Intel® Dialogic® DIALOG/4 Voice Board
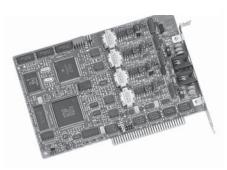
Retired product – This datasheet is for informational purposes only; the product detailed in this datasheet is no longer available. Consider migrating to the Intel Dialogic D/4PCIUF combined media board at http://www.intel.com/network/csp/products/8497web.htm.

The Intel® Dialogic® DIALOG/4 is a four-port voice board in a half-size ISA form factor that is ideal for the small/medium business (SMB) market segment. The DIALOG/4 has downloadable signal and call processing firmware that facilitates feature enhancement and helps developers build systems that scale from four to 64 ports.

Features and Benefits

Four independent voice processing ports in a single, half-size ISA slot supporting low- to medium-density voice systems

Downloadable signal and call processing firmware, Spring Ware, facilitates feature enhancement and provides field-proven performance based on over four million installed ports

C language application program interfaces (APIs) for MS-DOS*, Windows* 95, Windows NT*, OS/2*, UNIX*, and Linux* shorten the development cycle to get applications to market faster

Application generators available from third-party providers

Configure multiple DIALOG/4 boards in a single PC for easy and cost-effective system expansion, and to build scalable systems from four to 64 ports

Voice coding at dynamically selectable data rates, 24 Kb/s to 64 Kb/s, selectable on a channel-by-channel basis for optimal tradeoff in disk storage and voice quality
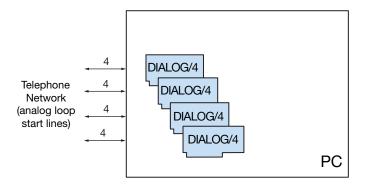
Enhanced telephone circuitry and automatic gain control (AGC) maintains recording quality over a wide dynamic range

Perfect digit DTMF (touchtone) provides reliable detection during voice playback - lets callers "type-ahead" through menus

Outbound call progress analyzes outgoing call status quickly and accurately

Supports PBXpert and PBXpert/32, complimentary utilities that simplify switch integration

Lifetime warranty

Configurations

The Intel® Dialogic® DIALOG/4 board shares a common hardware and firmware architecture with other Intel voice boards for maximum flexibility and scalability. You can easily add new features and/or expand the size of the system while protecting your original investment in hardware and application code. Applications can be ported to lower or higher line-density platforms with minimal modifications.
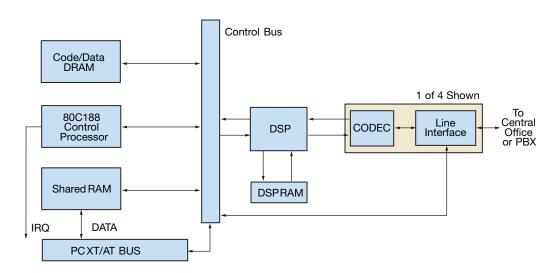
The DIALOG/4 board installs in IBM* PC XT*/AT* (ISA bus) and compatible computers (PC platforms based on the Intel386™, Intel486™, or Pentium® processors). The DIALOG/4 board provides everything required for building integrated voice solutions scalable from four to 64 ports.

Software Support

The Intel® Dialogic® DIALOG/4 board is supported by System Software and SDKs for MS-DOS*, Windows NT*, Windows* 95, and Linux*. These packages contain a set of tools for developing complex multichannel applications.

Applications

- Voice mail/voice messaging
- Interactive voice response
- Audiotex
- Inbound and outbound telemarketing
- Operator services
- Dictation
- Auto dialers
- Notification systems

Control Bus

Code/Data DRAM

80C188 Control Processor

Shared RAM

DSP

1 of 4 Shown

CODEC

Line Interface

To Central Office or PBX

DSP RAM

IRQ   DATA

PC XT/AT BUS

Functional Description

The Intel® Dialogic® DIALOG/4 board uses a unique dual-processor architecture that combines the signal processing capabilities of a digital signal processor (DSP) with the decision-making and data movement functionality of a general-purpose 80C188 control microprocessor. This dual-processor approach offloads many low-level decision-making tasks from the host computer, enabling development of more powerful applications.

This architecture
- handles real-time events
- manages data flow to the host PC for faster system response time
- reduces host PC processing demands
- processes DTMF and telephony signaling
- frees the DSP to perform signal processing on the incoming call

Each of four loop start telephone line interfaces on the DIALOG/4 board receives analog voice and telephony signaling information from the telephone network (see block diagram). Each line interface uses reliable, solid-state hook switches (no mechanical contacts) and FCC part 68 class B ring detection circuitry. This FCC-approved ring detector is less susceptible to spurious rings created by random voltage fluctuations on the network. Each interface incorporates circuitry that protects against high-voltage spikes and adverse network conditions, letting applications go off-hook any time during ring cadence without damaging the board.

Inbound telephony signaling (ring and loop current detection) are conditioned by the line interface and routed via a control bus to the control processor. The control processor responds to these signals, informs the application of

telephony signaling status, and instructs the line interface to transmit outbound signaling (on-hook/off-hook) to the telephone network.

The audio voice signal from the network is sent through a bandpass filter, conditioned by the line interface, and then applied to a COder/DECoder (CODEC) circuit. The CODEC filters, samples, and digitizes the inbound analog audio signal and passes the digitized signal to a Motorola* DSP.

Based on Spring Ware firmware loaded in DSP RAM, the DSP performs the following signal analysis and operations on this incoming data:
- automatic gain control (AGC) to compensate for variations in the level of the incoming audio signal
- applies an Adaptive Differential Pulse Code Modulation (ADPCM) or Pulse Code Modulation (PCM) algorithm to compress the digitized voice and save disk storage space
- detects the presence of tones — DTMF, MF, or an application-defined single- or dual-frequency tone
- silence detection to determine whether the line is quiet and the caller is not responding

For outbound data, the DSP performs the following operations:
- expands stored, compressed audio data for playback
- adjusts the volume and rate of speed of playback upon application or user request
- generates tones — DTMF, MF, or any application-defined general-purpose tone

The dual-processor combination also performs the following outbound dialing and call progress monitoring:

- transmits an off-hook signal to the telephone network
- dials out (makes an outbound call)
- monitors and reports results
  - line busy or congested
  - operator intercept
  - ring, no answer
  - or if the call is answered, whether answered by a person, an answering machine, a facsimile machine, or a modem

When recording speech, the DSP can use different digitizing rates from 24 Kb/s to 64 Kb/s as selected by the application for the best speech quality and most efficient storage. The digitizing rate can be selected on a channel-by-channel basis and can be changed each time a record or play function is initiated. Outbound processing is the reverse of inbound processing. The DSP processed speech is transmitted by the control microprocessor to the host PC for disk storage. When replaying a stored file, the microprocessor receives the voice information from the host PC and passes it to the DSP which converts the file into digitized voice. The DSP sends digitized voice to the CODEC to be converted into analog voice and then to the line interface for transmission to the telephone network.

The on-board microprocessor controls all operations of the DIALOG/4 board via a local bus and interprets and executes commands from the host PC. This microprocessor

- handles real-time events
- manages data flow to the host PC to provide faster system response time
- reduces PC host-processing demands
- processes DTMF and telephony signals before passing them to the application
- frees the DSP to perform signal processing

Communications between this microprocessor and the host PC is via the shared RAM that acts as an input/output buffer, increasing the efficiency of disk file transfers. This RAM interfaces to the host PC via the XT/AT bus. All operations are interrupt-driven to meet the demands of real-time systems. All DIALOG/4 boards installed in the PC share the same interrupt line. When the system is initialized, Spring Ware firmware to control all board operations is downloaded from the host PC to the on-board code/data RAM and DSP RAM. This downloadable firmware gives the board all of its intelligence and enables easy feature enhancement and upgrades.

Technical Specifications

	Number of ports	Four
	Max. boards/system	16
	Analog network interface	Onboard loop start interface circuits
	Microprocessor	80C188
	Digital signal processor	Motorola* DSP56001

Host Interface

	Bus compatibility	IBM PC XT/AT (ISA)
	Bus speed	4 MHz to 12 MHz, 70 nsec back-to-back bus cycle
	Shared memory	8 KB page, switch selectable on 8 KB boundaries
	Base addresses	D000h (default)
		A000h
		C000h
	Interrupt level	IRQ 2 to IRQ 7 jumper selectable
		One IRQ is shared by all DIALOG/4 boards

Telephone Interface†

	Trunk type	Loop start (or ground start for answer only)
	Impedance	600 Ohms nominal
	Ring detection	25 Vrms minimum
		15.3 Hz to 68 Hz, 150 Vrms maximum
	Loop current range	20 mA to 120 mA, DC (polarity insensitive)
	Receive signal/noise ratio	70 dB, referenced to –15 dBm
	Crosstalk coupling	–70 dB at 1 kHz channel to channel
	Frequency response	300 Hz to 3400 Hz ±3 dB (transmit and receive)
	Connector	Two RJ-11 type

Power Requirements

	+5 VDC	.75 A
	+12 VDC	40 mA
	–12 VDC	40 mA
	Operating temperature	0°C to +50°C
	Storage temperature	–20°C to +70°C
	Humidity	8% to 80% noncondensing
	Form factor	PC XT (ISA)
		7 in. long (17.5 cm)
		0.652 in. wide (1.63 cm)
		4.5 in. high (11.25 cm) (excluding edge connector)

Safety and EMI Certifications

	United States	FCC part 68 ID#: EBZUSA-65588-VM-E UL: E143032
	Canada	DOC: 885-4452A ULC: E143032
	Estimated MTBF	309,000 hours per Bellcore Method I
	Warranty	Intel® Telecom Products Warranty Information at http://www.intel.com/network/csp/products/3144web.htm

Spring Ware Technical Specifications

Audio Signal

Receive range	–50 dBm to –13 dBm (nominal), for average speech signals† configurable by parameter**
Automatic gain control	Application can enable/disable. Above –18 dBm results in full-scale recording, configurable by parameter.**
Silence detection	–38 dBm nominal, software adjustable**
Transmit level (weighted average)	–9 dBm nominal, configurable by parameter**
Transmit volume control	40 dB adjustment range, with application-definable increments and legal limit cap

Frequency Response

24 Kb/s	300 Hz to 2600 Hz ±3 dB
32 Kb/s	300 Hz to 3400 Hz ±3 dB
48 Kb/s	300 Hz to 2600 Hz ±3 dB
64 Kb/s	300 Hz to 3400 Hz ±3 dB

Audio Digitizing

24 Kb/s	ADPCM @ 6 kHz sampling
32 Kb/s	ADPCM @ 8 kHz sampling
48 Kb/s	µ-law PCM @ 6 kHz sampling
64 Kb/s	µ-law PCM @ 8 kHz sampling
Digitization selection	Selectable by application on function call-by-call basis
Playback speed control	Pitch controlled
	Available for 24 and 32 Kb/s data rates
	Adjustment range: ±50%
	Adjustable through application or programmable DTMF control

DTMF Tone Detection

DTMF digits	0 to 9, *, #, A, B, C, D per Bellcore* LSSGR Sec 6
Dynamic range	Default set to –36 dBm to –3 dBm per tone, configurable by parameter**
Minimum tone duration	40 ms, can be increased with software configuration
Interdigit timing	Detects like digits with a 40 ms interdigit delay. Detects different digits with a 0 ms interdigit delay.
Twist and frequency variation	Meets Bellcore LSSGR Sec 6 and EIA 464 requirements
Acceptable twist	10 dB
Signal/noise ratio	10 dB (referenced to lowest amplitude tone)
Noise tolerance	Meets Bellcore LSSGR Sec 6 and EIA 464 requirements for Gaussian, impulse, and power line noise tolerance
Cut-through	Detects down to –36 dBm per tone into 600 Ohm load impedance
Talk off	Detects less than 20 digits while monitoring Bellcore TR-TSY-000763 standard speech tapes. (LSSGR requirements specify detecting no more than 470 total digits.) Detects 0 digits while monitoring MITEL speech tape #CM 7291.

Global Tone Detection

Tone type	Programmable for single or dual
Max. number of tones	Application-dependent
Frequency range	Programmable within 300 Hz to 3500 Hz
Max. frequency deviation	Programmable in 5 Hz increments
Frequency resolution	Less than 5 Hz. NOTE: Certain limitations exist for dual tones closer than 125 Hz apart.
Timing	Programmable cadence qualifier, in 10 ms increments
Dynamic range	Programmable, default set at –36 dBm to +3 dBm per tone

Spring Ware Technical Specifications (cont.)

Global Tone Generation

	Tone type	Generate single or dual tones
	Frequency range	Programmable within 200 Hz to 4000 Hz
	Frequency resolution	1 Hz
	Duration	10 ms increments
	Amplitude	–43 dBm0 to –3 dBm0 per tone nominal, programmable

MF Signaling

	MF digits	0 to 9, KP, ST, ST1, ST2, ST3 per Bellcore LSSGR Sec 6, TR-NWT-000506 and CCITT Q.321
	Transmit level	Complies with Bellcore LSSGR Sec 6, TR-NWT-000506
	Signaling mechanism	Complies with Bellcore LSSGR Sec 6, TR-NWT-000506
	Dynamic range for detection	–25 dBm0 to –3 dBm0 per tone
	Acceptable twist	6 dB
	Acceptable freq. variation	Less than ±1 Hz

Call Progress Analysis

	Busy tone detection	Default setting designed to detect 74 out of 76 unique busy/congestion tones used in 97 countries as specified by CCITT Rec. E., Suppl. #2. Default uses both frequency and cadence detection. Application can select frequency only for faster detection in specific environments.
	Ring back detection	Default setting designed to detect 83 out of 87 unique ring back tones used in 96 countries as specified by CCITT Rec. E., Suppl. #2. Uses both frequency and cadence detection.
	Positive voice detection accuracy	>98% based on tests on a database of real world calls in North America. Performance in other markets may vary.
	Positive voice detection speed	Detects voice in as little as 1/10th of a second
	Positive answering machine detection accuracy	85% based on application and environment
	Fax/modem detection	Preprogrammed
	Intercept detection	Detects entire sequence of the North American tri-tone. Other SIT tones can be programmed.
	Dial tone detection before dialing	Application enable/disable
		Supports up to three different user-definable dial tones
		Programmable dial tone drop out debouncing

Tone Dialing

	DTMF digits	0 to 9, *, #, A, B, C, D per Bellcore LSSGR Sec 6, TR-NWT-000506
	MF digits	0 to 9, KP, ST, ST1, ST2, ST3
	Frequency variation	Less than ±1 Hz
	Rate	10 digits/s, configurable by parameter**
	Level	–4.0 dBm per tone, nominal, configurable by parameter**

Pulse Dialing

	10 digits	0 to 9
	Pulsing rate	10 pulses/s, nominal, configurable by parameter**
	Break ratio	60% nominal, configurable by parameter**

Spring Ware Firmware Technical Specifications (cont.)

Analog Display Services Interface (ADSI)

FSK generation per Bellcore TR-NWT-000030

CAS tone generation and DTMF detection per Bellcore TR-NWT-001273

** All specifications are subject to change without notice.

Hardware System Requirements

■ Intel386™, Intel486™, or Pentium® processor IBM PC AT (ISA) bus or compatible computer

■ Operating system hardware requirements vary according to the number of channels being used

Recommended Cables

The DIALOG/4 does not come with the Y adapter (RJ-14 to RJ-11) and is sold separately. The item market name for this cable is CBLRJ14TORJ11YA.

To learn more, visit our site on the World Wide Web at **http://www.intel.com**.

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Positive Answering Machine Detection/Positive Voice Detection
These performance results were measured using specific computer systems and/or components within specific lab environments and under specific system configurations. Any difference in system hardware, software design, or configuration may affect actual performance. The results are furnished for informational use only and should not be construed as a commitment by Intel. Intel assumes no responsibility or liability for any errors or inaccuracies.

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* Other names and brands may be claimed as the property of others.

† Average speech mandates +16 dB peaks above average and preserves −13 dB valleys below average.
‡ Analog levels: 0 dBm0 corresponds to a level of +3 dBm at tip-ring analog point. Values vary depending on country requirements; contact your Intel Sales Engineer.