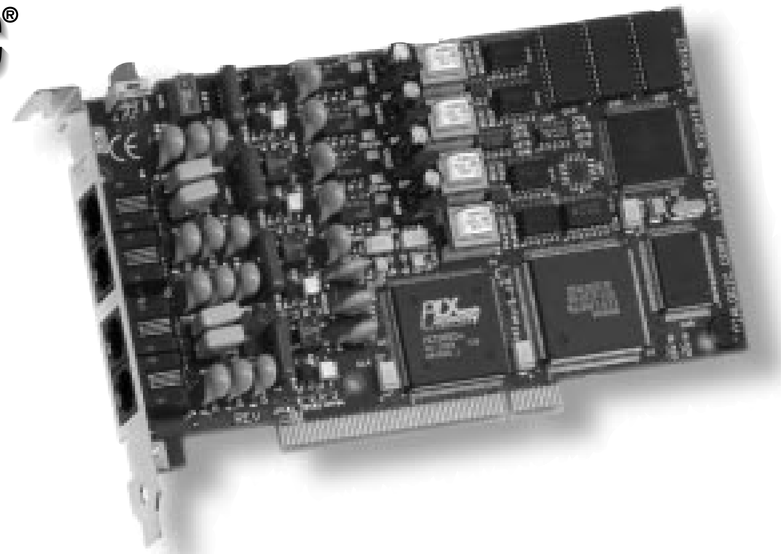
Dialogic®

an Intel company



D/4PCI

4-Port Voice Processing for Small and Medium Enterprise Applications

The four-line D/4PCI™ board is ideal for small- and medium-sized enterprise computer telephony (CT) applications that require high-performance, cost aggressive voice processing but don't need the large-scale system sophistication of SCbus™ or CT Bus™ based products. The D/4PCI board uses the same Dialogic application programming interface (API) as its predecessors, making it easy to scale existing applications to take advantage of its power and features. The D/4PCI board has improved voice quality and automatic gain control (AGC) so that even the weakest telephone signals can be recorded and replayed with complete clarity.

The D/4PCI board uses the latest digital signal processor (DSP) voice processing technology, making it ideal for server-based CT systems — particularly under the Windows® operating systems. Windows support includes TAPI and WAVE APIs which facilitate call control, recording, and playback of voice messages under the Microsoft Windows Open Services Architecture (WOSA) and lets developers quickly develop robust unified messaging applications. The D/4PCI voice processing board gives Windows NT® application developers a powerful platform for creating sophisticated interactive voice response (IVR) applications for the small- and medium-sized enterprise market. Caller ID support lets applications such as IVR receive calling party information via a telephone trunk line. Caller ID is supported for North America (CLASS protocol), the United Kingdom (CLI protocol), and in Japan (CLIP protocol).

The Dialogic Global Dial Pulse Detection™ (DPD) algorithm is available for the D/4PCI board, enabling applications to be deployed in countries with limited touchtone telephone service. Global DPD™ is optimized for a number of countries and provides superior dial-pulse detection.

Offered as additional software options, SpeechWorks-Host™ continuous speech recognition and TextTalk™ text-to-speech (TTS) software let you differentiate your offerings with state-of-the-art speech technologies for command and control of advanced IVR and unified messaging applications.

FEATURES AND BENEFITS

- Build flexible, cost-effective messaging and voice response platforms for small- and medium-sized enterprise applications
- Supports Windows NT® including TAPI/WAVE
- CTR-21 approvals mean expanded markets
- Caller ID lets applications perform intelligent call handling
- Delivers advanced call processing features and enables competitive differentiation by supporting software-based features such as
  - — Global Dial Pulse Detection™
  - — TextTalk™ text-to-speech
  - — SpeechWorks-Host™ continuous speech recognition
  - — PBXpert™ tone characterization utility
- Provides reliable DTMF detection during voice playback, letting callers "type-ahead" through voice menus for quicker completion of call transactions
- Ensures reliability via call progress analysis which monitors outgoing call status quickly and accurately
- Flexible voice coding at dynamically selectable data rates, 24 to 64 Kb/s, selectable on a channel-by-channel basis for optimal tradeoff in disk storage and voice quality
- Superior voice quality through enhanced telephone circuitry and automatic gain control

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SCSA

FEATURES AND BENEFITS

- Half-size PCI form factor enables developers to build cost-effective systems by using the most up-to-date industry-standard chassis. The ability to mix form factors offers a cost-effective transition to the PCI form factor.
- Compatible with legacy telephone switches in the United Kingdom and Northern Europe that use Earth Recall signaling

APPLICATIONS

- Networked voice messaging
- Automated attendant
- Interactive voice response
- Enhanced messaging

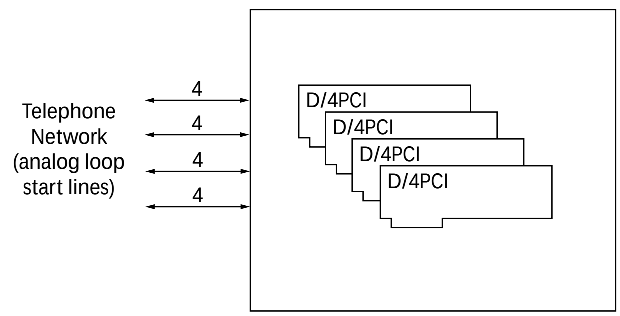
The D/4PCI board can also be used as a cost-effective platform for developing enhanced messaging applications.

With all of these advanced features in a half-size PCI board footprint, the D/4PCI board is perfect for client or small server system development. The board offers enhanced DSP power and memory capacity that provide a base level of performance for today's requirements as well as the "head room" for future application expansion via software-based technologies.

CONFIGURATIONS

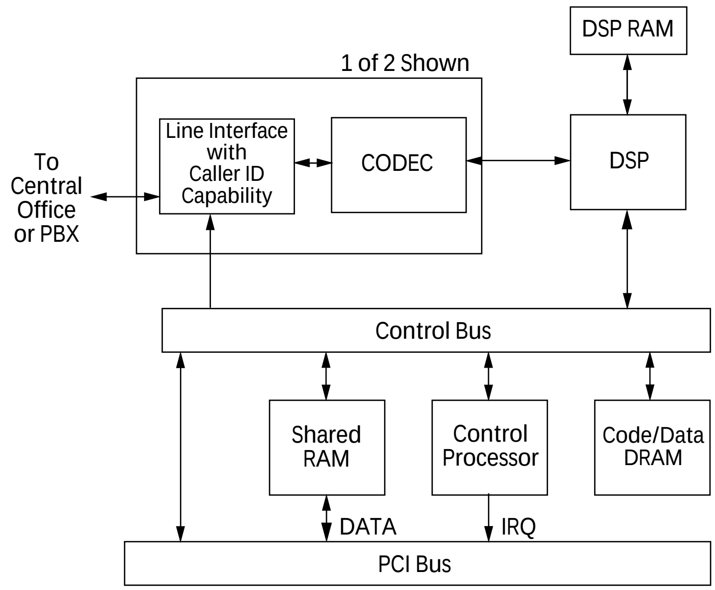
Use the D/4PCI board to build sophisticated messaging and IVR systems with optional technologies such as automatic speech recognition (ASR), TTS, Global DPD, and PBXpert™. The D/4PCI board shares a common hardware and firmware architecture with other Dialogic voice boards for maximum flexibility and scalability. More ports and new features can be added to a solution while protecting your original investment in hardware and application code. Applications can be ported to higher line density platforms with only minimum modifications.

The D/4PCI board installs in Intel® compatible computers (80486 or Pentium™ based PC platforms) and provides everything required for building integrated, non-CT Bus voice solutions, scalable from 4 to 64 ports.



SOFTWARE SUPPORT

The D/4PCI board is supported by the Dialogic System Software and SDK for Windows NT. The SDK contains all the documentation, demonstration code, and tools necessary for developing complex multichannel applications. ■



The D/4PCI voice processing board builds on the patented Dialogic dual-processor architecture that combines the signal processing capabilities of a DSP with the decision-making and data movement functionality of a general-purpose control microprocessor by using faster processors and considerably more memory. This dual-processor approach offloads many low-level decision-making tasks from the host computer, thus enabling easier 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, and frees the DSP to perform signal processing on the incoming call.

Each of the four loop start interfaces 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 and lets applications go off-hook any time during ring cadence without damaging the board.

Part of the telephone interface for the D/4PCI board 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 the board operate behind PBXs that require on-hook touchtone detection for their signaling.

Inbound telephony signaling (ring detection 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 bandpass filtered and 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 this digitized audio signal to a Motorola DSP.

Based on SpringWare™ firmware loaded in DSP RAM, the DSP performs the following signal analysis and operations on this incoming data:

- uses AGC to compensate for variations in the level of the incoming audio signal. The D/4PCI board also includes special circuitry to detect and amplify extremely weak line signals due to harsh telephone line conditions or back-to-back local loops often found in 800 (toll-free) service environments
- 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
- uses silence detection to determine when 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 (places 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 to 64 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 popular 11 kHz, 8-bit linear multimedia WAVE format is also supported on the D/4PCI voice board.

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 the 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 D/4PCI 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 PCI bus. All operations are interrupt-driven to meet the demands of real-time systems. All D/4PCI boards installed in the PC share the same interrupt line. When the system is initialized, SpringWare firmware is downloaded from the host PC to the on-board code/data RAM and DSP RAM to control all board operations. This downloadable firmware gives the board all of its intelligence and enables easy feature enhancement and upgrades. ■

Technical Specifications*

Number of ports	4
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	PCI (complies with PCISIG Bus Specification, Rev. 2.1)
PCI bus speed	33 MHz
Shared memory	8 KB page, PnP selectable on 16 KB boundaries
Base addresses	Selected by PCI BIOS
Interrupt level	One IRQ (IntA) shared by all boards

TELEPHONE INTERFACE:

Trunk type	Loop Start (or Ground Start for answer only)
Impedance	600 Ohm for D/4PCI. Matching complex impedance specified in CTR-21 for D/4PCI-Euro.
Ring detection	25 Vrms min., 15.3 Hz to 68 Hz, 150 Vrms max.
Loop current range	20 mA to 120 mA, DC (polarity insensitive), D/4PCI-Euro current limits at 60 mA per CTR-21 specifications
Crosstalk coupling	–80 dB at 3 kHz channel to channel
Frequency response	300 Hz to 3400 Hz ±3 dB (transmit and receive)
Connector	Four RJ-11

ENVIRONMENTAL REQUIREMENTS:

+5 VDC	650 mA
+12 VDC	55 mA
–12 VDC	53 mA
Operating temperature	0°C to +50°C
Storage temperature	–20°C to +70°C
Humidity	8% to 80% noncondensing
Form factor	PC AT (PCI); 6.9 in. long, 0.75 in. wide, 3.85 in. high (excluding edge connector)

REGULATORY CERTIFICATIONS:

United States	FCC part 68 ID#: EBZUSA-65588-VM-E REN: 1.0B UL: E143032
Canada	IC CS-03, CSA C22.2 No. 950 Load number: 5 ULC: E143032
Warranty	Lifetime

■ SpringWare 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 –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)

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

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 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-000506
Signaling mechanism	Complies with Bellcore LSSGR Sec 6, TR-NWT-000506
Dynamic range for detection	–25 dBm to –1 dBm per tone

■ SpringWare Specifications*

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.
Ringback detection	Default setting designed to detect 83 out of 87 unique ringback 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 tritone. Other SIT sequences can be programmed.
Dial tone detection before dialing	Application enable/disable. Supports up to three different user-definable dial tones. Programmable dialtone drop-out debouncing.

TONE DIALING:

DTMF digits	0 to 9, *, #, A, B, C, D; 16 digits 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 max., configurable by parameter†
Level	–5 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†

ANALOG CALLER IDENTIFICATION:

Applicable standards	Bellcore TR-TSY-000030 Bellcore TR-TSY-000031 TAS T5 PSTN1 ACLIP: 1994 (Singapore) British Telecom SIN 242 (Issue 01) British Telecom SIN 227 (Issue 01) Japan NTT CLIP
Modem standard	Bell 202 or V.23, serial 1200 b/s (simplex FSK signaling)
Receive sensitivity	–48 dBm to –1 dBm
Noise tolerance	Minimum 18 dB SNR over 0 dBm 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	600 Ohm for D/4PCI. Matching complex impedance specified in CTR-21 for D/4PCI-Euro.
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.

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* All specifications are subject to change without notice

† 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 Technical Sales Representative.

‡ Average speech mandates +16 dB peaks above average and preserves –13 dB valleys below average.

HARDWARE SYSTEM REQUIREMENTS

80486, Pentium or Intel compatible computer. Operating system hardware requirements vary according to the number of channels being used.

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