Datasheet

Intel

Intel® Dialogic® ProLine/2V

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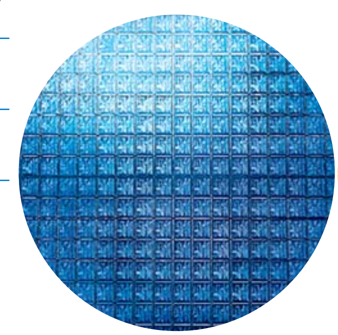
- *Intel® Dialogic® DIALOG/4 (**http://www.intel.com/network/csp/products/2419web.htm**)*
- *Intel® Dialogic® D/4PCI voice board (**http://www.intel.com/network/csp/products/5795web.htm**)*
- *Intel® Dialogic® D/41JCT-LS (**http://www.intel.com/network/csp/products/6925web.htm**)*
- *D/160SC-LS voice board (**http://www.intel.com/network/csp/products/1765web.htm**)*

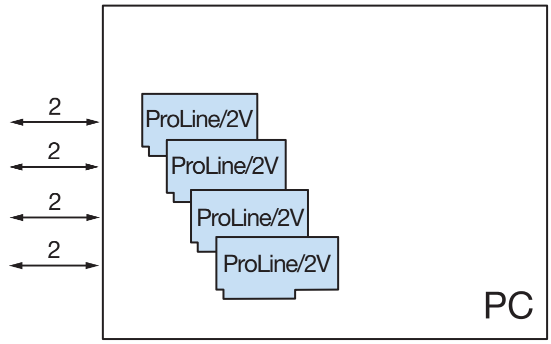
The Intel® Dialogic® ProLine/2V is a feature-rich, two-port voice board in a 2/3 length ISA form factor that is ideal for the small/medium business (SMB) market segment. The ProLine/2V has audio connectors and an electret microphone that allow for convenient online and offline recording and playback of system voice prompts.

Features and Benefits

Two independent voice processing ports in a single, 2/3-size ISA slot supporting low- to medium-density voice systems
Configure multiple ProLine/2V boards in a single PC for easy and cost-effective system expansion, and to build scalable systems from two to 32 ports
Audio connectors allow convenient offline recording and playback of system voice prompts
Electret microphone input jack allows convenient online recording of system voice prompts
Caller ID capability for “screen pop” applications (supports Bellcore* CLASS protocols)
Windows* 95 and Windows NT* Telephony API (TAPI*) support and .WAV audio capability
Optional Global DPD pulse-to-tone conversion software lets developers use the ProLine/2V in countries with limited touchtone telephone service
Voice coding at dynamically selectable data rates (24 Kb/s to 88 Kb/s, selectable on a channel-by-channel basis) provide optimal tradeoff between disk storage requirements and voice quality
Enhanced telephone circuitry and automatic gain control (AGC) maintains recording quality over a wide dynamic range
Downloadable Spring Ware signal and call processing firmware provides easy feature enhancement and field-proven performance based on over four million installed ports
Perfect digit DTMF (touchtone) provides reliable detection during voice playback — lets callers “type-ahead” through menus
Outbound call progress analysis analyzes outgoing call status quickly and accurately
C language application program interfaces (APIs) for MS-DOS*, Windows* 95, and Windows NT*
Third-party application generators available for rapid application development



Intel in Communications



Configurations

The Intel® Dialogic® ProLine/2V board shares a common hardware and firmware architecture with other Intel voice boards for maximum flexibility and scalability. Developers can easily add features and/or expand the size of the system while protecting their original investment in hardware and application code. Applications can be ported to lower or higher line-density platforms with minimal modifications.

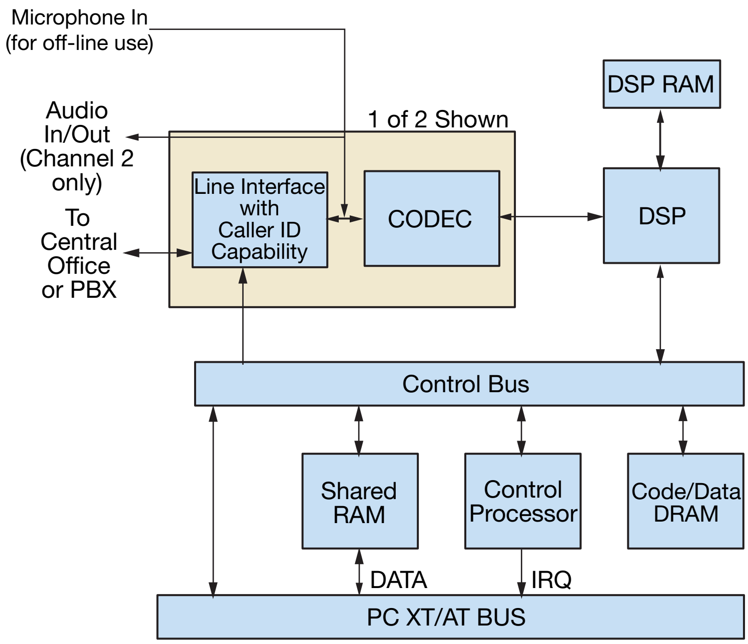
The ProLine/2V board installs in IBM* PC XT*/AT* (ISA bus) and compatible computers (PC platforms based on the Intel386™, Intel486™, or Pentium® processors). The ProLine/2V board provides everything developers need for building integrated voice solutions scalable from two to 64 ports.

Software Support

The Intel® Dialogic® ProLine/2V board is supported by System Software and Software Development Kits (SDKs) for MS-DOS*, Windows* 95, and Windows NT*. 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



Functional Description

The Intel® Dialogic® ProLine/2V 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 two loop start telephone line interfaces on the ProLine/2V board receive analog voice and telephony signaling information from the telephone network (see block diagram). Each telephone 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 also 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.

Part of the board's telephone interface includes an on-hook audio path that detects Caller ID information. Depending on the level of service offered by the local telephone provider, Caller ID can include the date, time, caller's telephone number, and (in some enhanced Caller ID environments) the name of the person calling. The on-hook audio path can also detect touchtones while the line is on-hook. This capability lets developers use the ProLine/2V board behind PBXs that require on-hook touchtone detection for their signaling.

The ProLine/2V also receives and transmits audio directly on one channel via line-level input and output jacks or directly into an electret microphone jack. This interface bypasses the telephony interface and lets you record prompts. Line-level input can be used to load prerecorded prompts or messages via line-level audio devices, such as a cassette tape recorder or compact disc player. The line-level output can also be used to monitor calls or play out files in a development environment.

Based on Spring Ware firmware loaded in DSP RAM, the DSP performs the following signal analysis and operations on the 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 88 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. The DSP processed speech is transmitted by the control microprocessor to the host PC for disk storage.

Outbound processing is the reverse of inbound processing. When replaying a stored file, the microprocessor receives the voice information from the host PC and passes it to the DSP which decodes the compressed file. 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 ProLine/2V 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 IBM* XT*/AT* bus.

All operations are interrupt-driven to meet the demands of real-time systems. ProLine/2V boards installed in the PC share the same interrupt line. When the system is initialized, Spring Ware firmware, which controls 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	Two
	Max. boards/system	16
	Analog network interface	Onboard loop start interface circuits
	Microprocessor	Intel® 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, 3, 4, 5, 7, 9, 10, 11, 12 jumper selectable
		One IRQ is shared by all ProLine/2V boards
Telephone Interface‡		
	Trunk type	Loop start
	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)
	Crosstalk coupling	–70 dB at 3 kHz channel to channel
	Frequency response	300 Hz to 3400 Hz ±3 dB (transmit and receive)
	Connector	Two RJ-11 type
Audio Interface		
	Line input impedance	10 KOhms
	Line input signal range	–32 dBv to –2 dBv, AC coupled mono or stereo
	Line input connector	0.137 in. (3.5 mm) stereo audio jack
	Line output impedance	600 Ohms
	Line output signal range	–32 dBv to –2 dBv, mono
	Line output connector	0.137 in. (3.5 mm) stereo audio jack
Microphone Interface		
	Microphone input impedance	10 KOhms
	Microphone input signal range	–55 dBv to –25 dBv, AC coupled mono or stereo,
		+5 Vdc phantom power for electret microphones only
	Microphone input connector	0.137 in. (3.5 mm) stereo audio jack
Power Requirements		
	+5 VDC	500 mA
	+12 VDC	35 mA
	–12 VDC	35 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)
		7.9 in. long (19.75 cm)
		0.75 in. wide (1.875 cm)
		3.85 in. high (9.625 cm) (excluding edge connector)

Technical Specifications** (cont.)

Safety and EMI Certifications

	United States	FCC part 68 ID#: EBZUSA-65588-VM-E REN: 1.0B UL: E143032
	Canada	IC CS-03, 885-4452A Load number: 5 ULC: E143032
	Estimated MTBF	345,000 hours per Bellcore Method I
	Warranty	Intel® Telecom Products Warranty Information at http://www.intel.com/network/csp/products/3144web.htm

Spring Ware Firmware Technical Specifications**

Audio Signal

	Receive range	–50 dBm to –3 dBm (nominal), for average speech signals† configurable by parameter‡
	Automatic gain control	Application can enable/disable. Above –30 dBm results in full-scale recording, configurable by parameter.‡
	Silence detection	–40 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

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.

Wave Audio

Supports 11 kHz linear PCM, 8-bit mono mode (available only when running Windows* 95 and Windows NT*)

DTMF Tone Detection

	DTMF digits	0 to 9, *, #, A, B, C, D per Bellcore* LSSGR Sec 6
	Dynamic range	Programmable, default set at –36 dBm to +0 dBm per tone
	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.

Spring Ware Firmware Technical Specifications** (cont.)

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 60 Hz apart.
	Timing	Programmable cadence qualifier, in 10 ms increments
	Dynamic range	Programmable, default set at –36 dBm to +0 dBm per tone
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 –1 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	80% to 90% 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	±0.5% of nominal frequency
	Rate	10 digits/s, configurable by parameter‡
	Level	–5 dBm0 per tone, nominal, configurable by parameter‡

Spring Ware Firmware Technical Specifications (cont.)

Pulse Dialing		
	10 digits	0 to 9
	Pulsing rate	10 pulses/s, nominal, configurable by parameter[‡]
	Break ratio	60% nominal, configurable by parameter[‡]
Analog Caller Identification		
	Applicable standards	Bellcore TR-TSY-000030
		Bellcore TR-TSY-000031
		TAS T5 PSTN1 ACLIP: 1994 (Singapore)
	Modem standard	Bell 202 or V.23, serial 1200 b/s (simplex FSK signaling)
	Receive sensitivity	–48 dBm (–50 dBv) to –1 dBm
	Noise tolerance	Minimum 18 dB SNR over 0 to –48 dBm dynamic range for error-free performance
	Data formats	Single Data Message (SDM) and Multiple Data Message (MDM) formats via API calls and commands
	Line impedance	AC coupled 600 Ohm (at 1.8 kHz) termination during Caller ID on-hook detection interval
	Message formats	ASCII or binary SDM, MDM message content
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.
† 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.

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

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