AMBE-2000™ Vocoder Chip
User's Manual

Version 3.0

DIGITAL VOICE SYSTEMS, INC.
The Speech Compression Specialists

AMBE-2000™ Vocoder Chip
User's Manual

Version 3.0
April, 2000
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Digital Voice Systems, Inc
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AMBE-2000™ Vocoder Chip END USER License Agreement

1.0 Preliminary Statements and Definitions

1.1 "END USER" shall mean the person and/or organization to whom the AMBE-2000™ Vocoder Chip was delivered or provided to as specified in the purchase order or other documentation. In the event that the END USER transfers his rights under this license to a third party as specified in section 2.2, then this third party shall become an "END USER".

1.2 Digital Voice Systems, Inc. (DVSI) has developed a voice coding method and algorithm (the "Technology") based on the Advanced Multi-Band Excitation ("AMBE®") voice coder. The technology codes speech at bit rates of 2.4 to 9.6 kilobits per second (kbps) including error correction bits.

1.3 "AMBE® Voice Compression Software" shall mean the speech coding software and/or firmware integrated into the AMBE-2000™ Vocoder chip integrated circuit.

1.4 "Voice Codec" shall mean the AMBE-2000™ Vocoder Chip integrated circuit, the AMBE® Voice Compression Software, firmware and associated documentation, including modifications, enhancements and extensions made by or for Digital Voice Systems, Inc. (DVSI) and including circuit diagrams, timing diagrams, logic diagrams, layouts, operating instructions and user manuals.

1.5 DVSI represents that it owns certain "Proprietary Rights" in the Technology and the AMBE® Voice Compression Software, including patent rights in the Technology, and patent rights, copyrights, and trade secrets in the AMBE® Voice Compression Software.

2.0 License Granted

2.1 Subject to the conditions herein and upon initial use of the AMBE-2000™ Vocoder Chip, DVSI hereby grants to END USER a non-exclusive, limited license to use the AMBE® Voice Compression Software in machine readable form solely on the AMBE-2000™ Vocoder Chip. Title to the AMBE® Voice Compression Software remains with DVSI. No license is granted for use of the AMBE® Voice Compression Software on other than the AMBE-2000™ Vocoder Chip. No license, right or interest in any trademark, trade name or service mark of DVSI is granted under this Agreement.

2.2 END USER shall not copy, extract, de-compile, reverse engineer or disassemble the AMBE® Voice Compression Software contained in the AMBE-2000™ Vocoder Chip.

3.0 Transfer of License

3.1 The END USER shall have the right to transfer the AMBE-2000™ Vocoder Chip and all rights under this Agreement to a third party by either (i) providing the third party with a copy of this Agreement or (ii) providing the third party with an agreement written by the END USER (hereinafter "END USER Agreement") so long as the END USER Agreement is approved in writing by DVSI prior to transfer of the AMBE-2000™ Vocoder Chip. The END USER Agreement shall contain comparable provisions to those contained herein for protecting the Proprietary Information from disclosure by such third party. Third parties shall agree to accept all the terms and conditions under either Agreement or the END USER Agreement.

4.0 Term and Termination

4.1 This Agreement is effective upon initial delivery of the Voice Codec and shall remain in effect until terminated in accordance with this agreement.

4.2 This Agreement shall terminate automatically without notice from DVSI if END USER fails to comply with any of the material terms and conditions herein. END USER may terminate this Agreement at any time upon written notice to DVSI certifying that END USER has complied with the provisions of Section 3.3.

4.3 Upon termination of this Agreement for any reason, END USER shall: (i) return all AMBE-2000™ Vocoder Chip purchased or acquired, or in Licensee's possession, to DVSI; (ii) have no further rights to any AMBE® Voice Compression Software or the Technology without a separate written license from DVSI; (iii) discontinue all use of the AMBE-2000™ Vocoder Chip;

5.0 Payments

5.1 In consideration of the materials provided as part of the Voice Codec, and in consideration of the license and rights in the AMBE® Voice Compression Software granted by DVSI, and in consideration of DVSI's performance of its obligations hereunder, END USER agrees to pay to DVSI the fee specified in DVSI's invoice.

6.0 Proprietary Notices

6.1 END USER shall not remove any copyright or proprietary notice on the AMBE-2000™ Vocoder Chip or on the AMBE® Voice Compression Software.

7.0 Proprietary Information

7.1 The parties agree that the AMBE® Voice Compression Software shall be considered Proprietary Information.

7.2 Except as otherwise provided in this Agreement, END USER shall not use, disclose, make, or have made any copies of the Proprietary Information, in whole or in part, without the prior written consent of DVSI.

8.0 Limited Warranty

8.1 DVSI warrants the Voice Codec to be free from defects in materials and workmanship under normal use for a period of ninety (90) days from the date of delivery.

8.2 Except as stated in Section 7.1, the Voice Codec is provided "as is" without warranty of any kind. DVSI does not warrant, guarantee or make any representations regarding the use, or the results of the use, of the Voice Codec with respect to its correctness, accuracy, reliability, correctness or otherwise. The entire risk as to the results and performance of the Voice Codec is assumed by the END USER. After expiration of the warranty period, END USER, and not DVSI or its employees, assumes the entire cost of any servicing, repair, replacement, or correction of the Voice Codec.

8.3 DVSI represents that, to the best of its knowledge, it has the right to enter into this Agreement and to grant a license to use the AMBE® Voice Compression Software to END USER.

8.4 Except as specifically set forth in this Section 7.0, DVSI makes no express or implied warranties including, without limitation, the warranties of merchantability or fitness for a particular purpose or arising from a course of dealing, usage or trade practice, with respect to the Voice Codec. Some states do not allow the exclusion of implied warranties, so the above exclusion may not apply to END USER. No oral or written information or advice given by DVSI or its employees shall create a warranty or in any way increase the scope of this warranty, and END USER may not rely on any such information or advice. The limited warranties under this section 7.0 give END USER specific legal rights, and END USER may have other rights which vary from state to state.

9.0 Limitation of Liability

9.1 In no event shall DVSI be liable for any special, incidental, indirect or consequential damages resulting from the use or performance of the Voice Codec whether based on an action in contract, tort (including negligence) or otherwise (including, without limitation, damages for loss of business profits, business interruption, and loss of business information), even if DVSI or any DVSI representative has been advised of the possibility of such damages.

9.2 Because some states do not allow the exclusion or limitation of liability for consequential or incidental damages, the above limitations may not apply to END USER.

9.3 DVSI's maximum liability for damages arising under this Agreement shall be limited to 20% (twenty percent) of the fees paid by END USER for the particular Voice Codec which caused the damages or that is the subject matter of, or is directly related to, the cause of action.

10.0 Taxes

10.1 All payments required under Section 4.0 or otherwise under this Agreement are exclusive of taxes and END USER agrees to bear and be responsible for the payment of all such taxes (except for taxes based upon DVSI's income) including, but not limited to, all sales, use, rental receipt, personal property or other taxes which may be levied or assessed in connection with this Agreement.

11.0 Export

11.1 United States export laws and regulations prohibit the exportation of certain products or technical data received from DVSI under this Agreement to certain countries except under a special validated license. As of November 30, 1999 the restricted countries are: Libya, Cuba, North Korea, Iraq, Serbia, Taliban in Afghanistan, Sudan, Burma, Yugoslavia and Iran. The END USER hereby gives its assurance to DVSI that it will not knowingly, unless prior authorization is obtained from the appropriate U.S. export authority, export or re-export, directly or indirectly to any of the restricted countries any products or technical data received from DVSI under this Agreement in violation of said United States Export Laws and Regulations. DVSI neither represents that a license is not required nor that, if required, it will be issued by the U.S. Department of Commerce. Licensee shall assume complete and sole responsibility for obtaining any licenses required for export purposes.

12.0 Governing Law

12.1 This Agreement is made under and shall be governed by and construed in accordance with the laws of the Commonwealth of Massachusetts, except that body of law governing conflicts of law. If any provision of this Agreement shall be held unenforceable by a court of competent jurisdiction, that provision shall be enforced to the maximum extent permissible, and the remaining provisions of this Agreement shall remain in full force and effect.

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1. Product Introduction

1.1 General Information

Digital Voice Systems Inc.'s AMBE-2000™ Vocoder Chip is an extremely flexible, high-performance, single chip, speech compression coder. It provides superior voice quality at low data rates. It provides a real-time, full-duplex implementation of the standard-setting AMBE® voice compression software algorithm. DVSI's patented AMBE® voice compression technology has been proven to outperform CELP, RELP, VSELP, MELP, ECELP, MP-MLQ, LPC-10, and other competitive technologies. Numerous evaluations have shown its ability to provide performance equal to today's digital cellular systems at under half the data rate. The AMBE® voice compression algorithm is used in applications throughout the world, including the next generation of digital mobile communication systems.

The AMBE-2000™ Vocoder chip provides a high degree of flexibility in selecting the speech and FEC (Forward Error Correction) data rates. The user can separately select these parameters in 50 bps increments for total rates from 2.4 kbps to 9.6 kbps. Typically for higher error rate channels, the user will apportion a greater percentage of the total bit rate to FEC coding. The AMBE-2000™ voice coder maintains natural voice quality and speech intelligibility at rates as low as 2.4 kbits/sec. The AMBE® algorithm's low complexity allows it to be fully integrated into a low cost, low power integrated circuit, the AMBE-2000™ Vocoder Chip.

The AMBE-2000™ Vocoder Chip offers similar features as to DVSI's AMBE-1000™ Vocoder Chip allowing it to be incorporated into systems already design for the AMBE-1000™ and is interoperable with other DVSI products. The AMBE-2000™ Vocoder Chip delivers improved performance and enhanced modes such as 4.0 kbps toll quality speech and convolutional FEC coding. Along with these enhancements the AMBE-2000™ Vocoder Chip employs a control interface along with the variable data rates and FEC selection as found in .

1.2 Advantages

- Superior Voice Quality
- Low Cost
- No External Memory Required
- Robust to Bit Errors & Background Noise
- Variable Data Rates - 2.0 kbps to 9.6 kbps
- Variable FEC Rates - 50 bps to 7.2 kbps
- Very Low Power (65mW @ 3.3V, 0.11mW Deep Sleep)
- Compact Single Chip Solution: 100 pin TQFP

Features

- High Quality Low Data Rate Speech Coding
- DVSI's Full Duplex AMBE® Voice Coder
- Supports Data Rates of 2.0 kbps to 9.6 kbps in 50 bps increments
- User Selectable Forward Error Correction rates
- Viterbi Decoder (rate 1/4 or more)
- 16 Level Soft Decision Decoding
- Voice Activity Detection (VAD) / Comfort Noise Insertion
- 16 ms. Echo Cancellation
- Single and Dual Tone (DTMF) Detection and Generation

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- Power-Down Mode

- Minimal algorithmic processing delay

- DTMF detection and regeneration with North American call progress tones

1.3 Typical Applications

- Satellite Communications

- Digital Mobile Radio

- Secure Communications

- Cellular Telephony and PCS

- Voice Multiplexing

- Voice Mail

- Multimedia Applications

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2 AMBE™-2000 Application Design Overview

2.1 Basic Operation

In its simplest model, the AMBE-2000™ can be viewed as two separate components, the **Encoder** and the **Decoder**. The Encoder receives an 8kHz. sampled stream of *speech* data (16-bit linear, 8-bit Alaw, or 8-bit ulaw) and outputs a stream of *channel* data at the desired rate. Conversely the Decoder receives a stream of *channel* data and synthesizes a stream of *speech* data. The timing for the interfaces for the AMBE-2000™ Encoder and Decoder are fully asynchronous.

Figure 2-A Basic Operation
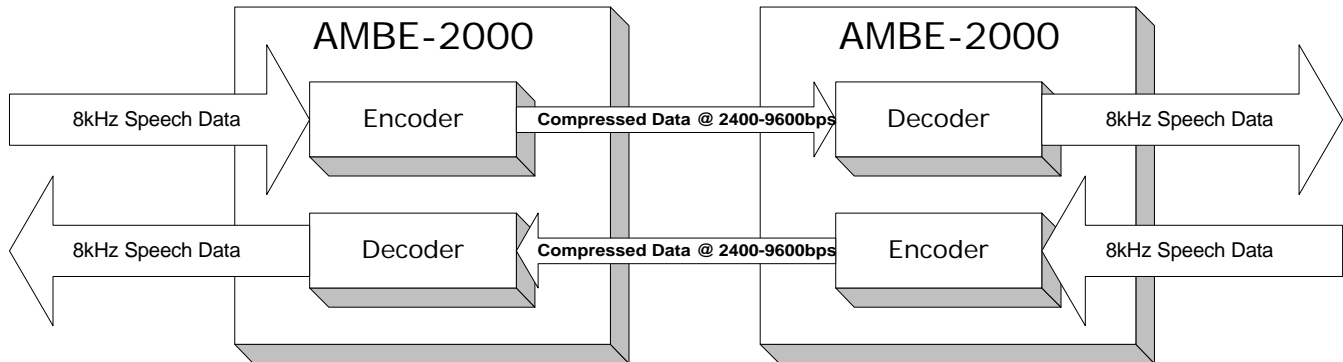
Typically the speech interface is an external A/D-D/A chip. The format of the incoming and outgoing speech data streams are coupled, that is to say they must be the same format (16-bit linear, 8-bit Alaw, or 8-bit μlaw). The channel interface is commonly (but not limited to) an 8 or 16 bit microprocessor or other suitable 'glue logic' hardware capable of performing the rudimentary formatting functions between the AMBE-2000™ channel format and the format of the system channel under design.

Optional functions of the chip, such as echo cancellation, voice activation/detection, power mode control, data/FEC rate selection, etc. are controlled either through hardware control pins (see Section 0) and/or through the decoder command interface (see Section 4) Data sent into the decoder for function control purposes is distinguished from the data to be decoded into speech through a channel format which is described in Section 4.

2.2 Initial Design Considerations

Some of the initial design considerations the application engineer will face are the following:

- Choice of A/D-D/A chip.
- Choice of Channel Interface.
- Speech and FEC Rates.

2.2.1 A/D – D/A Overview

The choice of the A/D-D/A chip is critical to designing a system with superior voice quality. Given that Alaw and μlaw companding chips are already incorporating some compression to reduce the number of bits per sample, it is recommended that, when possible, a 16-bit linear device be used for maximum voice quality. When choosing a device, pay particular attention to Signal to Noise ratios and Frequency Responses of any filters that may be present on the analog front end of these chips. The Alaw and μlaw interfaces are also provided for the design engineer who is trying to fit to pre-existing conditions or is under other cost type restraints.

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2.2.2   Channel Interface Overview

The channel interface is meant to be flexible to allow for easy integration with the system under design. The basic hardware unit of the interface is a serial port. The serial mode can run in *passive* or *active* modes. Simply stated, the control signal for serial mode can be derived by the AMBE-2000™ chip or they can be derived externally.

Under normal operation, every 20msec the encoder outputs a frame of coded bits, and the decoder needs to be delivered a frame of coded bits. There is some formatting of the data for both the encoder and the decoder. The primary purpose of the formatting is to provide alignment information for the encoded bit stream. The data has two formats, *Formatted* and *Unformatted*. Serial mode can run in either **Formatted** *or* **Unformatted** mode.

The **Formatted** and **Unformatted** modes are explained in full detail in Section 4, but essentially the two formats are trying to achieve the same function, to provide positional information regarding the outgoing and incoming coded data streams. In **Formatted** mode each 20msecs of output data from the encoder is preceded by a known structure. This structure also embeds some status type flags, meant for local control purposes, within it. The only data from the **Formatted** format that is typically sent across the transmission channel under design are the actual encoded bits at the desired rate.

In **Formatted** mode, it is the responsibility of the designed system to pass enough information along with the encoded bits such that the **Formatted** format needed by the decoder can be reconstructed on the other side. This extra information, or overhead, is going to be very specific to the system under design, but at a minimum needs to pass enough information to reliably reconstruct the 20msec frame structure at the other end for the decoder.

In **Unformatted** mode the data coming out of the encoder can be thought of as a continuous stream of voice data with the framing information embedded within the encoded bits. One advantage of this type of set-up is that the system does not have to add any bandwidth for overhead to the channel. The disadvantage is that the decoder needs 10-12 incoming frames in order to gain synchronization with the data stream before it can properly synthesize the speech waveform. Also, the **Unformatted** mode only commits a single bit per frame to maintaining data alignment. In higher error rate channels the performance will be improved by adding more bits per frame to the alignment information (which is more easily performed when using **Formatted** mode)

Additional flexibility is given to the channel interface to the encoder and decoder by allowing the AMBE-2000™ Vocoder Chip to run in *Passive* or *Active* modes. In Passive mode, data strobes are provided by an external source, while in Active mode, data strobes are provided by the AMBE-2000™ Vocoder Chip. The serial interfaces can be run in Passive or Active modes. See Section 4 for full details and timing for both **Formatted** and **Unformatted** data.

2.2.3   Speech and FEC Rate Selection Overview

The total coded bit rate is the sum of two components, the Speech Data and the Forward Error Correction (FEC) Data. The addition of FEC data to the speech data allows the decoder to be able to correct a limited amount of errors within each frame should they arrive corrupted. If the channel is expected to have more errors then more bits should be dedicated to FEC. At the same time, voice quality will increase if the number of speech bits can be maximized.

3 Hardware Information

3.1   Special Handling Instructions

The AMBE-2000 uses the TM320-LC541-66 core. For more details on handeling, electrical characteristics, packaging, or timing constraints please refer to the TMS320-C54x manual found at http://www.s.ti.com/sc/psheets/sprs039c/sprs039c.pdf (Adobe Acrobat). Although the AMBE-2000™ Vocoder Chip incorporates input protection circuitry, to avoid damage from the accumulation of a static charge, industry standard electrostatic discharge precautions and procedures must be employed during handling and mounting.

The 100 pin TQFP package design of the AMBE-2000™ Vocoder Chip allows it to be mounted by infrared reflow, vapor-phase reflow or equivalent processes. The peak package body temperature must not exceed 220◦C.
The AMBE-2000™ Vocoder Chip requires baking before mounting, if any of the following conditions exist:

- Humidity indicator card (included in packaging) shows exposure to > 20 % when read at 23◦C + 5◦C

- Devices were not shipped in a package designated as "moisture controlled."

- Not mounted within 168 hours of receipt, at factory conditions of $\leq$30◦C and <60% RH

- If the device has not been stored at $\leq$ 20% RH

DVSI's recommended bake out procedures:

- For low-temperature device containers: 192 hours at 40◦C + 5◦C / -0◦C and < 5% Relative Humidity

- For high-temperature device containers : 24 hours at 125◦C + 5◦C.

3.1.1   Storage

To insure maximum shelf life in long term storage, AMBE-2000™ Vocoder Chips should be kept in a moisture controlled package at <40◦C and <90% Relative Humidity

3.2   Pin Descriptions

Pin Number	Pin Descriptive Name	Pin Direction	Notes
77	CHANN_SEL1	Input	Channel Interface Selection Pins : Use these bits to select the channel interface type (formatted,unformatted active, passive) according to Table 4-A. See full description in section 4.2.
75	CHANN_SEL0	Input	Used with CH_SEL1 to select channel operation mode
85	CODEC_SEL1	Input	A/D-D/A Select Pins see Table 6-A to select the interface.
84	CODEC_SEL0	Input	
74	RATE_SEL4	Input	Coding Rate Select Pins : Use these bits to select the voice and FEC rates according to section 7.2. The coding rates are also selectable using the Control Word interface described in section 5.2.4.
73	RATE_SEL3	Input	
72	RATE_SEL2	Input	
71	RATE_SEL1	Input	
70	RATE_SEL0	Input	
86	VAD_EN	Input	Voice Activation Detection Enable Pin. Active HIGH. See Section 7.4. VAD can also be enabled/disabled using the Control Word interface as described in section 5.2.9.
78	ECHOCAN_EN	Input	Echo Canceller Enable Pin. Active HIGH. See Section 7.3. The Echo Canceller can also be enabled/disabled using the Control Word interface as described in section 5.2.9.
83	SLEEP_EN	Input	Standard Sleep Enable Pin. Active HIGH. See Section 7.6.1.
82	SLIP_EN	Input	Slip Control Enable Pin. Active HIGH. See Section 7.7.
68	X2/CLKIN	Input	Clock Input 1. 16.384 Mhz input. See Section 3.4
67	X1	Input	Output from internal oscillator for the crystal. If the internal oscillator is not used this pin should be unconnected.
69	RESETN	Input	AMBE-2000 Reset pin. Active LOW. See Section 3.3
20	EPR	Output	Encode Packet Ready: Following a reset, this signal will have a high to low transition to indicate the first packet is ready. The next packet will be ready approximately 20 msec later. **See Note 1.**
79	SOFT_EN	Input	Soft decision decoding enable. Enables 4 bit soft decision error decoder.
80	BAUD_SEL0	Input	Baud Rate Selector for unformatted serial mode See Table TBDL
81	BAUD_SEL1	Input	
32	CHAN_RX_DATA	Input	Channel Receive Data to AMBE-2000
42	CHAN_TX_DATA	Output	Channel Transmit Data from AMBE-2000
28	CHAN_RX_CLK	Input	Channel Receive Clock
34	CHAN_TX_CLK	Input	Channel Transmit Clock
38	CHAN_TX_STRB	I/O	Channel Transmit Frame Synchronization Pulse
30	CHAN_RX_STRB	Input	Channel Receive Frame Synchronization Pulse
29	CODEC_RX_STRB	Input	Frame synchronization pulse for A/D data. Should be connected to CODEC_TX_STRB
37	CODEC_TX_STRB	Input	Frame synchronization pulse for D/A data. Should be connected to CODEC_RX_STRB
31	CODEC_RX_DATA	Input	PCM Data from A/D Conveter to AMBE-2000

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Pin Number	Pin Descriptive Name	Pin Direction	Notes
41	CODEC_TX_DATA	Output	PCM Data from AMBE-2000 to D/A Conveter
27	CODEC_RX_CLK	Input	A/D Serial clock. Should be connected to CODEC_TX_CLK
33	CODEC_TX_CLK	Input	D/A Serial clock Should be connected to CODEC_RX_CLK
51	CLOCK_MODE	Input	If high enables crystal oscillator option for clock source. If low then external oscillator option is selected. See Section 3.4 for details.
8,11,12,23 36,39,44,45 46, 47,48 49,54,57,64 76,87,90	V$_{DD}$	Power	Supply Voltage
1,9,10,25,26 35,40,50,52 53,56,63,65 88,89	GND	Power	Ground
2,3,4,5,6,7, 13,14,15,16 17,18,19,21 22,24,43,55 58,59,60,61 62,66,91,92 93,94,95,96 97,98,99 100	No Connection		These pins must remain unconnected

Note 1: The AMBE-2000 expects an encoder packet to be read approximately every 20 msec. Following the initial reset, wait for EPR to go low and read the initial packet (t0). 20 msec later, the next packet (t1) should be ready. For packet t1 and all following packets use the procedure below:

1) Wait for slightly less than 20 msec.
2) Assert CHAN_TX_STB and read word on CHAN_TX_DATA.
3) If transmitted word not 0x13EC, discard it and repeat step 2.
4) If transmitted word 0x13EC, read 23 more words (rest of packet).

A new packet should be ready every 20 msec after the initial EPR high to low transition.   A packet read should take place every 20 msec. If there is a delay in the read (i.e. a packet is missed), it is recommended that the device be reset.

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3.3   Clock and Reset Timing

To reset the AMBE-2000 chip, the reset signal must be held low for a minimum of six clock cycles. The recovery time from reset is approximately 95 msec. In other words, 95 msec after the rising edge of the reset signal the AMBE-2000 starts processing PCM samples. The first packet will be ready after 252 PCM samples are read.

Figure 3-A X2/CLKIN and CLKOUT Timing Diagram
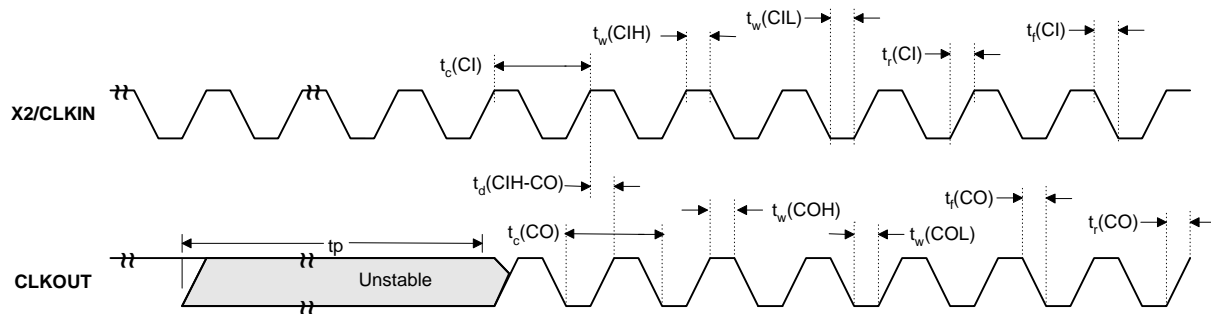
Table 3-A X2/CLKIN and CLKOUT Timing Parameters

Reference	Parameter		Min	Max	Units
$t_c(CI)$	Cycle time, X2/CLKIN	Integer PLL multiplier N (N=4)	20	400	ns
$t_f(CL)$	Fall time, X2/CLKIN			4	ns
$t_r(CL)$	Rise time, X2/CLKIN			4	ns
$t_w(CIL)$	Pulse duration, X2/CLKIN low		6		ns
$t_w(CIH)$	Pulse duration, X2/CLKIN high		6		ns
t_p	Transitory phase, PLL lock-up time			50	μS
$t_c(CO)$	Cycle time, CLKOUT (typical is $t_c(CI)$/4		15		ns
$t_d(CIH-CO)$	Delay time, X2/CLKIN high/low to CLKOUT high/low		4	16	ns
$t_f(CO)/t_r(CO)$	Fall/Rise time, CLKOUT (typical is 2 ns)				
$T_w(COL)$	Pulse duration, CLKOUT low		H-4	H	ns
$T_w(COH)$	Pulse duration, CLKOUT high		H-4	H	ns

- CLKOUT is shown for reference only. It is not to be connected in the circuit.

- H = 7.629 ns

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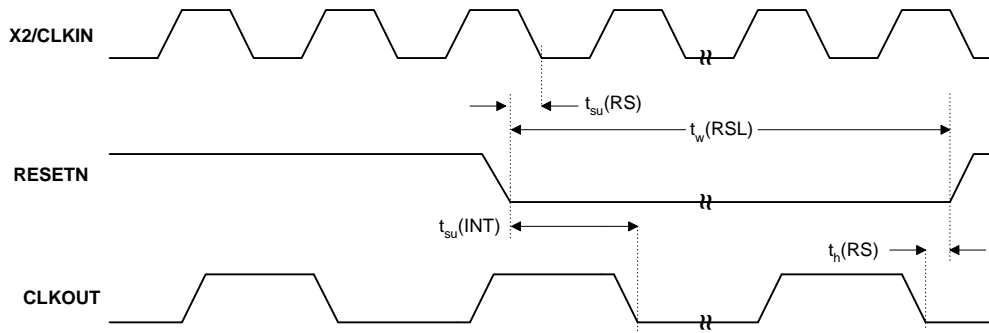
Figure 3-B Hardware Reset Timing Diagram

Table 3-B Reset Timing Parameters

Reference	Parameter	Min	Max	Units
t_h(RS)	Hold time, RS after CLKOUT low	0		ns
t_w(RSL)	Pulse duration, RS low	50		µS
t_{su}(RS)	Setup time, RS before X2/CLKIN low	5		ns
t_{su}(INT)	Setup time, INTn, NMI, RS before CLKOUT low	10		ns

- CLKOUT is shown for reference only. It should not be connected in the circuit.

3.4   Crystal / Oscillator Usage

The AMBE-2000™ Vocoder Chip has an input clock frequency of 16.384 MHz. Two options are outlined below in providing this signal. The option used is specified by the CLOCK_MODE pin 51.

The following points should be noted when designing any printed circuit board layout:

- Keep the crystal and external capacitors as close to the CLK_I and CLK_I2 pins as possible to minimize board stray capacitance.

- Keep X2/CLKIN and X1 away from high frequency digital traces (example CLKOUT) to avoid coupling.

3.4.1 TTL Clock Source

If CLOCK_MODE pin is low then a TTL/CMOS source is used as the clock input. Connect X2/CLKIN and X1 as follows:

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Figure 3-C X2/CLKIN and X1 with TTL Clock Source
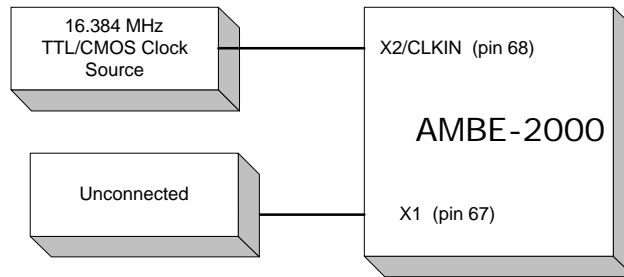
3.4.2 Crystal Oscillator

The Crystal Oscillator option is selected with CLOCK_MODE pin set to a high level. To use the crystal oscillator, connect the crystal across X2/CLKIN and X1 along with one external capacitor from each of these pins to ground. Recommended values for C1 and C2 is 10 pF.

Figure 3-D X2/CLKIN and X1 with Crystal Oscillator

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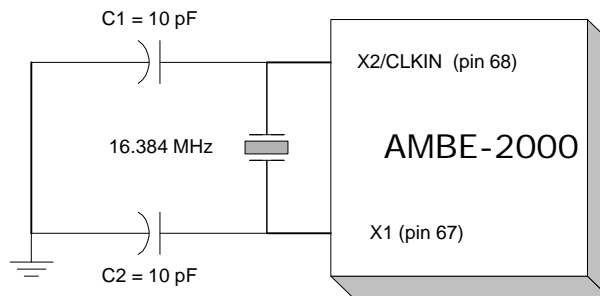
3.5   Package Description

100 pin TQFP (Thin Quad Flat Pack)
All Dimensions are in millimeters

Figure 3-E Package Dimensions

Not Drawn to Scale

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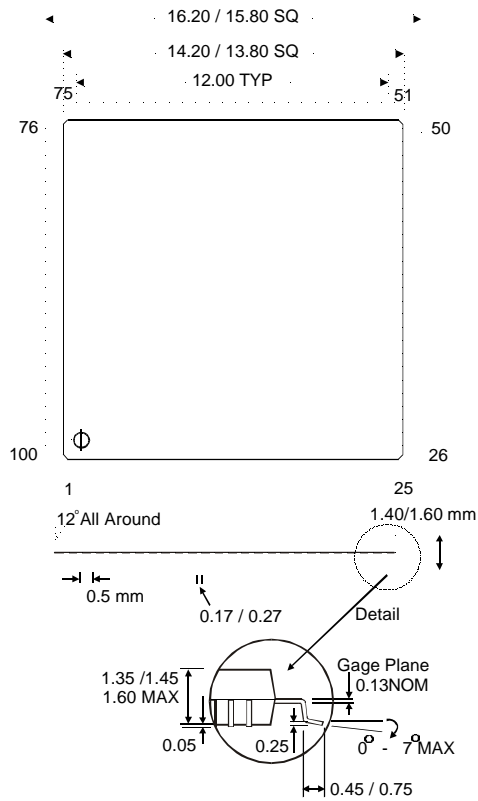
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3.6   Normal Operating Conditions

Table 3-C Normal Operating Conditions

Normal Operating Conditions	
Operating Voltage	3.3V
Operating Case Temperature Range	-40°C to 100°C
Storage Temperature Range	-55°C to 150°C

3.7   Absolute Maximum Ratings

Stresses in excess of the Absolute Maximum Ratings can cause permanent damage to the device. These are absolute stress ratings only. Functional operation of the device is not implied at these or any other conditions in excess of those given in the operational sections of the data sheet. Exposure to Absolute Maximum Ratings for extended periods can aversely affect device reliability.

Table 3-D Absolute Maximum Ratings

Absolute Maximum Ratings	
Voltage Range on any Pin with Respect to Ground	-0.3V to 4.6V

3.8   Electrical Characteristics and Requirements

Table 3-E Recommended Operating Conditions

Parameter		Min	Nom	Max	Unit
DV_{DD}	Device Supply Voltage	3	3.3	3.6	V
V_{SS}	Supply Voltage, GND	-	0	-	V
V_{IH}	High-level input voltage, I/O	2.5	-	$DV_{DD} + 0.3$	V
V_{IL}	Low-level input voltage	-0.3	-	0.8	V
I_{OH}	High-level output current	-	-	-300	μA
I_{OL}	Low-level output current	-	-	1.5	mA

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Table 3-F Electrical Characteristics over Recommended Operating Case Temperature Range (Unless Otherwise Noted)

Parameter	Test Conditions	MIN	TYP†	MAX	Unit
V_{OH} High-level output voltage‡	$V_{DD} = 3.3 \pm V$, $I_{OH} = MAX$	2.4			V
V_{OL} Low-level output voltage‡	$I_{OL} = MAX$			0.4	V
I_I Input current in high impedance ($V_I = V_{SS}$ to V_{DD})	$V_{DD} = MAX$, $V_I = V_{SS}$ to V_{DD}	-10		10	μA
C_I Input capacitance			10		pF
C_O Output capacitance			10		pF

† All values are typical unless otherwise specified.

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4   Channel Interface

4.1   Overview

The Channel Interface is the general term used for the interface for the compressed bits coming from the encoder and the compressed bits going to the decoder. This same interface is also used to output *status* information from the encoder and decoder such as whether a DTMF tone has just been detected in the speech input, or whether the decoder has detected and synthesized a frame of silence. Additionally, this interface is used to perform more complex control operations on both the encoder and decoder (usually at start-up). These control functions include speech data and FEC rate control as well as enabling features such as echo cancellation.

It is important to realize that not all data being output from the AMBE-2000 is intended for transmission over the channel. Status type of data is typically only useful at the 'local' end. In most voice transmission systems, the actual encoded bits are extracted from the channel formatting, combined into the systems transmission stream, sent over the transmission path, extracted from the transmission path at the receiving end, and reassembled into the AMBE-2000's channel format for synthesis by the decoder.

Figure 4-A Channel Interface Overview
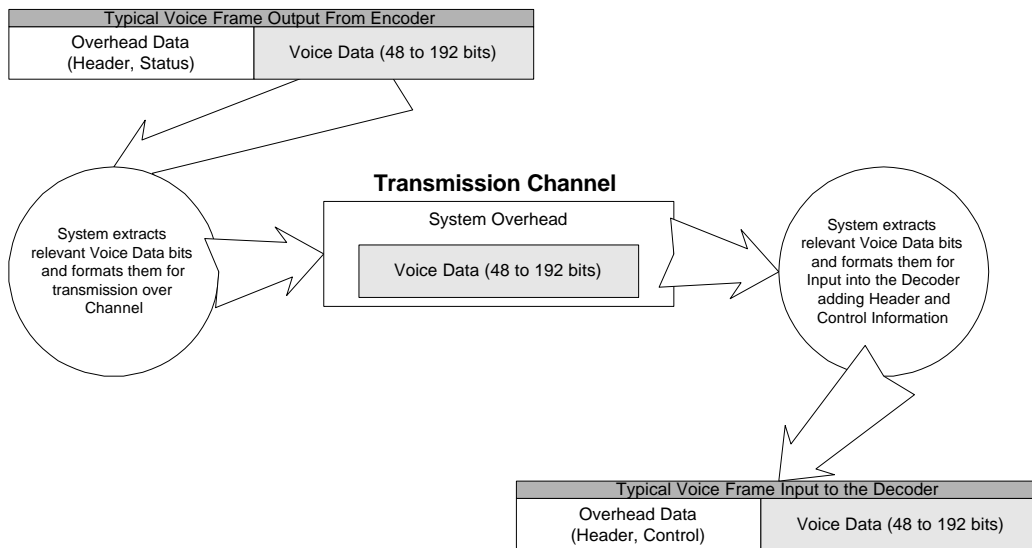
4.2   Serial Configuration Selection

The hardware interface to the Channel Interface is configured as a serial interface based exclusively on the hardware settings of CHANN_SEL[1-0]. See Table 4-A.

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Table 4-A Channel Interface Selection Table

Port Type	Interface Select Pins	
	CH_SEL1 (pin 77)	CH_SELO (pin 75)
Active Formatted	0	0
Active Unformatted	0	1
Passive Formatted	1	0
Passive Unformatted	1	1

Table 4-B Unformatted Bit per Word Selection Table

Number of Voice Data Bits per Word	Interface Select Pins	
	BAUD_SEL1 (pin 81)	BAUD_SELO (pin 80)
1	0	0
2	0	1
3	1	0
4	1	1

All transfers occur through a serial port. The serial port inputs and outputs a 16 bit word for every write and read strobe signal respectively. Serial mode can be **formatted** or **unformatted**. Within the **unformatted** mode, the data is input and output in 16 bits words still but with only 1 to 4 voice data bits carried within each word. These four configurations can be seen in Table 4-B. See section 4.3 for all the details on the serial interface.

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4.3   Channel Serial Mode

The signals in Table 4-C make up the serial channel interface. The serial channel mode transfers data in and out of the AMBE-2000™ using 16 bit words on the two data lines **CHAN_RX_DATA** and **CHAN_TX_DATA**.   The selection of the **formatted** or **unformatted** format of this data is made using information in Table 4-A.

Table 4-C Channel Serial Interface Pin Descriptions

Pin Symbol	Pin Direction	Pin Number	Description
EPR	Out	20	**Encoder Packet Ready**: Following a reset, this output signal will have a high to low transition to indicate that the encoder has a frame of data to output. The next packet will be ready approximately 20 msec later. **See Note 1.**
CHAN_RX_DATA	In	32	**Serial Data Input**: 16 bits of channel data are input on **CHAN_RX_DATA**, synchronous to **CHAN_RX_CLK**, with each **CHAN_RX_STRB** pulse.
CHAN_RX_CLK	In	28	**Serial Input Clock**: In coordination with **CHAN_RX_STRB**, **CHAN_RX_DATA** is latched by the AMBE-2000™ on the falling edges of **CHAN_RX_CLK**.
CHAN_RX_STRB	In	30	**Input (Write) Data Strobe**: This signal indicates to the AMBE-2000™ when the data on **CHAN_RX_DATA** will be latched by **CHAN_RX_CLK**. See Figure 4-B.
CHAN_TX_DATA	Out	42	**Serial Data Output**: 16 bits of channel data are output on **CHAN_TX_DATA**, synchronous to **CHAN_TX_CLK**, with each **CHAN_TX_STRB** pulse.
CHAN_TX_CLK	In	34	**Serial Output Clock**: In coordination with **CHAN_TX_STRB**, the data on **CHAN_TX_DATA** is output by the AMBE-2000™ on the falling edges of **CHAN_TX_CLK**.
CHAN_TX_STRB	Active Output/ Passive Input	38	**Output (Read) Data Strobe**: This signal indicates to the AMBE-2000™ when to bring the data to the **CHAN_TX_DATA** pin. See Figure 4-B.

Note 1: The AMBE-2000 expects an encoder packet to be read approximately every 20 msec. Following the initial reset, wait for EPR to go low and read the initial packet (t0). 20 msec later, the next packet (t1) should be ready. For packet t1 and all following packets use the procedure below:

1) Wait for slightly less than 20 msec.
2) Assert CHAN_TX_STB and read word on CHAN_TX_DATA.
3) If transmitted word not 0x13EC, discard it and repeat step 2.
4) If transmitted word 0x13EC, read 23 more words (rest of packet).

A new packet should be ready every 20 msec after the initial EPR high to low transition.   A packet read should take place every 20 msec. If there is a delay in the read (i.e. a packet is missed), it is recommended that the device be reset.

4.3.1 Low Level Timing for Passive and Active Serial Mode

Figure 4-B Low Level Timing for Passive and Active Serial Mode
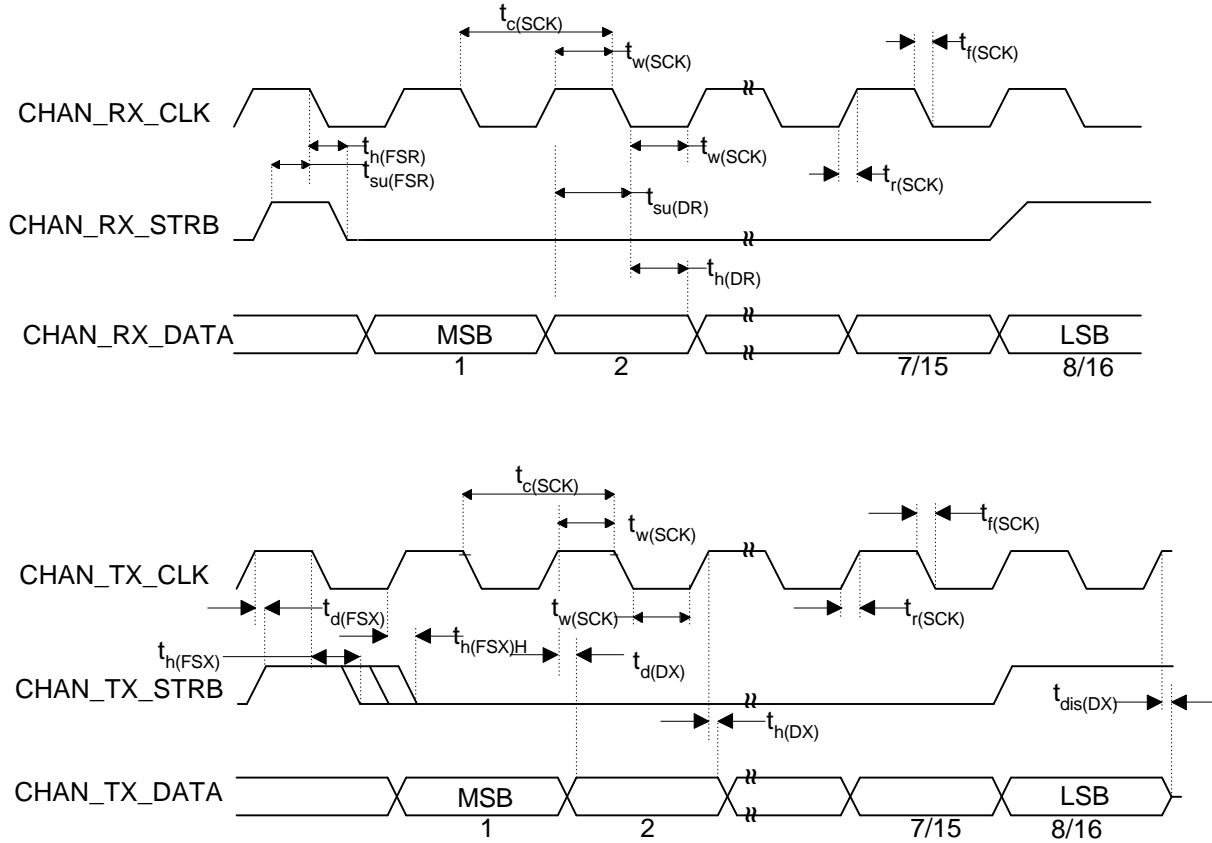
Table 4-D Switching Characteristics Over Recommended Operating Conditions for Serial Port Receive

Reference	Parameter	3.3 Volts		Units
		Min	Max	
$t_{h(FSR)}$	Hold time, FSR after CLKR falling edge	6		ns
$t_{h(DR)}$	Hold time, DR after CLKR falling edge	6		ns

Table 4-E Switching Characteristics Over Recommended Operating Conditions for Serial Port Receive

Reference	Parameter	3.3 Volts		Units
		Min	Max	
$t_{d(SCK)}$	Delay time, DX valid after CLKX rising (Passive Mode)		25	ns
$t_{c(SCK)}$	Cycle time, serial port clock	6H		ns
$t_{f(SCK)}$	Fall time, serial port clock		6	ns
$t_{r(SCK)}$	Rise time, serial port clock		6	ns
$t_{w(SCK)}$	Pulse duration, serial port clock low/ high	3H		ns
$t_{su(FSR)}$	Setup time, FSR before CLKR falling edge	6		ns
$t_{su(DR)}$	Setup time, DR before CLKR falling edge	6		ns
$t_{d(FSX)}$	Delay time, CLKX rising to FSX		15	ns
$t_{d(DX)}$	Delay time, CLKX rising to DX (Active Mode)		15	ns
$t_{h(FSX)}$	Hold time, FSX after CLKX falling edge	6		ns
$t_{h(FSX)H}$	Hold time, FSX after CLKX rising edge		2H-5	ns
$t_{dis(DX)}$	Disable time, CLKX rising to DX		20	ns
$t_{h(DX)}$	Hold time, DX valid after CLKX rising edge	-5		ns

- Note: H = 7.629 ns; however, do not operate serial clocks any faster than 2.048 MHz. Thus $t_{c(SCK)}$ should be a minimum of 488.3 nsec.

5 Channel Data Format

The channel interface is responsible for outputting the compressed data from the encoder and inputting compressed data to the decoder. In addition to these most basic functions the channel interface is also capable of reporting certain events, such as the detection of a DTMF tone. The channel interface can also control certain selectable functions of the AMBE-2000™, such as the voice coding rate. This chapter will describe how the AMBE-2000™ uses the channel interface to multiplex these capabilities.

There are two formats to the data, **Formatted** and **Unformatted**, both of which operate in serial mode. Generally speaking the **Unformatted** mode is used only when the connection between the AMBE-2000™ and the channel under design is relatively direct, and the designer wants to simplify the extraction of the relevant voice data. In this mode configuration is accomplished using hardwired pins. In most cases, when a controller is present between the AMBE-2000™ and the channel, the system designer will find that using the **Formatted** format is more flexible.

5.1 Formatted Format

The **Formatted** format is a 24 by sixteen-bit word format for a total of 48 bytes or 384 bits. Every 20 milliseconds the encoder outputs 24 sixteen-bit words, and likewise the decoder expects to receive 24 words. The format of the input and output frames are detailed below. The first 12 sixteen bit words are made up of header, ID and status or control information. The remaining 12 sixteen bit words make up the encoded data bit field. These 12 words, or 192 bits, will be fully populated with relevant voice data only when the AMBE-2000 is operating in a 9600bps mode (9600 bits/sec ÷ 50 frames/sec = 192 bits/frame). Otherwise, when the data rate is less than 9600bps, the coded voice bits are filled starting from the MSB of the first word in the field, leaving any unused bits as zeros. It is important to note here that even when the AMBE-2000 is operating at less than 9600bps, *all* 384 bits of the **Formatted** format (including any unused trailing zeros) must be transferred out of the encoder and into the decoder.

5.2 Formatted Input Format

The format of the **Formatted** input is shown in Figure 5-A. Keep in mind that even though the channel data in this **Formatted** input format is closely associated with the decoder, the control information will apply to both encoder *and* decoder functions.

Figure 5-A Basic Formatted Input Format

			Word #	Description	
20 ms frame	24 sixteen-bit words = 48 bytes = 384 bits	(12) 16 bit words of overhead (192 bits)	0	Header always set to 0x13EC	
			1	Power Control (8bits)	Control Word 1 (8 bits) – see Table 5-B
			2	Rate info 0	
			3	Rate info 1	
			4	Rate info 2	See Tables 5-C and 5-D
			5	Rate info 3	
			6	Rate info 4	
			7	Unused in Input	
			8	Unused in Input	
			9	Unused in Input	
			10	DTMF Control – see Tables 5-E and 5-F	
			11	Control Word 2 – see Table 5-G	
		(12) 16 bit words of data (192 bits)	12	Channel Data	
			13	Channel Data	
			14	Channel Data	
			15	Channel Data	
			16	Channel Data	
			17	Channel Data	
			18	Channel Data	
			19	Channel Data	
			20	Channel Data	
			21	Channel Data	
			22	Channel Data	
			23	Channel Data	

5.2.1 Formatted Input : Word 0 : Header

The decoder uses the header information to synchronize with the beginning of each 20 millisecond frame. this 16 bit word **MUST** be **0x13EC**.

5.2.2 Formatted Input : Word 1 : Power Control

Set the encoder to **0x00** in the 8-bit Power Control field of an output frame for normal use. For Power Down Mode, set this value to **0x55**. This causes the AMBE-2000 to enter low power mode. To exit low power mode, the device must be reset through the hardware.

Table 5-A Formatted Input : ID Values Summary

ID	Type	Description
0x00	Voice Data	This ID value instructs AMBE-2000™ to operate in normal fashion
0x55	Low Power Mode	When this mode is activated the AMBE-2000™ Vocoder Chip will go into a mode which conserves power, where no voice packets are being processed.

5.2.3 Formatted Input : Word 1 : Control Word 1

Use the 8-bit control word to set various functions.

Table 5-B Control Word 1 Format

Control Word 1 – 8-bits (See Table 1-F)							
7: MSB	6	5	4	3	2	1	0 : LSB
Lost Frame Indicator	Unused	Unused	Unused	Unused	Unused	**CNI**	Unused

Lost Frame Indicator : Setting the Lost Frame Indicator bit to a 1 will cause the AMBE-2000™ decoder to construct the voice frame using the parameters from the previous frame. This is an effective way to mask the effects of short periods of data loss. This bit should be set by the user when channel data is lost or corrupted. It works by replacing the frame of corrupted data with the previous, intact, frame.

Comfort Noise Insertion (CNI) : Setting the CNI bit will cause the decoder to output a frame of comfort noise. This bit is used with systems that are capable of discontinuous transmission (DTX).

5.2.4 Formatted Input : Words 2-6 : Rate Info 0, Rate Info 1, Rate Info 2, Rate Info 3, Rate Info 4

The initial rate of the AMBE-2000 is set through the hardware pins RATE_SEL[4-0] (see Section 7.2 and Tables 7-A and 7-B) after resetting the device. The coding rate can then be modified for both the encoder and the decoder through the Unformatted channel interface. The Rate Selector control field (describe in section) determines where to apply the rate change.

The AMBE-2000 used these five words to set the source and FEC coding rates. Tables 5-C and 5-D list predefined value for various source and FEC rates. The software configurations in Table 4-C are compatible for use with the AMBE-1000™. If compatibility is not an issue, use the software codes in Table 4-D to select speech and FEC rates to optimize use of the AMBE-2000™.

Table 5-C Rate Selection Using Rate Info 0-4, compatible w/ AMBE-1000™

Rate Info 0	Rate Info 1	Rate Info 2	Rate Info 3	Rate Info 4	Speech Rate (bps)	FEC Rate (bps)	Total Rate (bps)
0x9030	0x0000	0x0000	0x0000	0x4330	2400	0	2400
0x902f	0x0000	0x0000	0x0000	0x6930	2350	50	
0x9348	0x0000	0x0000	0x0000	0x6f48	3600	0	3600
0x9243	0x0080	0x0000	0x0000	0x5348	3350	250	
0xab50	0x0000	0x0000	0x0000	0x3950	4000	0	4000
0x934b	0x0080	0x0000	0x0000	0x3950	3750	250	
0xab60	0x0000	0x0000	0x0000	0x7960	4800	0	4800
0xab5b	0x0080	0x0000	0x0000	0x6860	4550	250	
0x9348	0x2030	0x0000	0x0000	0x7060	3600	1200	
0x923e	0x2800	0x0000	0x0000	0x7460	3100	1700	
0xab53	0x2c00	0x0000	0x0000	0x5680	4150	2250	6400
0xab58	0x3000	0x0000	0x0000	0x4490	4400	2800	7200
0xbf9b	0x0080	0x0000	0x0000	0x49a0	7750	250	8000
0xab5d	0x3400	0x0000	0x0000	0x31a0	4650	3350	
0xbfc0	0x0000	0x0000	0x0000	0x72c0	9600	0	9600
0xab16	0xe400	0x0000	0x0000	0x67c0	4850	4750	

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Table 5-D Rate Selection Using Rate Info 0-4, AMBE-2000™ only

Rate Info 0	Rate Info 1	Rate Info 2	Rate Info 3	Rate Info 4	Speech Rate (bps)	FEC Rate (bps)	Total Rate (bps)
0x0028	0x0000	0x0000	0x0000	0x6428	2000	0	2000
0x5048	0x0000	0x0000	0x0000	0x3948	3600	0	3600
0x5250	0x0000	0x0000	0x0000	0x4150	4000	0	4000
0x1030	0x0001	0x0000	0x341a	0x6750	2400	1600	
0x5360	0x0000	0x0000	0x0000	0x6c60	4800	0	4800
0x5250	0x2010	0x0000	0x0000	0x7460	4000	800	
0x5048	0x0001	0x0000	0x2412	0x6860	3600	1200	
0x1030	0x0005	0x180c	0x3018	0x7360	2400	2400	
0x6b80	0x0000	0x0000	0x0000	0x6c80	6400	0	6400
0x5250	0x0001	0x0000	0x542a	0x5280	4000	2400	
0x5258	0x0009	0x1e0c	0x4127	0x7390	4400	2800	7200
0x7fa0	0x0000	0x0000	0x0000	0x52a0	8000	0	8000
0x5250	0x0005	0x2010	0x6834	0x72a0	4000	4000	
0x7fc0	0x0000	0x0000	0x0000	0x69c0	9600	0	9600
0x5048	0x000e	0x4010	0x6a2e	0x65c0	3600	6000	
0x1030	0x000e	0x681a	0x511b	0x76c0	2400	7200	

5.2.5 Formatted Input : Word 7 : Unused in Input

Should be set to 0x0000

5.2.6 Formatted Input : Word 8 : Unused in Input

Should be set to 0x0000

5.2.7 Formatted Input : Word 9 : Unused in Input

Should be set to 0x0000

5.2.8 Formatted Input : Word 10 : DTMF Control

Use this word to set DTMF tones. See Table 5-F for a list of tones and their corresponding values.

Table 5-E DTMF Control Format

DTMF Control – 16-bits															
15: MSB	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0: LSB
DTMF Amplitude								DTMF Digit Detect/ Generate							

Table 5-F DTMF Codes for Digit Detect/ Generate

DTMF Code	DTMF Digit
0x80	1
0x84	2
0x88	3
0x81	4
0x85	5
0x89	6
0x82	7
0x86	8
0x8A	9
0x87	0
0x83	*
0x8B	#
0x8C	A
0x8D	B
0x8E	C
0x8F	D
0xff	deactivate

To deactivate the Digit Detect/Genetrate function, set the DTMF Code to 0xff. Dial, ring, and busy tones are standard North American call progress tones.

DTMF Amplitude

The DTMF Amplitude runs from 3 to –60 dBm0. This value is a signed bite (example: 0x03 = 3, 0x00 = 0, 0xC4 = -60).

5.2.9 Formatted Input : Word 11 Control Word 2

Table 5-G Control Word 2 Format

Control_Word 2 – 16-bits (See Table 1-F)															
Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
Decoder Output Volume Control								Unused 0	Unused 0	VAD	Unused 0	SL	EC	RIS	

Rate Information Selector (RIS) :
Use these 2 bits to select which part(s) of the vocoder will be affected by the rate selection words.

Table 5-H Rate Information Selection Codes

value	Area Controlled
0x0	Encoder and Decoder
0x1	Encoder only
0x2	Decoder only
0x3	Neither Encoder nor Decoder

Echo Canceller (EC) :
Set this bit to 1 to enable the echo cancel function on a frame by frame basis. This must be set to 1 in every packet sent to continue use.

Sleep (SL) :
Set this bit to 1 to enter Sleep mode on a frame by frame basis. Sleep is a low power mode, not to be confused with Power Down Mode. The sleep function must be enabled in every packet to continue in sleep mode. Set this bit to 0 to exit Sleep mode.

VAD :
In order to set the Voice Activity Detector on, set the VAD bit to 1. To disable Voice Activity Detection, set the VAD bit to 0.

5.2.10 Formatted Input : Words 12-23 : Channel Data

This is the field that contains the actual coded bits. Input of the data begins with the MSB of the first word in this field and continues through with the final bit output being the LSB of the final word. If the data rate selected is less than 9600bps then the unused bits in each frame are zero and populate the end of the field. As is noted in the Channel Interface definitions, these unused bits must still be clocked out of the AMBE-2000™. The packet must always consist of 24 words.

5.3   Formatted Output Format

The format for Formatted output data is shown in Figure 5-B. It is only the bits in the Voice Data Bits field, which are transmitted along with framing information (data used to locate the start of each frame for proper reconstruction at the decoder) over the channel. The first 192 bits provide overhead information, which is sometimes useful to the host but is generally not transmitted over the channel.

Figure 5-B Basic Formatted Output Format

Word #	Description	
0	Header always set to 0x13ec	
1	Power Control (8bits)	Control Word 1 (8 bits) – see Table 5-J
2	Rate info 0	
3	Rate info 1	
4	Rate info 2	
5	Rate info 3	
6	Rate info 4	
7	Bit Error Rate	
8	Unused	
9	Unused	
10	DTMF Control – see Table 5-L	
11	Control Word 2 – see Table 5-N	
12	Channel Data	
13	Channel Data	
14	Channel Data	
15	Channel Data	
16	Channel Data	
17	Channel Data	
18	Channel Data	
19	Channel Data	
20	Channel Data	
21	Channel Data	
22	Channel Data	
23	Channel Data	

(20 ms frame; 24 sixteen-bit words = 48 bytes = 384 bits; (12) 16 bit words of overhead (192 bits); (12) 16 bit words of data (192 bits))

5.3.1 Formatted Output : Word 0 : Header

The header is a 16 bit word that begins each valid frame corresponding to 20 milliseconds of speech. This field will always be **0x13EC**.

5.3.2 Formatted Output : Word 1 : ID (Power Control)

The encoder will always use **0x00** in the 8-bit field of an output frame.

5.3.3 Formatted Output : Word 1 :Control Word 1

This 8-bit control word indicates the activity of various functions.

Table 5-J Control Word 1 Format

Control Word 1 – 8-bits (See Table 1-A)							
7: MSB	6	5	4	3	2	1	0 : LSB
Unused	Unused	Decoder Frame Repeat	Decoder Silence Detect	Unused	Unused	Encoder Silence Detect	Encoder DTMF Detect

Decoder Frame Repeat: When the Decoder Frame Repeat flag is set to 1, the decoder is reporting that the last frame decoded was a repeat of the previous frame.

Decoder Silence Detect: When the Decoder Silence Detect flag is set to 1, the decoder is reporting that the last frame decoded was a comfort noise frame.

Encoder DTMF Detect: The Encoder DTMF Detected Flag will be set to a 1 when the encoder detects a DTMF tone. The DTMF Detect option is controlled by the DTMF Detect/Generate bits in control Word 10 described in section 5.2.8 enable the DTMF Detect option.

Silence Detect: The Encoder Silence Detected Flag will be set to 1 when no voice activity is detected. The Silence Detect option is controlled by and can be disabled by the VAD as described in section 5.2.9.

5.3.4 Formatted Output : Words 2-6 : Rate Info 0, Rate Info 1, Rate Info 2, Rate Info 3, Rate Info 4.

Words 2-6 in the packet indicate the rate at which the AMBE-2000™ encoder is operating . These words are output. See tables 5-C and 5-D for corresponding values.

5.3.5 Formatted Output: Word 7 : Bit Error Rate

This status field is used for the decoder to report bit error information. The 16 bit number output is used to compute the BER using the following calculation

$$\% \ BER = (Word \ 7) / (32768)$$

5.3.6 Formatted Output : Word 10 : DTMF Control

This word corresponds to the DTMF Detection capabilities of the vocoder. It uniquely identifies specific tones recognized by the encoder. See table 5-J for a list of tones and their corresponding values.

Table 5-K DTMF Tone Detection Parameters

DTMF Tone Detection Requirement	Value	Description
Minimum Input Level	-25 dBm0	An input signal shall not be rejected as a DTMF tone if its amplitude is greater than -25 dBm0 (maximum sinusoid dBm0 is defined as+3.17 dBm0).
Minimum Signal to Noise Distortion ratio	15 dB	In order for an input signal to correspond to a valid DTMF tone, the ratio of inband to out-of-band energy must be greater than 15dB. Inband energy is defined to be the energy in frequency components within ±3.5% of the two frequencies defined by the DTMF frequencies. Out-of-band energy is defined to be the total energy minus in the inband energy.
Minimum Frequency Tolerance	±1.5%	An input signal shall not be rejected as a DTMF tone if both of its principal frequency components are within ±1.5% of the frequencies needed for the DTMF tone.
Maximum Frequency Tolerance	±3.5%	An input signal shall not be rejected as a DTMF tone if either of its principal frequency components are outside ±3.5% of the frequencies needed for the DTMF tone.
Normal Twist Range	8-10 dB	An input signal does not correspond to a valid DTMF tone if the energy contained within the low frequency band is more than 10 dB greater than the energy contained in the high frequency band. An input signal shall not be rejected as a DTMF if energy contained within the low frequency band is less than 8 dB greater than the energy contained in the high frequency band. Each low and high frequency band is limited to ±3.5% of the frequencies needed for the DTMF tone.
Reverse Twist Range	4-10 dB	An input signal does not correspond to a valid DTMF tone if the energy contained within the high frequency band is more than 10 dB greater than the energy contained in the low frequency band. An input signal shall not be rejected as a DTMF if energy contained within the high frequency band is less than 4 dB greater than the energy contained in the low frequency band. Each low and high frequency band is limited to ±3.5% of the frequencies needed for the DTMF tone.
Minimum Tone Duration	45 mS	An input signal shall not be rejected as a DTMF tone as long as its time duration is greater than 45 mS. In addition a minimum of two frames will be transmitted of the DTMF tone if a valid tone is detected. The duration of a tone is defined by the points at which the envelope is 20 dB below its peak value.

Table 5-L DTMF Control Format

DTMF Control – 8-bits (See Table 1-F)															
15: MSB	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0: LSB
DTMF Amplitude								DTMF Digit Detect/ Generate							

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Table 5-M DTMF Codes for Digit Detect/ Generate

DTMF Code	DTMF Digit
0x80	1
0x84	2
0x88	3
0x81	4
0x85	5
0x89	6
0x82	7
0x86	8
0x8A	9
0x87	0
0x83	*
0x8B	#
0x8C	A
0x8D	B
0x8E	C
0x8F	D
0xff	deactivate

To deactivate the Digit Detect/Genetrate function, set the DTMF Code to 0xff. Dial, ring and busy tones are standard North American call progress tones.

DTMF Amplitude

The DTMF Amplitude runs from 3 to –60 dBm0. This value is a signed bite (example: 0x03 = 3, 0x00 = 0, 0xC4 = -60).

5.3.7 Formatted Output : Word 11 : Control Word 2

Table 5-N Control Word 2 Format

Control_Word 2 – 16-bits (See Table 1-A)															
Bit 15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
Decoder Output Volume Control								Unused 0	Unused 0	Unused 0	Unused 0	SL	EC	RIS	

Rate Information Selector (RIS) :

This field is used to indicate which part of the device is affected by words 2-6, the rate control words.

Table 5-O Rate Information Selection Codes

value	Area Controlled
0x0	Encoder and Decoder
0x1	Encoder only
0x2	Decoder only
0x3	Neither Encoder nor Decoder

Echo Canceller (EC) :
A 1 in this output field indicates that the echo canceller has been enabled.

Sleep (SL) :
A 1 in the this field indicates the device has been put into sleep mode.

Decoder Output Volume Control : Indicates the current decoder volume.

5.3.8   Formatted Output : Words 12-23 : Channel Data

This is the field that contains the actual coded bits. Output of the data begins with the MSB of the first word in this field and continues through with the final bit output being the LSB of the final word. If the data rate selected is less than 9600bps then the unused bits in each frame are zero and populate the end of the field. As is noted in the Channel Interface definitions, these unused bits must still be clocked out of the AMBE-2000™. The packet must always consist of 24 words.

5.4   Unformatted Serial Format

The Unformatted Format for the channel data is useful for applications which desire minimal glue logic between the AMBE-2000™ and the channel hardware. The use of minimal hardware in place of a microcontroller can be realized using this data format. Another distinct difference in this data format is that framing information (data which carries the positional information relating to the coded bits) is embedded into the data stream itself. Using this data format, the system designer need only transfer the coded data itself. A single bit each frame is 'borrowed' from the voice data to embed the framing information. Keep in mind that this 'borrowed' bit reduces the effective voice coding rate quality by 50 bits per second. For example, a system with no FEC running at 2450 bps in Unformatted mode will sound equivalent to one running at 2400 bps in Formatted mode.

The designer should also be aware that it takes approximately 15 frames (300 milliseconds) for the decoder to attain synchronization with the incoming stream before it can output synthesized speech. Systems which are attempting to save power by shutting down transmission during periods of silence, and then resuming during periods of speech can not handle this 300 millisecond delay for each synchronization, and thus should use Formatted mode with a more sophisticated framing method.

The 16 bit per word format, pictured in Section 5.2, is maintained in this mode but only a fraction of the full 16 bits is used to transfer the coded data. The user selects whether 1, 2, 3 or 4 bits will be transferred in each word based on pins BAUD_SEL[0:1] and Table 4-A. **IMPORTANT** : The voice coding data rate selected must be evenly divisible by the number of voice data bits per word selected.

5.4.1   Unformatted Serial Output Format

The Unformatted output format contains 1 to 4 bits within each 16 bits serial output word. The formats which contain more than one bit each word the MSB of the data bits is considered first in the transmission. In Unformatted mode, only the coded voice data bits are output. None of the superfluous information that exists in formatted mode is available in this mode. The number of words that need to be transferred out of the encoder for each 20 millisecond frame will be the number of bits per frame divided by the number of bits per word. So a system coding at 4800 bps with 3 bits per word will need to read 32 ($[4800 \div 50] \div 3 = 32$) words each frame.

Table 5-P Unformatted Serial Output Data Format

Bits per Word See **Table 4-A**	Data				Unused											
	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
1 bit per Word Format	**D** msb	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
2 bits per Word Format	**D** msb	**D**	0	0	0	0	0	0	0	0	0	0	0	0	0	0
3 bits per Word Format	**D** msb	**D**	**D**	0	0	0	0	0	0	0	0	0	0	0	0	0
4 bits per Word Format	**D** msb	**D**	**D**	**D**	0	0	0	0	0	0	0	0	0	0	0	0

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5.4.2 Unformatted Serial Input Format

The Unformatted Input format contains 1 to 4 bits within each 16 bits serial output word. For the formats which contain more than one bit each word the MSB of the data bits is considered first in the transmission. In Unformatted mode, the header data from Formatted mode is dropped and each 16 bit write contains 1 to 4 coded voice data bits. The number of words that need to be transferred into the decoder for each 20 millisecond frame will be the number of bits per frame divided by the number of bits per word. So a system coding at 4800 bps with 3 bits per word will need to write exactly 32 ([4800 ÷50] ÷3 = 32) words each frame.

Table 5-Q Unformatted Serial Input Data Format

Bits per Word See **Table 4-A**	Data				Control Offset				Control Data							
	15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0
1 bit per Word Format	**D** msb	0	0	0	0	0	0	X	X	X	X	X	X	X	X	X
2 bits per Word Format	**D** msb	**D**	0	0	0	0	0	X	X	X	X	X	X	X	X	X
3 bits per Word Format	**D** msb	**D**	**D**	0	0	0	0	X	X	X	X	X	X	X	X	X
4 bits per Word Format	**D** msb	**D**	**D**	**D**	0	0	0	X	X	X	X	X	X	X	X	X

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6   A/D-D-A Interface

6.1   A/D-D/A Overview

The interface from the analog world of speech and the AMBE-2000™ is typically an A/D-D/A chip. Selection of the A/D-D/A chip should be made carefully, with a preference given to 16 bit linear devices. Additionally, consideration should be given for signal to noise ratios and filtering characteristics typically built into many such devices. Generally speaking, the flatter the frequency response over the voice spectrum (20-4000Hz) the better the overall system will sound.

The AMBE-2000™ Vocoder Chip operates with a speech data sample rate of 8kHz for both the A/D and D/A interfaces. This 8kHz data is input and output using a serial port on the AMBE-2000.
In order to simplify the process of configuring the interface to the A/D-D/A chip, a number of preset configurations can be chosen through the CODEC_SEL[1-0] pins shown in Table 6-A. These preset configurations control signal directions for the interface as well as the sequence of programming words for the programmable devices, specifically the AD7331AR.

6.2   Configuring the A/D-D/A Interface using CODEC_SEL[1-0]

In order to simplify the process of configuring the A/D-D/A interface certain preset configurations are available to the user. Selection of these preset modes is made through the 3 hardware pins CODEC_SEL[1-0]. In Table 6-A, the 2 digit binary value for CODEC_SEL[1-0] corresponds to the levels present on the hardware pins, with a 0 corresponding to GND, and a 1 corresponding to V_{CC}.

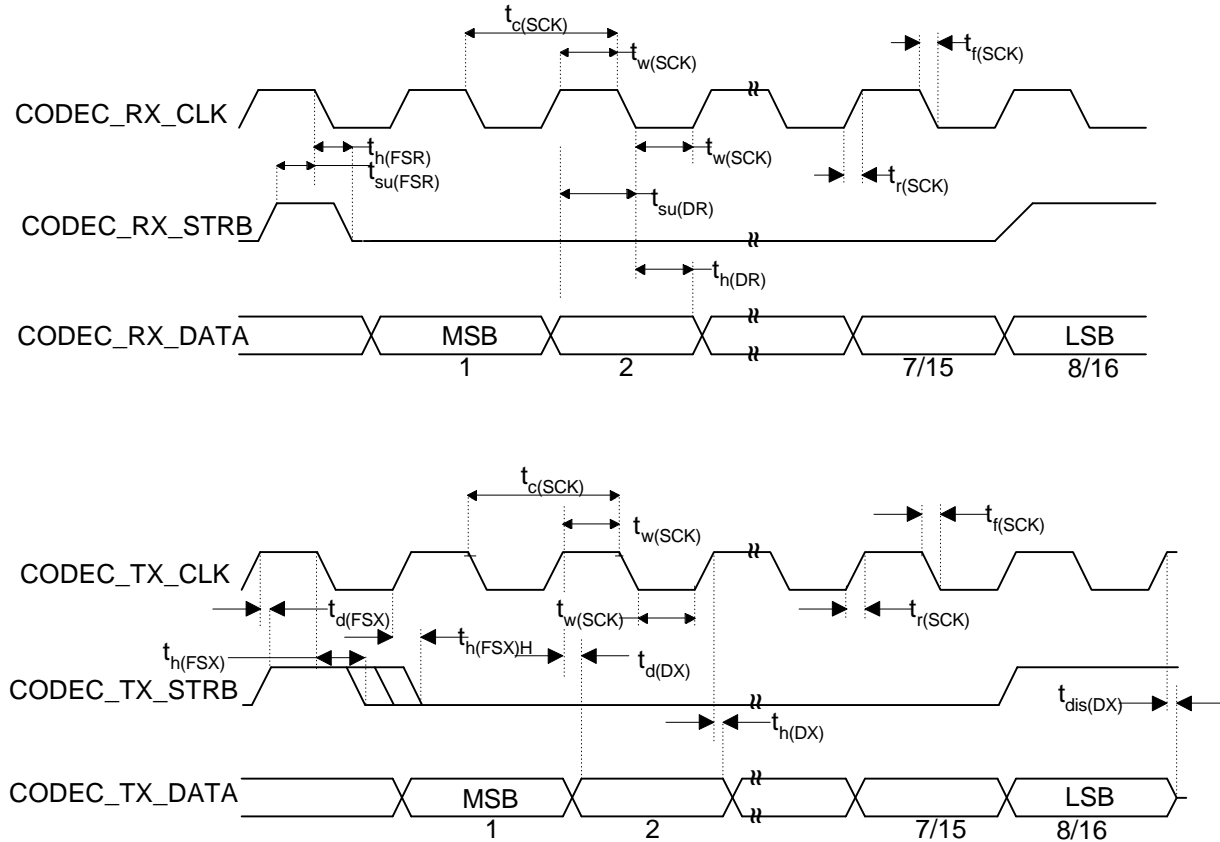**Table 6-A CODEC_SEL[1-0] : A/D-D/A Hardware Configuration Values**

A/D-D/A Type	CODEC_SEL[1-0] pins 85,84
Generic 16 bit Linear 8Khz	00b
Analog Devices 7331 32kHz	01b
Generic μlaw 8kHz	10b
Generic Alaw 8kHz	11b

6.3　Low Level A/D–D/A Timing

Low Level Timing for A/D-D/A in Active and Passive Modes

Figure 6-A Low Level Timing for A/D-D/A in Active and Passive Modes

Table 6-B Switching Characteristics Over Recommended Operating Conditions for Serial Port Receive

Reference	Parameter	3.3 Volts		Units
		Min	Max	
$t_{h(FSR)}$	Hold time, FSR after CLKR falling edge	6		ns
$t_{h(DR)}$	Hold time, DR after CLKR falling edge	6		ns

Table 6-E Switching Characteristics Over Recommended Operating Conditions for Serial Port Receive

Reference	Parameter	3.3 Volts		Units
		Min	Max	
$t_{d(SCK)}$	Delay time, DX valid after CLKX rising		25	ns
$t_{c(SCK)}$	Cycle time, serial port clock	6H		ns
$t_{f(SCK)}$	Fall time, serial port clock		6	ns
$t_{r(SCK)}$	Rise time, serial port clock		6	ns
$t_{w(SCK)}$	Pulse duration, serial port clock low/ high	3H		ns
$t_{su(FSR)}$	Setup time, FSR before CLKR falling edge	6		ns
$t_{su(DR)}$	Setup time, DR before CLKR falling edge	6		ns
$t_{d(FSX)}$	Delay time, CLKX rising to FSX		15	ns
$t_{h(FSX)}$	Hold time, FSX after CLKX falling edge	6		ns
$t_{h(FSX)H}$	Hold time, FSX after CLKX rising edge		2H-5	ns
$t_{dis(DX)}$	Disable time, CLKX rising to DX		20	ns
$t_{h(DX)}$	Hold time, DX valid after CLKX rising edge	-5		ns

- Note: H = 7.629 ns; however, do not operate serial clocks any faster than 2.048 MHz. Thus $t_{c(SCK)}$ should be a minimum of 488.3 nsec.

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7 Special Functions

7.1 Hardware vs. Software Selection Note

Many of the functions of the AMBE-2000™ can be accessed through both a hardware and software interfaces to the device. The following hardware inputs, CHANN_SEL[1-0], RATE_SEL[4-0], CODEC_SEL[1-0], VAD_EN, ECHOCAN_EN, and SLEEP_EN, are only accessed for input during the first 200 microseconds after a hardware reset on RESETN. For predictable operation these signals *must* remain stable over this time period. After this initialization period the functions that these pins access can only be reconfigured through the Control Word interface described in section 5.3.4. Changes on these pins after the 200 microseconds initialization period after reset are ignored, unless another reset is performed.

7.2 Coding Rate Selection

The Voice coding rate as well as the FEC coding rate can be selected individually on the AMBE-2000™. These rates are selected by using a Control Word as described in section 5.3.4, or through hardware pins RATE_SEL[4-0] subject to the restrictions in section 7.1. The five input pins RATE_SEL[4-0] give 16 preconfigured voice/FEC rates. The voice and FEC rates are individually configurable in 50 bit per second intervals. If rates other than these are desired then the Control Word method of configuring the rates must be used.

Table 7-A Rate Selection Using Rate Info 0-4, compatible w/ AMBE-1000™

RATE_SEL4 Pin	RATE_SEL3 Pin	RATE_SEL2 Pin	RATE_SEL1 Pin	RATE_SEL0 Pin	Speech Rate (bps)	FEC Rate (bps)	Total Rate (bps)
0	0	0	0	0	2400	0	2400
0	0	1	0	1	2350	50	
0	0	0	0	1	3600	0	3600
0	1	0	1	1	3350	250	
0	1	1	1	1	4000	0	4000
0	1	1	1	0	3750	250	
0	0	0	1	1	4800	0	4800
0	0	1	1	1	4550	250	
0	0	0	1	0	3600	1200	
0	1	0	0	0	3100	1700	
0	1	0	1	0	4150	2250	6400
0	1	0	0	1	4400	2800	7200
0	1	1	0	0	7750	250	8000
0	1	1	0	1	4650	3350	
0	0	1	0	0	9600	0	9600
0	0	1	1	0	4850	4750	

Table 7-B Rate Selection Using Rate Info 0-4, AMBE-2000™ only

RATE_SEL4 Pin	RATE_SEL3 Pin	RATE_SEL2 Pin	RATE_SEL1 Pin	RATE_SEL0 Pin	Speech Rate (bps)	FEC Rate (bps)	Total Rate (bps)
1	1	1	1	1	2000	0	2000
1	0	0	0	0	3600	0	3600
1	0	0	0	1	4000	0	4000
1	0	1	1	0	2400	1600	
1	0	0	1	0	4800	0	4800
1	1	0	0	0	4000	800	
1	0	1	1	1	3600	1200	
1	1	0	0	1	2400	2400	
1	0	0	1	1	6400	0	6400
1	1	0	1	0	4000	2400	
1	1	0	1	1	4400	2800	7200
1	0	1	0	0	8000	0	8000
1	1	1	0	0	4000	4000	
1	0	1	0	1	9600	0	9600
1	1	1	0	1	3600	6000	
1	1	1	1	0	2400	7200	

7.3   Echo Cancellation

The AMBE-2000™ Vocoder Chip provides a 16 millisecond echo canceller that is suitable for canceling the local echo caused by a 2-to-4 wire hybrid and can achieve echo cancellation of approximately 30dB or more. Only the linear portion of the echo is cancelled, so circuits should be designed to minimize non-linearities.

The AMBE-2000™ Vocoder Chip echo canceller operates by sending a 240 millisecond audible training sequence to the A/D-D/A immediately following a reset. Best results will be achieved if the analog circuit causing any echo is stable at this time. If the analog circuit changes substantially following this training , the echo canceller must be re-initialized, by resetting the AMBE-2000™ Vocoder Chip for optimum performance.

Figure 7-A Typical Echo Path
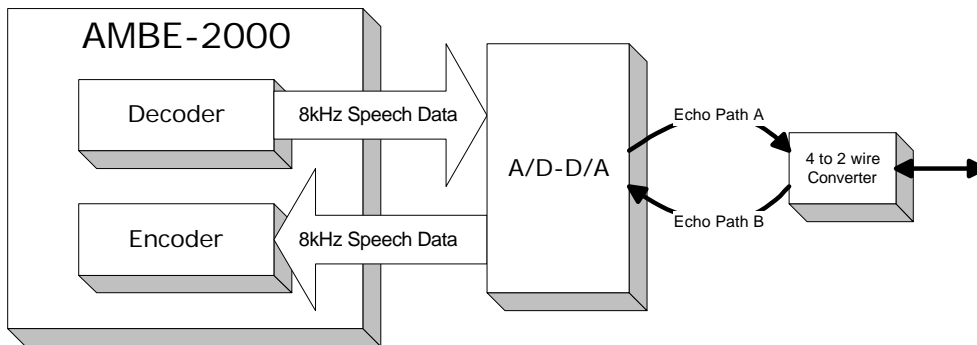
The Echo Return Loss (ERL) of the analog circuit must be 6dB or more (in diagram ERL = Echo Path A – Echo Path B) for proper echo canceller operation. Linear A/D-D/A chips will generally provide better echo cancellation performance than μlaw or Alaw chips due to lower quantization noise.

The echo canceller can be activated either through the hardware pin 78, ECHOCAN_EN, or through the Control Word interface described in section 5.2.9. See section 7.1 for important note about the ECHOCAN_EN pin.

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7.4 Voice Activation Detection (VAD), Comfort Noise Insertion (CNI)

The Voice Activation Detection (VAD) algorithm along with the Comfort Noise Insertion (CNI) feature of the AMBE-2000™ chip performs useful functions in systems trying to convert periods of silence, that exist in normal conversation, to savings in system bandwidth or power.

With the VAD functions enabled, periods of silence will be denoted by the encoder in two ways. First, the encoder will output a silence frame (in-band). This silence frame contains information regarding the level of background noise which allows the corresponding decoder to synthesize a "Comfort Noise" signal at the other end. The comfort noise is intended to give the listener the feeling that the call is still connected, as opposed to producing absolute silence which can give the impression that the call has been "dropped". Second, the Encoder Silence Detected flag is set in Control Word 1 of the Formatted Output format described in section 5.3.3.

VAD can be enabled in one of two ways. A high signal on the hardware pin VAD_EN (pin 86), subject to the restrictions of section 7.1, enables VAD. The Control Frame described in section 5.2.9 describes how to enable/disable the VAD algorithm once the AMBE-2000™ has begun operating.

If the VAD features are being used to reduce transmit power during times of conversational silence, DVSI recommends that a silence frame be transmitted at the start of the period and approximately each 500-1000 milliseconds thereafter. This is to ensure that the parameters regarding the levels of background noise are transmitted to the decoder for the smoothest audible transitions between synthesized speech and synthesized silence.

There is a *silence threshold* value is –25 dB in the VAD algorithm.

The synthesis of a Comfort Noise frame by the decoder is *not* dependant on VAD being enabled. The decoder will produce a comfort noise frame if it receives an in-band silence frame (produced only by an encoder with VAD enabled).

7.5 Dual Tone Multiple Frequency, Detection and Generation

The AMBE-2000™ Vocoder Chip is capable of detecting, transmitting, and synthesizing DTMF tones. DTMF features are always enabled. Detection of a DTMF tone by the encoder sets the DTMF Digit Detect in the DTMF Control Format found in section 5.4.6. Which DTMF tone is detected along with amplitude information is placed in the bits 0-7 described in section 5.3.6. Additionally, the encoder passes the DTMF data in-band (within the regular voice data bits) so that normal DTMF tones pass seamlessly from the encoder to the decoder for synthesis.

The decoder synthesizes a DTMF tone in response to reception of an in-band DTMF tone frame.

7.6 Normal Power and Power Saving Modes

Power savings can be achieved during times of longer inactivity of the AMBE-2000™ chip by placing it into one of three available Low Power Modes. The chip can be placed into low-power and stand-by modes via hardware or software Control Words. In low power modes the A/D-D/A port will be disabled, concurrently halting any processing of voice frames in either direction. Depending on the low power state selected, either a Wake Up Control Word or a hardware reset on RESETN is necessary to return the AMBE-2000™ to normal operation.

7.6.1 Standard Sleep Mode

The standard sleep is the only low power mode that can be entered into either through hardware or software. The AMBE-2000™ Chip can be placed into Standard Sleep mode either by setting SLEEP_EN (pin 83) high, subject to the restrictions of section 7.1, or through software by using Control Word 2 with bit 3 set to 1 as described in section 5.2.9.

SLEEP_EN should be tied high if you plan to configure the A/D-D/A chip from Standard Sleep mode upon power-up or reset.

When using software SLEEP_EN with A-law or μ-law codecs, it is important to note that if packets are sent to the decoder while it is in sleep mode, noise will be heard at the output. It is recommended that no packets be sent to the decoder until it is commanded to wake up.

7.6.2   Power Down

Power Down provides the lowest power usage of the sleep modes, the only drawback to this mode is the necessity of a hardware reset on RESETN (pin 69) to resume normal operation.

Table 7-C Summary of Power Saving Modes

Sleep Mode	Enter State via	Return to Normal Operation via	Wake Up Time	Power Consumption 3V	
				Crystal	CMOS TTL
Normal Operation	N/A	N/A	N/A	Approx. 65mW	
Standard Sleep	SLEEP_EN pin at reset OR Control Word	Control Word	N/A	24 mW	36 mW
Power Down	Control Word	RESETN	200 μ secs.	**0.11 mW**	**0.11 mW**

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7.7   Slip Enable

In any real time communication system, clock skew issues must be anticipated to keep the flow of data smooth from one end of the system to the other. The SLIP_EN (pin 82) signal allows the encoder of the AMBE-2000™ to react to small slips in the encoder channel signals. When the AMBE-2000™ is in active (parallel or serial) mode, the chip produces the signals internally for the transfer of data. Because the transmission channel will then likely be driven by this timing, the necessity of controlling slip becomes a moot point.

Any time the AMBE-2000™ encoder channel is in one of the passive modes and the channel timing is asynchronous to the A/D-D/A clock (very rarely are these two interfaces coupled) then the SLIP_EN pin should be set active high.

The AMBE-2000™ Vocoder chip processes speech in voice frames that are approximately 20 ms in duration. When configured appropriately the chip provides a slip control feature that automatically adjusts the frame size to either 160 or 161 speech samples per frame. This slip control feature allows the vocoder chip to compensate for drift between the frame and sample rate clocks on the order of approximately 0.6% (6,000 ppm.) The vocoder chip also accepts Slip Control Packets that extend the range of allowable frame sizes to be between 159 and 161 samples per voice frame. When properly used these Slip Control Packets provide the designer with additional flexibility in dealing with clock drift.

There are three recommended methods for using slip control on the AMBE-2000™ Vocoder Chips which are described below. The system designer should select the method that best meets the needs of their system configuration. Also included is some background information on the operation of the AMBE-2000 in passive mode

In order to help understand Slip control feature here is a brief description on reading encoder packets from the AMBE-2000 in passive formatted mode.

When transmitting a packet, the AMBE-2000 writes a Header = 0x13ec followed by 23 words of data, followed by 0xfffe into the transmit (i.e. output) buffer. The terminating word 0xfffe is written into the transmit buffer by the AMBE-2000 at the end of each encoder packet. Normally in passive mode this terminating word is in the transmit buffer at the beginning of each transmission cycle (from the previous frame) and so it is the first word output whenever a packet is transmitted. If the encoder packet is ready, then the second output word will be the packet Header=0x13ec followed by 23 words of data. However if the packet is not ready then the AMBE-2000 will continue to output the terminating word (0xfffe) until the packet is ready and placed in the transmit buffer. At this point the full 24 word packet beginning with the Header will be output on subsequent transmissions. This process continues for each packet transmission which occurs nominally every 20 ms, provided that each 24 word packet (Header + 23 data words) is read in full. If the full 24 words of the packet are not read from the AMBE-2000, then the chip's transmit buffer will contain some words left over from a previous packet. Generally these words would be 0x0000 for lower data rates which don't use the last words of the packet. This case should be avoided:

It is recommended that in passive formatted mode the system always read packets by requesting words from the AMBE-2000 until a packet Header is received and then continuing to request 23 additional output words from the AMBE-2000 until a total of 24 words beginning with the Header word = 0x13ec are received. Any words output by the AMBE-2000 prior to the Header should be ignored by the system (except for monitoring as discussed in Method 3 below).

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Word #	Value	Description
0	0x13ec	Header
1	0x07xx	Slip Control Packet ID, xx=CWD1 (See table 5-B) Typical value xx=00
2	0x1355	Slip Control indicator for AMBE-2000
3	0x009f	Slip Control data
4	0x0000	Must be 0x0000
5	0x0000	Must be 0x0000
6	0x0000	Must be 0x0000
7	0x0000	Must be 0x0000
8	0x0000	Must be 0x0000
9	0x0000	Must be 0x0000
10	0x0000	Must be 0x0000
11	0x0000	Must be 0x0000
12-23		Channel Data for AMBE® voice decoder

Table 7D: Slip Control Packet

Method 1 - Internal Slip Control

This is the simplest method of slip control and is the default method provided slip control is enabled (SLIP_EN is high) on the chip. In this method the vocoder chip's internal slip adjustment of 160 to 161 sample per frame is used. In order to work properly, the system designer must set-up the sample rate and packet timing so that the following constraint is met:

160 < Average-Frame-Interval * (1 +/- Frame-Drift) * Sample-Rate * (1 +/- Sample-Drift) < 161

For example using an Average-Frame-Interval = 20 ms, and assuming 100 parts per million oscillator accuracy (i.e Frame-Drift = Sample-Drift = .0001), then the above constraint requires:

8002 Hz < Sample-Rate < 8048 Hz.

In practice Sample-Rate=8002 Hz would be preferred since it is closer to the nominal value of 8000 Hz.

In another example the Sample-Rate = 8000 Hz. as provided by a PCM source. Again assuming 100 parts per million oscillator accuracy for both clocks (i.e. Frame-Drift = Sample-Drift = .0001), then the above constraint requires:

 20.0041 ms. < Average-Frame–Interval < 20.1209 ms.

This can be achieved by slightly decreasing the channel bit rate or adding an extra bit into the channel bit stream every several voice frames.

Method 2 - Extended Slip Control with Periodic Slip Control Packets

The AMBE-2000 can provide extended slip compensation through the insertion of Slip Control Packets. One method of using this capability is for the system to periodically insert these Slip Control Packets into the data stream sent to the AMBE-2000. Note that for this method slip control must be enabled (SLIP_EN is high) on the chip. This approach gives the designer a way to accommodate clock drift while providing very flexible frame-interval and sample-rate timing. Furthermore minimal system overhead is required. In this method a Slip Control Packet is generated by the system by setting the Control Words as shown in Table 7D above, where the Channel Data is the compressed voice data being sent to the AMBE decoder. In the Periodic Slip Control method such a Slip Control Packet is input into the vocoder chip every N frames. The value of N must be selected by the design engineer to meet the following constraint:

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$$0.25 > (1/N) > \text{Average-Frame-Interval} * \text{Sample-Rate} * (\text{Frame-Drift} + \text{Sample-Drift})$$

For example with an Average-Frame-Interval = .02 (i.e. 20 ms) and a Sample-Rate = 8000, then with 100 parts per million oscillator accuracy (i.e. Frame-Drift = Sample-Drift = .0001), then above constraint equates to $4 < N < 31.25$, and N=30 would be a reasonable selection. In this case the system would input the specified Slip Control Packet into the AMBE-20X0 vocoder chip every 30'th frame enabling the vocoder chip to adjust for the actual clock drift.

RECOMMENDED
**Method 3 - Extended Slip Control with as needed Slip Control Packets**

The preferred method for using Slip Control Packets is to monitor the availability of data from the AMBE-2000 vocoder chip and to only input Slip Control Packets into the data stream sent to the AMBE-2000 as needed. Note that for this method slip control must be enabled (SLIP_EN is high) on the chip. This method provides compensation for the widest range of clock drift (1.2% or 12000 ppm), with the greatest flexibility in frame-interval and sample-rate timing. In this method the same Slip Control Packets shown in Table 7D are inserted into the data stream going to the AMBE-2000. However, unlike in the Periodic method, the packets are not input at regular intervals but are instead only input to the chip when needed. The recommended procedure for this Method of Slip Control is for the system to read a packet from the AMBE-2000 at regular fixed frame intervals where the fixed interval must be within the range [19.875 – 20.125] ms for a sample rate = 8000 Hz. The system application should check each word output by the AMBE-2000 and should continue requesting words from the chip until the packet Header followed by 23 data words are received. If the words received before the Header word consist of only a single termination word (0xfffe) then no further action is required. However if two or more termination words are received prior to the Header then the system should input a Slip Control Packet to the AMBE-2000 on the next available transmission into the chip (i.e. the next packet going into the AMBE-2000 decoder should be a Slip Control Packet). Once this Slip Control Packet is input into the AMBE-2000 it will respond within 1-2 frames by advancing the time when packets are ready for transmission by 125 microseconds.   Note that this procedure also may require a small amount of buffering in the system to account for the fact that the packet my not be ready for some small time (< 125 microseconds) after it is first requested.

8 Appendices

8.1   Example A/D-D/A Usage

The following examples of A/D-D/A chips have been included to show connections necessary for interfacing to a number of popular chips.

8.1.1   AD73311

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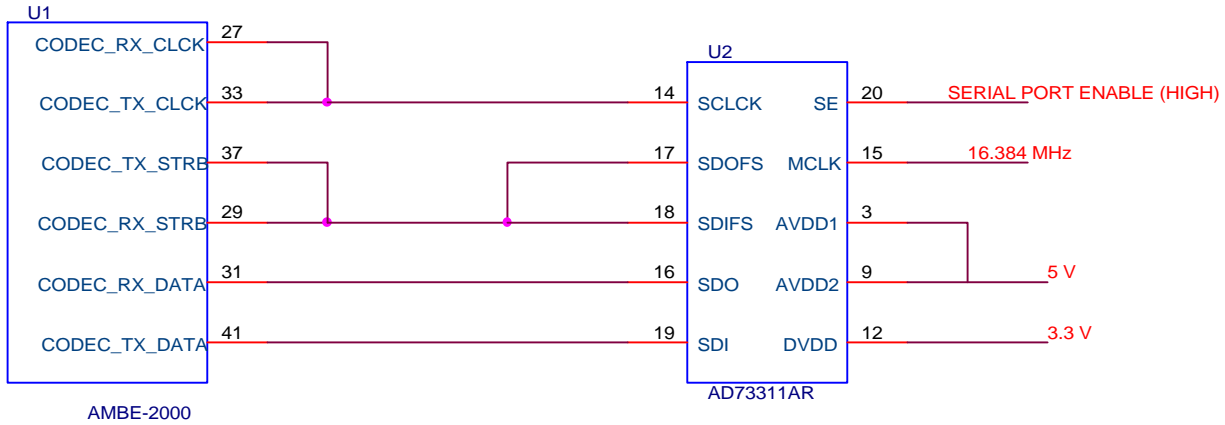
History of Revisions			
Revision Number	Date of Revision	Description	Pages
1.0	November 1999	Initial Version	
1.1	April 2000	Pin descriptions	11-12
		Serial Configuration Selection	19
		Channel Serial Interface Pin Descriptions	20
		Table 6-A CODEC_SEL[1-0]: A/D- D/A Hardware Configuration Values	35
1.2	May 2000	Deleted the word parallel	9
		Table 5-N Control Word Format: Removed VAD in bit 5. This is not used as an output.	32
1.3	August 2000	Added Application note for AD 73311AR	43
1.4	October 2000	Clarification on VAD	28
		Expanded Description of Control Word 1	29
		Table 5-M added DTMF Code 0xff	32
		Channel and Codec Timing Diagrams and Tables	21, 22, 36, 37
1.5	November 2000	Added Note describing H	22, 37
2.0	January 2001	Clarified/corrected the following pages.	10, 11, 13, 14, 16,17, 18, 20, 21, 22, 23, 27, 28, 29, 32, 34, 35, 36, 40

History of Revisions			
Revision Number	Date of Revision	Description	Pages
2.1	February 2001	Updated Timing Diagrams and Tables for Channel and Codec	21, 22, 36, 37
2.2	February 2001	Changed Pin Description CLK_I to X2/CLKIN and CLK_I2 to X1	11
		Updated timing diagram and table for X2/CLKIN and RESETN	13,14
		Replaced …set the DTMF Code to 0x00 to …set the DTMF Code to 0xff	27
3.0	August 2001	Company address updated	2
		Removed description of Decoder Output Volume Control	28
		Modified SLEEP_EN Section 7.6.1	40
		Modified description of EPR	11,12,21
		Added Detailed Explination of Slip Enable Control	43,44
		Added description of Decoder Silence Detect, Decoder Frame Repeat and Encoder DTMF Detect bits	30

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