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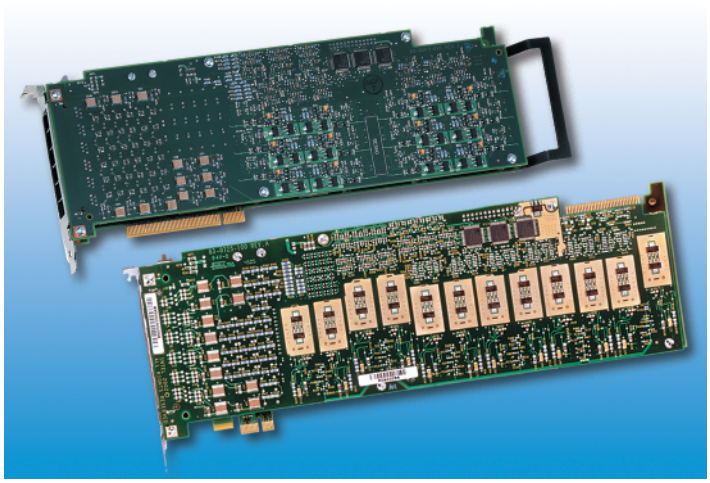
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Find the *Sangoma Technologies / Dialogic D/120JCT-LS* at our website: *Click HERE*

SANGOMA

Datasheet
JCT Media Boards

Dialogic® D/120JCT-LS Media Boards by Sangoma



The Dialogic® D/120JCT-LS Media Board is a 12-port analog PCI or PCI Express board well-suited for developing advanced communications applications that require multimedia resources. This high performance, scalable product supports voice, fax, and software-based speech recognition processing in a single PCI or PCI Express slot, providing 12 analog telephone interface circuits for direct connection to analog loop start lines.

Dialogic® JCT Media Boards – including this model - 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 this model, 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 segments. 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 G.726 bit exact and GSM coders	Enables implementation of unified messaging applications that meet VPIM standards
Supports Continuous Speech Processing (CSP)	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 high-density speech solutions.
A-law or μ-law voice coding at dynamically selectable data rates, 24 kbit/s to 64 kbit/s, selectable on a channel-by-channel basis	Allows for a beneficial tradeoff between disk storage and voice quality
Telcordia CLASS, UK CLI, Japanese Caller ID, and other international protocols	Supports an international Caller ID capability via on-hook audio path
A variety of country-specific approvals	Expands an application's ability to serve several global market segments at no extra cost
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.
Supports up to four (4) channels of DSP-based on-board fax	Reduces the number of boards per system

Technical Specifications

Number of ports	12
Maximum boards per system	8 (Linux and Windows). Number may be limited by application and system performance 1
CT Bus loads per board	20
Maximum CT Bus loads per system	On-board loop start interface (12)
Analog network interface	CT Bus
Resource sharing bus	Intel 80486 GXSF running at 32.768 MHz with 2 MB SDRAM
Control microprocessor	Freescale DSP56303 @ 100 MHz, with 128Kx24 private SRAM
Digital signal processor	Linux, Windows: Details at https://wiki.sangoma.com/display/DVC/Dialogic+Voice+Cards
Supported operating systems	Yes
CSP	Yes
FAX	Analog loop start
Signaling	

Host Interface — PCI

	Complies with PCI-SIG Bus Specification, Rev. 2.2; Universal slot (5 V or 3.3 V)
Bus compatibility	33 MHz maximum
Bus speed	
Bus mode	32-bit
Shared memory	32 KB to 64 KB page
Interrupt	1 IRQ (INTA) shared by Dialogic® JCT PCI Media Boards
I/O ports	None

Physical Dimensions — PCI

Standard-height, full length form factor
12.28 in. (31.2 cm) long
4.2 in. (10.67 cm) high

Power Requirements — PCI

+5 VDC	1.2 A typical; 1.4 A maximum
+12 VDC	235 mA typical; 285 mA maximum
–12 VDC	80 mA typical; 100 mA maximum

Host Interface — PCI Express

Bus compatibility	Complies with PCI-SIG PCI Express Base Specification, Rev. 1.1; x1 or higher compatible
Bus speed	2.5 GHz maximum per direction
Shared memory	32 KB to 64 KB page
Interrupt	Legacy INTA emulation shared by Dialogic® JCT PCIe Media Boards
I/O ports	None

Physical Dimensions — PCI Express

Standard-height, full length form factor
12.28 in. (31.2 cm) long
4.2 in. (10.67 cm) high

Power Requirements — PCI Express

+3.3 VDC	1.12 A typical, 1.4 A maximum
+12 VDC	800 mA typical, 900 mA maximum

Environmental Requirements — PCI and PCI Express

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

Telephone Interface†

Trunk type	Loop start Ground start for inbound applications with AC ringing
Impedance	600 Ohms nominal
Ring detection	40 Vrms to 130 Vrms, 15.3 Hz to 68.0 Hz (each configurable by parameter*)
Loop current range	20 mA to 60 mA, (Euro) 20 mA to 120 mA, polarity insensitive
Echo return loss	17 dB minimum (at country impedance)
Crosstalk coupling	>–75 dB
Speech digitization	64 kbit/s, µ-law PCM
Frequency response	300 Hz to 3400 Hz ±3 dB (transmit and receive)
Connector	RJ14; 6 jacks (each jack supports 2 channels)

Reliability

Estimated MTBF	Per Telcordia Method PCI: 154,000 hours PCI Express: 154,000 hours

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 Dialogic account manager*

Springware/JCT Technical Specifications

Facsimile

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	–40 dBm to –7 dBm nominal, configurable by parameter**
Automatic gain control	Application can enable/disable Above –22 dBm results in full-scale recording, configurable by parameter**
Silence detection	–40 dBm nominal, software adjustable**
Transmit level (weighted average)	–9.5 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

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
Noise tolerance	Meets Telcordia LSSGR Sec 6 and EIA 464 requirements for Gaussian, impulse, and power line noise tolerance
Cut-through	Local echo cancellation permits 100% detection with a >4.5 dB return loss line
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	Programmable for single or dual
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	± 5 Hz. Separation of dual-frequency tones is limited to 62.5 Hz at a signal-to-noise ratio of 20 dB
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 per Telcordia LSSGR Sec 6, TR-NWT-000506
Frequency variation	Less than ±1 Hz
Rate	10 digits/s maximum, configurable by parameter**
Level	–4.0 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)
Modem standard	Bell 202 or V.23, serial 1200 bits/sec (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
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 Telcordia TR-NWT-000030

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

*** 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*

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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SANGOMA TECHNOLOGIES

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