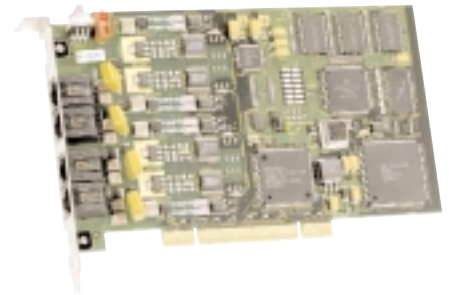




Intel® Dialogic® D/4PCIU Voice Board and Intel® Dialogic® D/4PCIUF Combined Media Board

Four-Port Voice and Fax Processing for Small and Medium Enterprise Applications

The four-line Intel® Dialogic® D/4PCIU and D/4PCIUF voice and fax processing boards are ideal for small- and medium-sized enterprise computer telephony (CT) applications that require high-performance, cost-aggressive voice and fax processing, but don't require the large-scale system sophistication of SCbus- or CT Bus-based products. Both boards use the same application



Features and Benefits

Build flexible, cost-effective messaging and voice response platforms for small- and medium-sized enterprise applications

Supports DSP-based onboard fax to reduce the number of boards per system (D/4PCIUF only)

Universal PCI edge-connector for compatibility with 3.3 V and 5.0 V bus signals enables deployment in a wide variety of PCI chassis from popular manufacturers

Supports Windows NT*, Windows* 2000, Windows XP*, and Linux* operating systems

Supports G.726 bit exact and GSM coders, letting developers implement unified messaging applications that meet VPIM standards

Conformity to the European R&TTE Directive and a variety of country-specific approvals mean expanded market segments

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 DPD
- PBXpert tone characterization utility

Provides reliable DTMF detection during voice playback, letting callers “type-ahead” through voice menus for quicker completion of call transactions

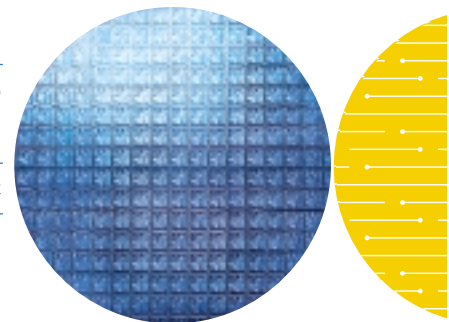
Ensures reliability via call progress analysis which monitors outgoing call status quickly and accurately

Flexible voice coding at dynamically selectable data rates, 24 Kb/s 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 AGC

Half-size PCI form factor lets developers 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.

Intel in
Communications



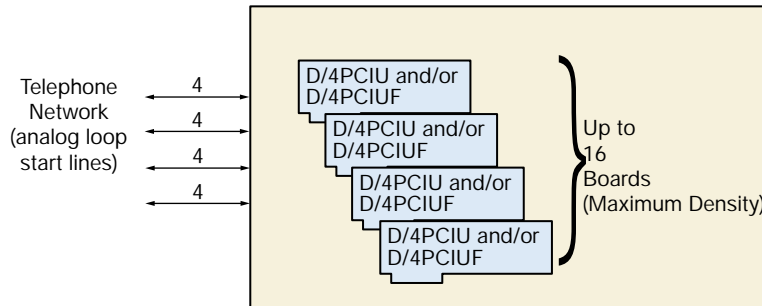


Figure 1. Configuration Example

programming interface (API) as their predecessor (D/4PCI), making it easy to scale existing applications to take advantage of their power and features. Both boards have 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/4PCIU and D/4PCIUF boards use the latest digital signal processor (DSP) voice processing technology, making it ideal for server-based CT systems — particularly under Windows* and Linux* operating systems. The D/4PCIU and D/4PCIUF voice and fax processing boards give Windows NT*, Windows XP*, and Windows 2000, as well as Linux application developers, a powerful platform for creating sophisticated interactive voice response (IVR) applications for the small- and medium-sized enterprise market segment. 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 Global Dial Pulse Detection (DPD) algorithm is available for the D/4PCIU and D/4PCIUF boards, enabling applications to be deployed in countries with limited touch-tone telephone service. Global DPD is optimized for a number of countries and provides superior dial-pulse detection.

With all of these advanced features in a half-size PCI board footprint, the D/4PCIU and D/4PCIUF boards are

perfect for client or small server system development. The boards offer enhanced DSP power and memory capacity that provides 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/4PCIU and D/4PCIUF boards to build sophisticated messaging and IVR systems with optional technologies such as Global DPD and PBXpert. Both boards share a common hardware and firmware architecture with other Intel Dialogic voice and fax 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/4PCIU and D/4PCIUF boards install in computers (PC platforms based on Pentium® processors) and provides everything required for building integrated, non-CT Bus voice solutions, scalable from four to 64 ports.

Software Support

The D/4PCIU and D/4PCIUF boards are supported by System Software and Software Development Kits (SDKs) for Windows NT, Windows XP, Windows 2000, and Linux operating systems. The SDKs contain all the documentation, demonstration code, and tools

Applications

- Networked voice messaging
- Automated attendant
- IVR
- Enhanced messaging

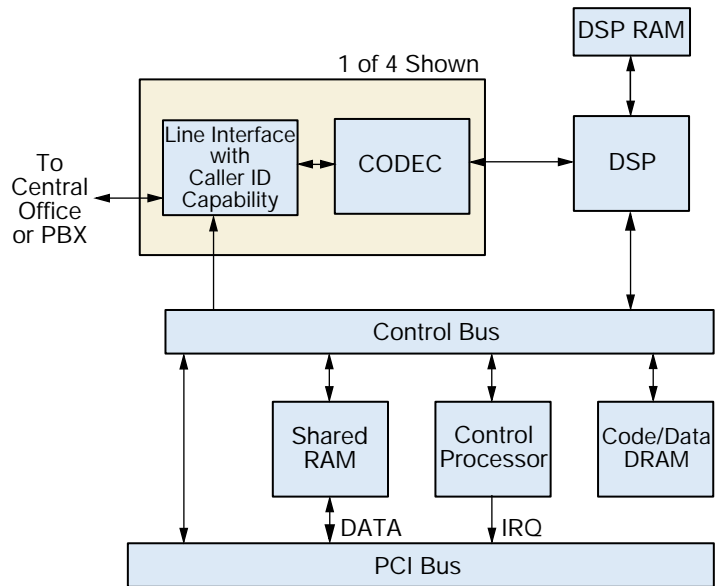


Figure 2. Functional Description

necessary for developing complex multichannel applications.

Functional Description

The D/4PCIU and D/4PCIUF voice and fax processing boards build on a 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 micro-processor 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 dual tone multifrequency (DTMF) and telephony signaling, and frees the DSP to perform signal processing on the incoming call.

Each of the four loop start interfaces receives analog voice and telephony signaling information from the telephone network (see Figure 2). Each telephone line interface uses reliable, solid-state hook switches (no mechanical contacts), and provides an on-hook path for ring detection. 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/4PCIU and D/4PCIUF boards 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 touch-tones while the line is on-hook. This capability lets the boards operate behind private branch exchanges (PBXs) that require on-hook touch-tone 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 Spring Ware 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/4PCIU and D/4PCIUF boards also include 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 Kb/s 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/4PCIU and D/4PCIUF voice and fax boards.

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/4PCIU and D/4PCIUF boards 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/4PCIU and D/4PCIUF boards installed in the PC share the same interrupt line. When the system is initialized, Spring Ware 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® 80C186 @ 34.8 MHz
Digital signal processor	Motorola* DSP56303 @ 100 MHz w/256 K × 24 private SRAM

Host Interface

Bus compatibility	PCI (complies with PCISIG Bus Specification, Rev. 2.1)
PCI bus speed	33 MHz max.
Shared memory	32 KB page
Base addresses	Selected by PCI BIOS
Interrupt level	One IRQ (IntA) shared by all boards

Telephone Interface

Trunk type	Loop start
Impedance	600 Ohm for D/4PCIU. Matching complex impedance specified in TBR-21 for D/4PCIU-EURO.
Ring detection	15 Vrms min., 15 Hz to 68 Hz
Loop current range	20 mA to 120 mA, DC (polarity insensitive)
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
Operating temperature	0°C to +50°C
Storage temperature	-20°C to +70°C
Humidity	8% to 80% noncondensing
Form factor	Universal slot (5 V or 3.3 V) PC AT (PCI) 6.9 in. (17.25 cm) long 0.75 in. (1.875 cm) wide 3.85 in. (9.625 cm) high (excluding edge connector)

Safety and EMI Certifications

United States	FCC part 68 ID#: EBZX07BD4PCIU REN: 0.7B UL: E96804
Canada	IC 885A-D4PCIU REN: 0.7B ULC: E96804
Estimated MTBF	434,000 hours per Telcordia Issue 1, Method 1, Case 1
Warranty	Intel® Telecom Products Warranty Information at http://www.intel.com/network/csp/products/3144web.htm

Spring Ware Technical Specifications**

Facsimile (available on D/4PCIUF only)

Fax compatibility	ITU-T G3 compliant (T.4, T.30), and ETSI NET/30 compliant
Data rate	14,400 b/s (v.17) send 9600 b/s receive
Variable speed selection	Automatic step-down to 12,000 b/s, 9600 b/s, 7200 b/s, 4800 b/s, and lower
Transmit data modes	MH (Modified Huffman) MR (Modified Read)
Receive data modes	MH, MR
File data formats	TIFF/F (Tagged Image File Format) for transmit/receive MH and MR
ASCII-to-fax conversion	Host-PC-based conversion Direct transmission of text files All Windows fonts supported Page headers generated automatically
Error correction	Detection, reporting, and correction of faulty scan lines
Image widths	8.5 in. (21.5 cm), 10 in. (25.4 cm), and 11.9 in. (30.2 cm)
Image scaling	Automatic horizontal and vertical scaling between page sizes
Polling modes	Normal and turnaround
Image resolution	Normal (203 pels/in. × 98 lines/in.; 203 pels/2.5 cm × 98 lines/2.5 cm) Fine (203 pels/in. × 196 lines/in.; 203 pels/2.5 cm × 196 lines/2.5 cm)
Fill minimization	Automatic fill bit insertion and stripping

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)

Spring Ware Technical Specifications (cont.)****DTMF Tone Detection**

DTMF digits	0 to 9, *, #, A, B, C, D per Bellcore LSSGR Sec 6
Dynamic range	Programmable, default set at -45 dBm to -3 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 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 -3 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 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 -3dBm per tone
Acceptable twist	6 dB
Acceptable freq. variation	Less than ± 1 Hz

Spring Ware Technical Specifications** (cont.)

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	Standard
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 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; 20 pulses/s for Japan 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/4PCIU and D/PCIUF. Matching complex impedance specified in TBR-21 for D/4PCIU-EURO and D/4PCIUF-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.

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

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**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.

