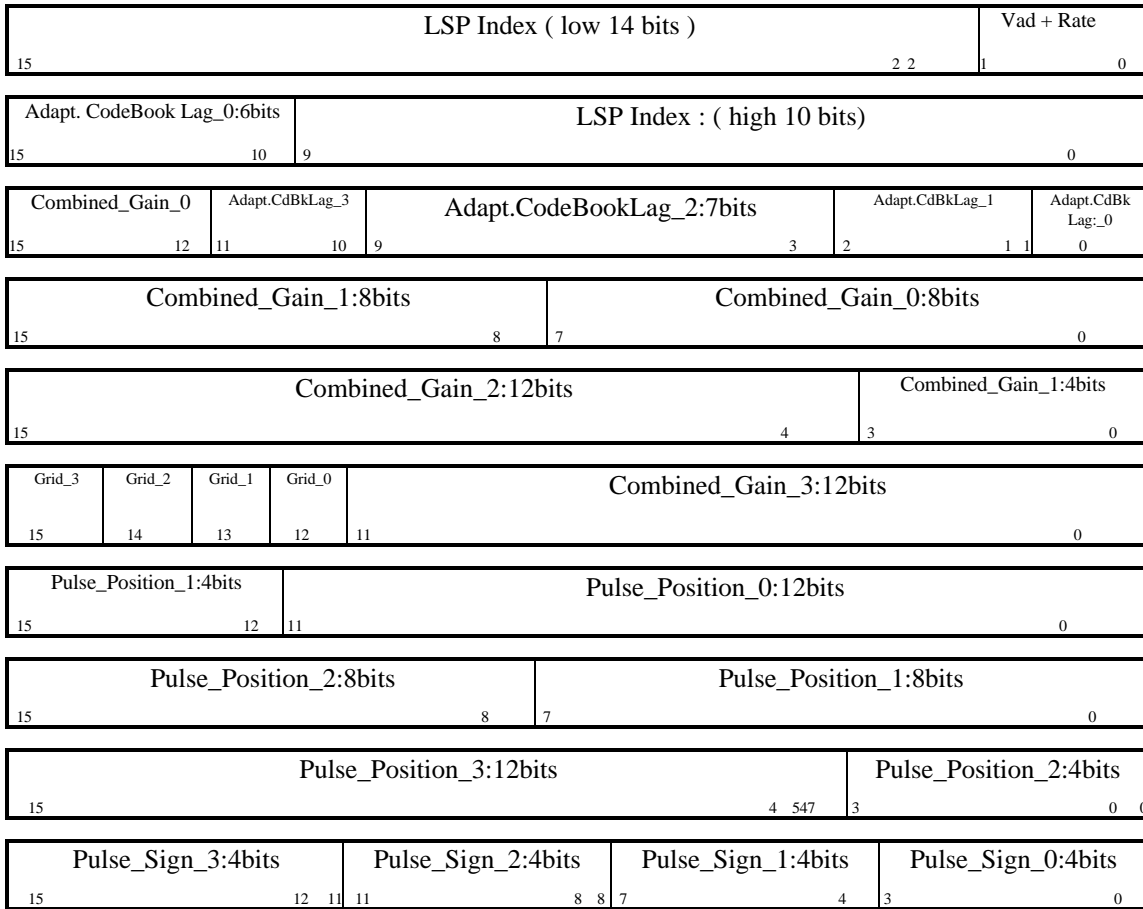
Appendix E :G.723.1 Frame Structure

This appendix shows the bit packing scheme used for compressed speech blocks for the G.723.1 6.3 and 5.3 bit rates. Note that the bit packing arrangement has changed from that used by the (pre-release) v3.0 of G.723.1 that was incorporated in the original CT8020A11AQC engineering samples and v4.0 in the CT8020A11BQC devices.

The following shows the bit packing used by all versions of the CT8021 and the CT8020D11AQC for the 6.3 data rate. The compressed speech frame consists of a block of 12 16-bit words. The least significant 2 bits indicate the type of compressed speech block.



The following shows the bit packing used by the CT8020 and CT8021 for the 5.3 data rate. The compressed speech frame consists of a block of 10 16-bit words. The least significant 2 bits indicate the type of compressed speech block.



G.723.1 Version History

The CT8020D11AQC (firmware revision 0114) and all versions of the CT8021 implement (final-release) version 5.1 of G.723.1. Earlier versions of the CT8020 included pre-release versions of G.723.1 which are not directly compatible with the final v5.1 implementation. Transcoding between versions is possible using conversion software available from DSP Group. Below is a brief description of the G.723.1 revision history and its effect on the CT8020.

v3.0 September 1995 - used in initial CT8020A11AQC engineering samples

v4.0 January 1996 - approximately the same as v3.0 but with different bit packing structure for the compressed speech frames. This version is included in the CT8020A11BQC devices.

v4.1 April 1996 - fixed some instability problems in the G.723.1 algorithm encountered when pure sinewaves encoded.

v5.0 May 1996 - added silence compression VAD/CNG as Annex A of G.723.1. VAD = voice activity detection. CNG = comfort noise generation. This creates 4 possible speech frame sizes:

24 bytes 6.3 kbits/sec

20 bytes 5.3 kbits/sec

4 bytes CNG frame

1 byte silent frame

v5.1 Oct/Nov 1996, minor bug fix release affecting only the VAD feature. This version is implemented in the CT8020D11AQC

The main coder/decoder in the v5.1, v5.0 and v4.1 revisions are identical. When using the v5.1 coder with the VAD feature is disabled, the v5.1 and v4.1 encoders produce identical bit-streams. So inter-working between these versions of G.723.1 is straight forward.

It is possible to convert the compressed data between the various revisions so that, for example, chips to v3.0 can inter-work with v5.1. This requires software to run on the host. This is fairly simple code that re-packs bits and performs some table look-up operations on some bit-fields. DSP Group provides 'C' reference source code for use with the CT8020. This code may be freely incorporated into user software for use with CT8020 systems. DSP Group recommends that this code be incorporated into all software that works with the CT8020A11AQC and CT8020A11BQC devices to provide compatibility with the final G.723.1 v5.1 standard. Systems incorporating older versions of the G.723.1 standard should use the conversion software to promote to the final v5.1 G.723.1 version format.

Version 5.1 of G.723.1 has been approved by the ITU-T as the final version. There will be no further changes to this standard.

The changes introduced by the ITU in version 4.1 of G.723.1 necessitated equivalent changes to be made to the TrueSpeech 4.8 and 4.1 kbps speech coders. As a result of this conversion software must be included in order to enable inter-working between the (DSP Group proprietary) 4.8/4.1 rates in the A11AQC/A11BQC chips and the final CT8020D11AQC silicon.

G.723.1 Control Bits

Following are the control bit assignments as implemented in the various G.723.1 (pre-release) revisions:

v3.0 (**m.s.** 2 bits of first 16-bit word) CT8020A11AQC

bit 15	bit 14	
0	0	(reserved for silence, but not implemented)
0	1	6.3 rate
1	0	5.3 rate
1	1	(reserved)

v4.0 (**l.s.** 2 bits of first 16-bit word) CT8020A11BQC

bit 1	bit 0	
0	0	(reserved for silence, but not implemented)
0	1	5.3 rate
1	0	6.3 rate
1	1	(reserved)

v4.1 and v5.1 (**l.s.** 2 bits of first 16-bit word) - final formal release CT8020D11AQC and all CT8021

bit 1	bit 0	
0	0	6.3 rate
0	1	5.3 rate
1	0	4 byte silence frame (v5.1 only)
1	1	1 byte silence (not transmitted) frame (v5.1 only)

Ordering Information

CT8021A11AQC

128-pin PQFP Device with Commercial Temperature Range (0° - 70°C)

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