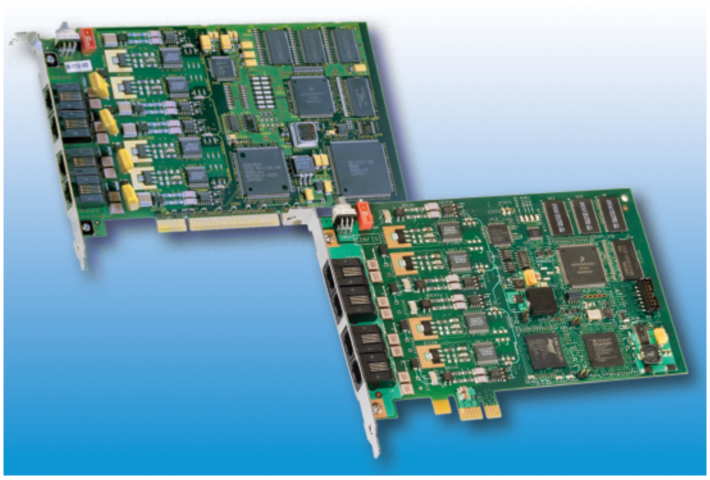
Dialogic® D/4PCIUF and D/4PCIU4S Media Boards by Sangoma



The Dialogic® D/4PCIUF (voice + fax) and D/4PCIU4S (voice + speech/ CSP) media boards are 4-port analog PCI or PCI Express half size boards that are part of the family of Dialogic® JCT Media Boards.

Dialogic® JCT Media Boards – including these models - can be used by developers to provide small- and medium-sized enterprise Computer Telephony (CT) applications that require high-performance voice and fax processing. Among the features and benefits of these boards, and other Dialogic® JCT Media Boards, are the following. They use Digital Signal Processor (DSP) voice processing technology, making them well-suited for server-based CT systems under Windows and Linux. They also provide a powerful platform for creating sophisticated Interactive Voice Response (IVR) applications for the small and medium-sized enterprise market segment. Their 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). Features such as fax and software-based speech recognition processing enable unified messaging applications. They also provide Automatic Gain Control (AGC), so even a weak telephone signal can be recorded and replayed with clarity.

Features	Benefits
Supports up to four channels of DSP-based on-board fax (D/4PCIUF models only)	Reduces the number of boards per system
Supports up to four channels of continuous speech processing (D/4PCIU4S models only)	Provides a flexible speech processing technology, which, when coupled with efficient drivers, off-loads critical real-time signal processing in speech-enabled applications to on-board DSPs. Reduces system latency, increases recognition accuracy, and improves overall system response time for speech solutions.
Separate models available with Universal PCI or PCI Express edge connector	Universal PCI form factor compatible with 3.3 V and 5.0 V bus signals; and PCI Express form factor compatible with x1 lane configuration or higher.
A variety of country-specific approvals	Expands an application's ability to serve several global market segments
Supports G.726 and GSM coders	Implements unified messaging applications that meet VPIM standards
Voice coding on a channel-by-channel basis	Allows for a beneficial tradeoff between disk storage and voice quality
Half-size PCI or PCI Express form factor	Cost-effective systems can be built using the up-to-date Commercial Off-The-Shelf (COTS) chassis

Technical Specifications

Number of ports	4
Maximum boards per system	16
Analog network interface	On-board loop start interface circuits
Control microprocessor	Intel 80C186 @ 34.8MHz
Digital signal processor	Freescale DSP56303 @ 100 MHz, with 128Kx24 private
Supported operating systems	Linux, Windows: Details at https://wiki.sangoma.com/display/DVC/Dialogic+Voice+Cards
CSP	Yes on D/4PCIU4S models only
FAX	Yes on D/4PCIUF models only
Signaling	Analog loop start

Host Interface — PCI and PCI Express

Bus compatibility	PCI: Complies with PCI-SIG Bus Specification, Rev. 2.2
	PCIe: Complies with PCI-SIG PCI Express Base Specification, Rev. 1.1; x1 or higher compatible
PCI bus speed	33 MHz maximum
Shared memory	32 KB page
Base addresses	Selected by PCI or PCI Express BIOS
Interrupt	PCI; 1 IRQ (INTA) shared by Dialogic® JCT Media Boards
	PCIe; Legacy INTA emulation shared by Dialogic® JCT PCIe Media Boards

Physical Dimensions

Standard-height, half-length form factor
6.88 in. (17.46 cm) long
0.75 in. (1.875 cm) wide
3.85 in. (9.625 cm) high (excluding edge connector)

Power Requirements — PCI

+5 VDC	650 mA

Power Requirements — PCI Express

+12 VDC	450 mA maximum

Environmental Requirements — PCI and PCI Express

Operating temperature	+32°F (0°C) to +122F°F (+50°C)
Storage temperature	–4°F (–20°C) to +158°F (+70°C)
Humidity	8% to 80% noncondensing0

Telephone Interface†

Trunk type	Loop start
	Ground start for inbound applications with AC ringing
Impedance	600 Ohm (nominal). Matching complex impedance specified in TBR-21 for D/4PCIU-EURO
Ring detection	15 Vrms minimum, 15 Hz to 68 Hz (each configurable by parameter*)
Loop current range	20 mA to 120 mA, DC (polarity insensitive)
Crosstalk coupling	–80 dB at 3 kHz channel-to-channel
Connector	4 RJ-11

Approvals, Compliance and Warranty

Country-specific safety and telecom approvals	https://portal.sangoma.com
Warranty information	https://www.sangoma.com/warranties

† 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 account manager*

Springware/JCT Technical Specifications

Facsimile (available on D/4PCIUF models only)

Fax compatibility	ITU-T G3 compliant (T.4, T.30) ETSI NET/30 compliant
Maximum Data rate	14.4 kbit/s (v.17) send 9.6 kbit/s (v.29) receive
Variable speed selection	Automatic step-down to 12,000 bit/s, 9600 bit/s, 7200 bit/s, 4800 bit/s, and lower
Transmit data modes	Modified Huffman (MH) Modified Read (MR)
Receive data modes	MH, MR
File data formats	Tagged Image File Format-Fax (TIFF-F) for transmit/receive MH and MR
ASCII-to-fax conversion	Host-PC-based conversion Direct transmission of text files Windows fonts supported Page headers generated automatically
Error correction	Detection, reporting, and correction of faulty scan lines
Image widths	1728 pixels 2048 pixels 2432 pixels
Image scaling	Automatic horizontal and vertical scaling between page sizes
Polling modes	Normal Turnaround
Image resolution	Normal (203 pels/in. x 98 lines/in.; 203 pels/2.54 cm × 98 lines/2.54 cm) Fine (203 pels/in. x 196 lines/in.; 203 pels/2.54 cm × 196 lines/2.54 cm)
Fill minimization	Automatic fill bit insertion and stripping

Audio Signal

Receive range	–50 dBm to –9 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, capped according to country-specific regulations

Frequency Response

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

Audio Digitizing

13 kbit/s	GSM 6.10 @ 8 kHz sampling
24 kbit/s	4-bit OKI ADPCM @ 6 kHz sampling
32 kbit/s	4-bit OKI ADPCM @ 8 kHz sampling
32 kbit/s	G.726 @ 8 kHz sampling
48 kbit/s	G.711 μ-law PCM @ 6 kHz sampling
64 kbit/s	G.711 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 kbit/s and 32 kbit/s data rates Adjustment range: ±50% Adjustable through application or programmable DTMF control

Wave Audio

Record/Play 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 Telcordia LSSGR Sec 6
Dynamic range	–38 dBm0 to –3 dBm0 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 Telcordia LSSGR Sec 6 and EIA 464 requirements
Acceptable twist	10 dB
Signal/noise ratio	10 dB (referenced to lowest amplitude tone)
Noise tolerance	Meets Telcordia 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 Telcordia TR-TSY-000763 standard speech tapes (LSSGR requirements specify detecting no more than 470 total digits) Detects zero (0) digits while monitoring MITEL speech tape #CM 7291

Global Tone Detection

Tone type	Generate single or dual tones
Maximum number of tones	Application dependent
Frequency range	Programmable within 300 Hz to 3500 Hz
Maximum 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 –6 dBm0 to –3 dBm0 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	Programmable within –43 dBm to –3 dBm per tone

MF Signaling

MF digits	0 to 9, KP, ST, ST1, ST2, ST3 per Telcordia LSSGR Sec 6, TR-NWT-000506 and ITU-T Q.321
Transmit level	Complies with Telcordia LSSGR Sec 6, TR-NWT-000506
Signaling mechanism	Complies with Telcordia 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
Ring back tone detection
Positive voice detection
Positive answering machine detection
Fax/modem detection
Intercept detection
Dial tone detection before dialing

Tone Dialing

DTMF digits	0 to 9, *, #, A, B, C, D; 16 digits per Telcordia 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	Telcordia TR-NWT-000030 Telcordia TR-NWT-000031 Telcordia TR-NWT-001188 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
Data formats	Single Data Message (SDM) and Multiple Data Message (MDM) formats via API calls and commands
Impedance	600 Ohm for D/PCIUF Matching complex impedance specified in TBR-21 for D/4PCIUF-EURO.
Message formats	ASCII or binary SDM, MDM message content

Analog Display Services Interface (ADSI)

FSK generation per Telcordia TR-NWT-000030

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

† *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 Representative.*

Ordering Information

Please see the Models tab for this product

ABOUT SANGOMA

Sangoma Technologies Corporation is a trusted leader in delivering globally scalable Voice-Over-IP telephony systems, both on-site and cloud-based. As the communication landscape evolves and businesses invest in new strategies to provide effective communications, Sangoma Technologies is your trusted partner; delivering Unified Communications solutions for SMBs, Enterprises, OEMs, Carriers, and service providers.

Founded in 1984, Sangoma Technologies Corporation is publicly traded on the TSX Venture Exchange (TSX VENTURE: STC).

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