

ProLine/2V

Feature-Rich, Two-Port Voice Processing Board

The feature-rich ProLine/2V™ with its compact 2/3 length, XT height footprint, is an ideal solution for small computer telephony system development. It provides two telephone line interface circuits approved for direct connection to analog loop start lines. A unique dual-processor architecture comprising a Digital Signal Processor (DSP) and a general-purpose microprocessor handles all telephony signaling and performs DTMF (touchtone) and audio/voice signal processing tasks. You can install multiple ProLine/2V boards in a single PC chassis for system expansion up to 32 ports.

Windows® 95 and Windows® NT include TAPI/WAVE support which facilitates recording and playback of messages or system prompts via the ProLine/2V board's audio connectors and provides a base TAPI platform for Windows 95 and Windows NT application development. WAVE support increases your choices when recording and playing back audio files. You can record voice prompts directly through the ProLine/2V microphone input jack and play them back using the ProLine/2V board's WAVE capability. You can also convert audio from compact disc and CD-ROM sources (with the help of PC-based utilities) for use in your computer telephony applications.

Caller ID capability lets you create applications where the incoming caller's number can be used to search a database to create a "screen pop" of information about the caller. Additionally, you can use Caller ID to provide access to an enhanced level of services in a voice mail or IVR system.

The Global DPD™ Dial Pulse Detection algorithm from Dialogic is available for the ProLine/2V and lets you use the product in countries that have limited touchtone telephone service. Offered as a ProLine/2V software option, Global DPD can also be optimized on a country-by-country basis to provide superior dial pulse detection wherever it is used.

The on-board DSP executes downloaded SpringWare firmware algorithms to provide variable voice coding at 24 and 32 Kb/s ADPCM, and 48 and 64 Kb/s μ -law PCM. Sampling rates and coding methods are selectable on a channel-by-channel basis. Applications may dynamically switch the sampling rate to optimize data storage or voice quality as the need arises. SpringWare also provides reliable DTMF detection, DTMF cut-through, and talk off/play off suppression over a wide variety of telephone line conditions. Enhanced tele-

FEATURES AND BENEFITS

- Two independent voice processing ports in a single, 2/3-size PC ISA slot support low- to medium-density voice systems
- Audio connectors allow convenient off-line recording and playback of system voice prompts
- Electret microphone input jack allows convenient on-line recording of system voice prompts
- Windows® 95 and Windows NT® Telephony API (TAPI) support and .WAV audio capability
- Caller ID capability for "screen pop" applications (supports Bellcore CLASS Protocols)
- Optional Global DPD™ pulse-to-tone conversion software lets you 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 maintains recording quality over a wide dynamic range

- Downloadable SpringWare™ signal and call processing firmware provides easy feature enhancement and field-proven performance based on over two million installed ports
- PerfectDigit™ DTMF (touchtone) provides reliable detection during voice playback — allows callers to “type-ahead” through menus
- Patented outbound call progress PerfectCall™ analyzes outgoing call status quickly and accurately
- Configure multiple boards in a single PC for easy and cost-effective system expansion. Build scalable systems from 2 to 32 ports.
- C language application program interfaces (APIs) for MS-DOS®, Windows 95 and Windows NT
- Third-party application generators available for rapid application development

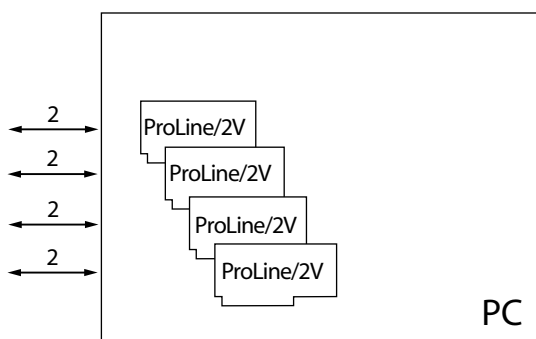
phone circuit design and automatic gain control maintain recorded voice quality even at extremely low signal levels.

Dialogic voice products offer a rich set of advanced features, including state-of-the-art DSP technology and signal processing algorithms, for building the core of any computer telephony system. With industry-standard ISA and PCI bus expansion boards and a variety of channel densities to choose from, you can integrate Dialogic voice products easily into exactly the type of system you require at a price and performance level unmatched in the computer telephony industry.

CONFIGURATIONS

The ProLine/2V board shares a common hardware and firmware architecture with other Dialogic voice boards for maximum flexibility and scalability. Add features or grow the system while protecting your investment in hardware and application code. With only minimum modifications, you can easily port applications to higher line density platforms.

The ProLine/2V board installs in IBM® PC XT®/AT® (ISA bus) and compatible computers (80386, 80486, or Pentium™-based PC platforms). The ProLine/2V board provides everything you need for building integrated voice solutions scalable from 2 ports to 32 ports.



SOFTWARE SUPPORT

The ProLine/2V is supported by Dialogic System Software and Software Development Kits 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
- Telecomputing servers
- Notification systems
- On-line data entry/query

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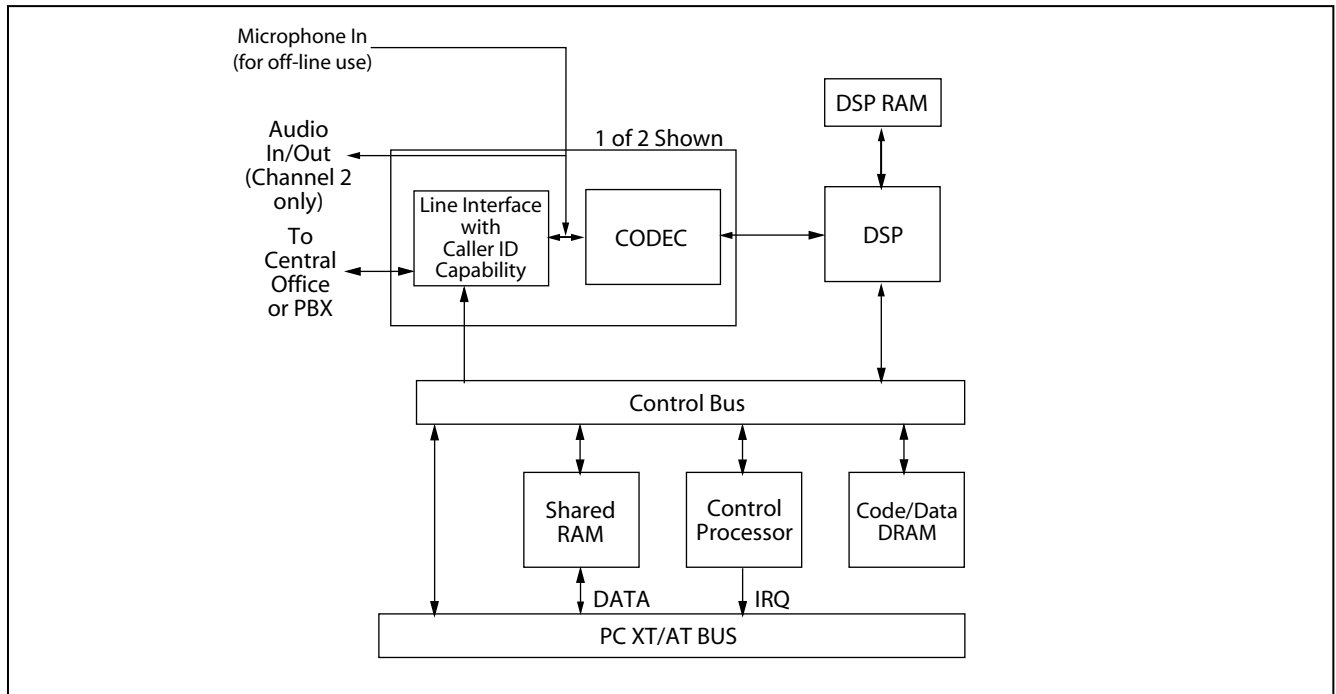
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Functional Description



The ProLine/2V board uses a unique dual-processor architecture that combines the signal-processing capabilities of a DSP with the decision-making and data movement functionality of a general purpose 80C188 control microprocessor. This dual-processor approach off-loads many low-level decision-making tasks from the host computer and makes it easier to develop 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, and frees the DSP to perform signal processing on the incoming call.

Each of two analog 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 Type B ring detection circuitry. This FCC-approved ring detec-

tor 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 and allows applications to go off-hook any time during ring cadence without damaging the board.

The line interface conditions the inbound telephony signaling (ring detection and loop current detection) and routes it 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 bandpass filtered and conditioned by the line interface and then applied to a CODEC (COder/DECoder) circuit. The CODEC filters, samples, and digitizes the inbound analog audio signal and passes this digitized audio 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 you 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 allows you to 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. You can use the line-level output to monitor calls or play out files in a develop-

ment environment.

The SpringWare firmware loaded into the DSP RAM provides the following signal analysis and operations on the incoming data:

- automatically controls the gain to compensate for variations in the level of the incoming audio signal
- applies an ADPCM (Adaptive Differential Pulse Code Modulation) or PCM (Pulse Code Modulation) 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 tone
- detects silence 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
 - call answered (differentiates whether answered by a person, answering machine, fax machine, or modem)

When recording speech, the DSP can use different digitizing rates from 24 to 88 Kb/s as selected by the application for the best speech quality and most efficient storage. The digitizing rate is 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 playing back 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 transmis-

sion 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 signaling before passing them to the application, and 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 and thus increases 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 ProLine/2V boards installed in the PC share the same interrupt line. When the system is initialized, SpringWare firmware, which controls all board operations, is downloaded from the host PC to the on-board code/data RAM and DSP RAM. SpringWare gives the board all of its intelligence and enables easy feature enhancement and upgrades. ■

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■ Technical Specifications*

	Number of ports	2
	Max. boards/system	16
	Analog network interface	On-board loop start interface circuits
	Microprocessor	Intel® 80C188
	Digital signal processor	Motorola DSP56002
HOST INTERFACE:		
	Bus compatibility	IBM PC XT/AT (ISA)
	ISA bus speed	4 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 or 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 min., 15.3 to 68 Hz, 150 Vrms max.
	Loop current range	20 to 120 mA, dc (polarity insensitive)
	Crosstalk coupling	-70 dB at 3 kHz channel to channel
	Frequency response	300 Hz to 3400 Hz \pm 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	3.5 mm stereo audio jack
	Line output impedance	600 ohms
	Line output signal range	-32 dBv to -2 dBv, mono
	Line output connector	3.5 mm stereo audio jack
MICROPHONE INTERFACE:		
	Mic input impedance	10 Kohms
	Mic input signal range	-55 dBv to -25 dBv, AC coupled mono or stereo, +5vdc phantom power for electret microphones only
	Mic input connector	3.5 mm microphone 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, 0.75 in. wide, 3.85 in. high (excluding edge connector)

■ Technical Specifications* (cont.)

REGULATORY CERTIFICATIONS:

United States	FCC part 68 ID#: EBZUSA-65588-VM-E REN: 1.0B UL: E143032
Canada	IC CS-03, 885 4452 A Load number: 5 ULC: E143032

WARRANTY:

Lifetime

■ SpringWare Technical Specifications*

AUDIO SIGNAL:

Receive range	-50 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.

GLOBAL TONE DETECTION™:

Tone type	Programmable for single or dual
Max. number of tones	Application dependent
Frequency range	Programmable within 300 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 to 4000 Hz
Frequency resolution	1 Hz
Duration	10 msec increments
Amplitude	-43 dBm to -3 dBm per tone, 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-506

■ SpringWare Technical Specifications* (cont.)

MF SIGNALING (cont.):

Signaling mechanism	Complies with Bellcore LSSGR Sec 6, TR-NWT-506
Dynamic range for detection	-25 dBm to -1 dBm 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
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 sequences 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; 16 digits per Bellcore LSSGR Sec 6, TR-NWT-506
MF digits	0 to 9, KP, ST, ST1, ST2, ST3
Frequency variation	$\pm 0.5\%$ of nominal frequency
Rate	10 digits/s max., configurable by parameter†
Level	-5 dBm per tone, nominal, configurable by parameter†

PULSE DIALING:

10 digits Pulsing rate	0 to 9 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 (@ 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 Dialogic Sales Engineer.

HARDWARE SYSTEM REQUIREMENTS

- 80386, 80486, or Pentium 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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