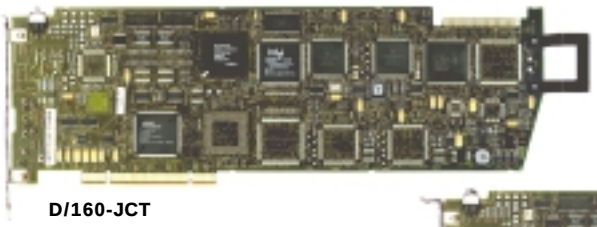
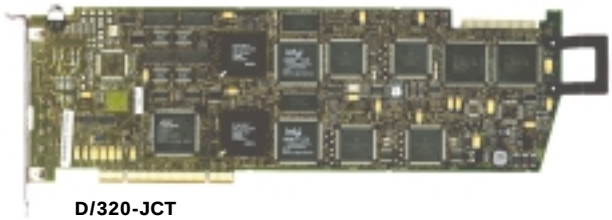
Datasheet

Intel® Dialogic® D/160-JCT and D/320-JCT 16- and 32-Port Voice Processing Shared Resource Boards



D/160-JCT



D/320-JCT

Intel® Dialogic® JCT Series resource boards provide an H.100-compliant universal PCI form factor with rich media features such as voice processing, fax, tone signaling, global tone detection and generation, and call progress analysis, making them ideal for service providers and large enterprises.

Intel in Communications

Features and Benefits

Larger systems, smaller cost: 16 or 32 independent voice channels in a single PC slot connect to digital network interfaces. Build larger analog or digital computer telephony (CT) systems while reducing overall system cost by using a PC chassis.

Onboard fax: Supports DSP-based onboard fax to maximize the number of boards in the system.

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

PCI form factor: Offered in industry-standard 32-bit PCI form factor with universal connector.

Eliminates silence: Silence-compressed recording eliminates silence and preserves hard disk space.

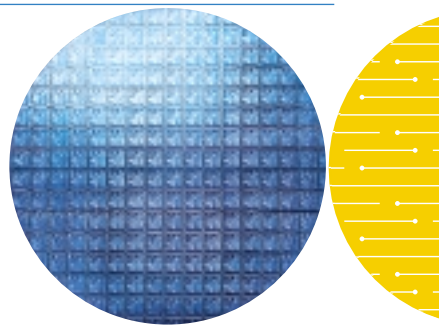
Standards-based H.100 connector: H.100 connector lets developers use the industry-standard CT Bus and increases the board's capacity to interoperate with other CT Bus-compatible boards.

Downloadable firmware: Downloadable signal and call processing firmware provides easy feature enhancement and field-proven performance based on more than four million installed ports.

Multi-application development: Support for Intel® NetMerge™ Converged Communications Server Software facilitates multi-application development.

Board Watch support: Supports Board Watch software, the SNMP-compatible software for remote CT board management.

Software development kits: Software development kits (SDKs) for Windows NT*, Windows* 2000, and Linux* operating systems yield faster time to market



The Intel® Dialogic® D/160-JCT and D/320-JCT boards are the next generation of Spring Ware-based 16- and 32-port voice resource boards in the PCI form factor. They are ideal for developers looking to provide cost-effective, highly scalable, high-density communications applications requiring multimedia resources like voice and fax in a single PC slot. Offering a rich set of advanced features, these boards support state-of-the-art digital signal processing (DSP) technology and industry-standard PCI bus and CT Bus technologies. Onboard DSP-based fax lets developers maximize the number of boards in the system for multimedia communications applications like Web-enabled call centers, unified messaging, or interactive voice response. The option to use new voice coders such as GSM and G.726 (the de facto standards when complying with Voice Profile for Internet Messaging [VPIM] standards) provides the capability to build unified messaging solutions while extending existing legacy messaging systems.

A unique multiprocessor architecture comprised of DSPs and general-purpose microprocessors performs all DTMF (touchtone) and audio/voice signal processing tasks. Onboard DSPs provide variable voice encoding at bit rates of 24 and 32 Kb/s for Adaptive Differential Pulse Code Modulation (ADPCM), 48 and 64 Kb/s µ-law or A-law for Pulse Code Modulation (PCM), 13 Kb/s for GSM, and 32 Kb/s for G.726. Sampling rates and coding methods are selectable on a channel-by-channel basis. Applications can dynamically switch sampling rate and coding method to optimize data storage or voice quality as the need arises. Spring Ware firmware also provides reliable DTMF detection, DTMF cut-through, and talk off/play off suppression over a wide variety of telephone line conditions.

Configurations

Use high-density digital voice resource boards to develop sophisticated, multimedia communications systems incorporating capabilities such as voice processing, fax, and text-to-speech (TTS). These boards share a common hardware and software architecture with other Intel® SCbus and CT Bus boards for maximum flexibility and scalability. You can add features and grow the system while protecting your investment in hardware and application code. Applications can be easily ported to lower- or higher-density platforms with only minimum modifications.

High-density digital voice resource boards install in any PCI-based PC or server (PCI bus or mixed PCI/ISA bus) and compatible computer. Each board occupies a single expansion slot; up to 6 boards can be configured in a system. The number of boards and channels supported varies depending on the application, the operating system, the amount of disk I/O required, the number of CT Bus loads per board, and the host computer's CPU(s) and power supply.

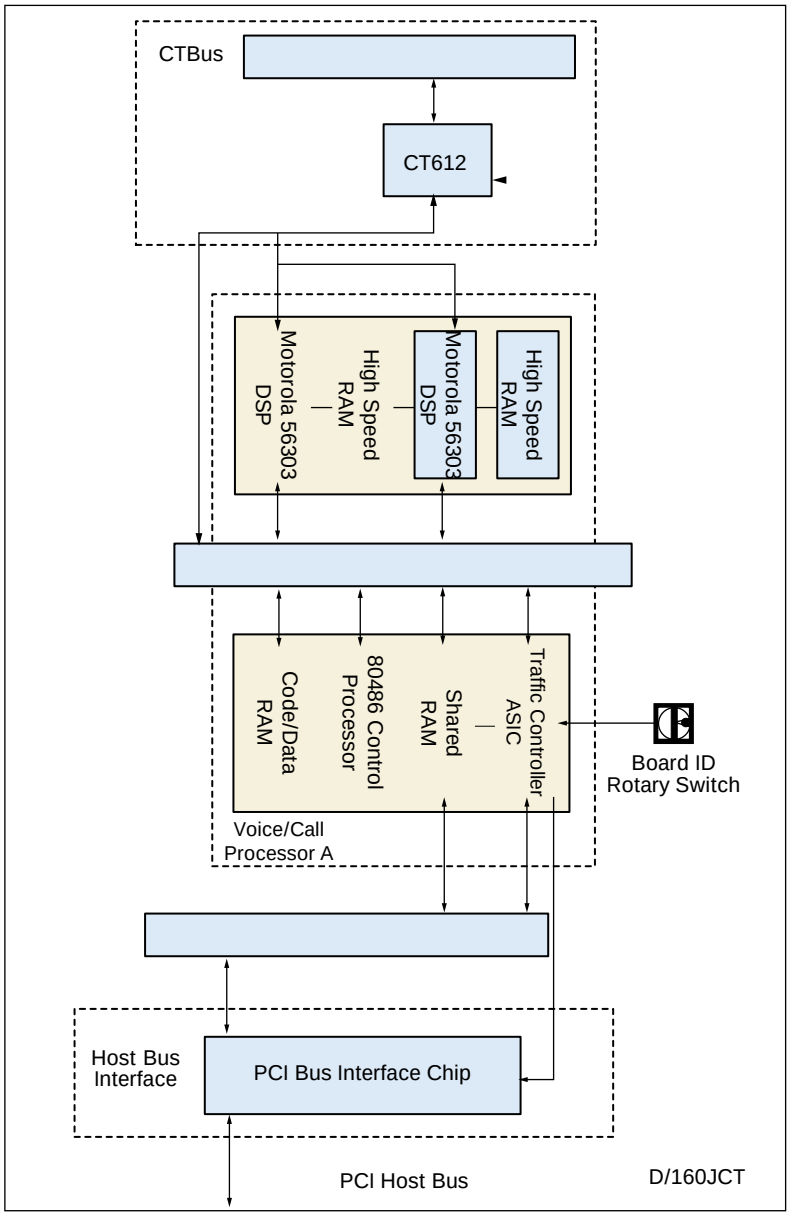
Software Support

Intel Dialogic D/160-JCT and D/320-JCT boards are supported by the Intel® Dialogic® System Software and software development kits (SDKs) for Windows NT*, Windows* 2000, and Linux* operating systems. These packages contain tools for developing sophisticated, multimedia communications applications.

These boards also support Board Watch software, the SNMP-compatible software for remote computer telephony (CT) board management. Board Watch software simplifies the management of CT devices and lowers the total cost of operation. Centralized management capabilities provide a single point of configuration and inventory for all network devices. Fault management for high-availability systems includes diagnostics, detection, and recovery capabilities.

Applications

- Messaging and enhanced services
- Contact center and e-Business
- PC-PBX



D/160-JCT Typical Configuration

Functional Description

D/160-JCT

The D/160-JCT board connects to network interface boards through the CT Bus interface for external network connections. A DSP resource receives digital voice data via a CT612 module, which acts as an interface to the CT Bus. The DSP processes the data based on Spring Ware firmware loaded in its high-speed RAM. Each DSP performs the following signal analysis and operations on this incoming data:

- Applies automatic gain control (AGC) to compensate for variations in the level of the incoming audio signal
- Applies an ADPCM, PCM, GSM, or G.726 algorithm to compress the digitized voice and save disk storage space
- Detects the presence of tones, whether DTMF, MF, or an application-defined, single- or dual-frequency 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)
- Monitors and reports call progress results (line busy or congested, operator intercept, ring, no answer, or if the call is answered, whether it is answered by a person or an answering machine, fax machine, or modem).

When recording speech, the DSP can use digitizing rates from 13 to 64 Kb/s as selected by the application for the best speech quality and most efficient storage. The digitizing rate, selected on a channel-by-channel basis, can be changed each time a record or play function is initiated. The DSP-processed speech is transmitted by the control processor to the host PC for disk storage. When replaying a stored file, the processor retrieves the voice information from the host PC and passes it to the DSP, which converts the file into digitized voice. For CT Bus/SCbus configurations, the internal local buses operate at 2.048 Mb/s. The onboard control processor(s) controls all operations of the board via local buses and interprets and executes commands from the host PC. These processors:

- Handle real-time events
- Manage data flow to the host PC to provide faster system response time
- Reduce PC host processing demands
- Process DTMF before passing them to the application
- Free the DSPs to perform signal processing

Communication between a processor and the host PC is via the shared RAM, which acts as an input/output buffer, increasing 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. When the system is initialized, Spring Ware firmware is downloaded from the host PC to the onboard code/data RAM and DSP RAM to control all board operations. This firmware gives the board all of its intelligence and enables easy feature enhancement and upgrades. The Traffic Controller ASIC is the Intel486™ processor interface that handles all peripheral devices (CT612 and DSPs) and host PC functions (programmable interrupts and shared RAM).

D/320-JCT

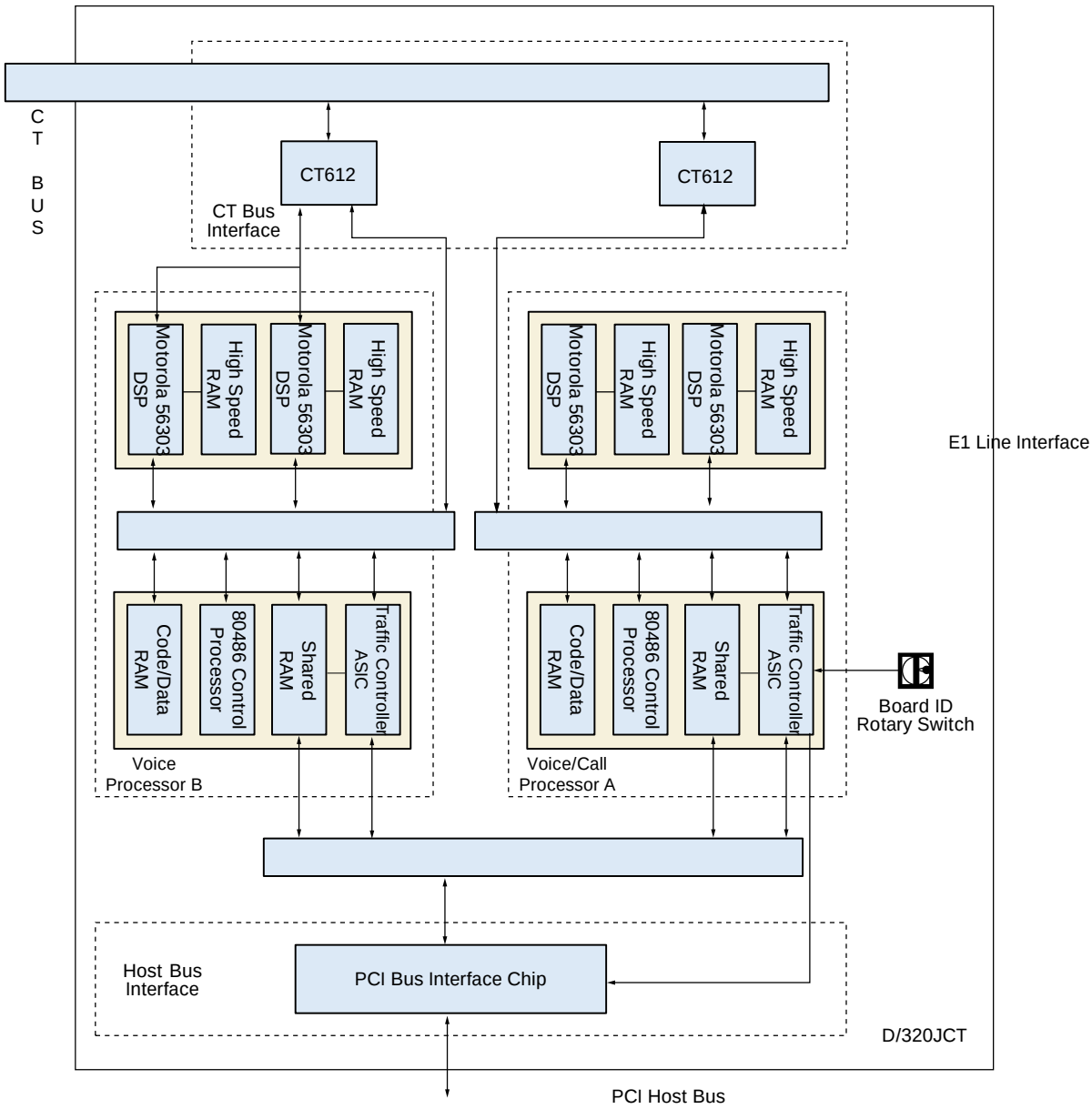
The D/320-JCT board connects to network interface boards through the CT Bus interface for external network connections. A DSP resource receives digital voice data via a CT612 module, which acts as an interface to the CT Bus. The DSP processes the data based on Spring Ware firmware loaded in its high-speed RAM. Each DSP performs the following signal analysis and operations on this incoming data:

- Applies AGC to compensate for variations in the level of the incoming audio signal
- Applies an ADPCM, PCM, GSM, or G.726 algorithm to compress the digitized voice and save disk storage space
- Detects the presence of tones, whether DTMF, R2MF, or an application-defined, single- or dual-frequency 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, R2MF, or any application-defined, general-purpose tone
- Monitors and reports call progress results (line busy or congested, operator intercept, ring, no answer, or if the call is answered, whether answered by a person or an answering machine, fax machine, or modem).

When recording speech, the DSP can use digitizing rates from 13 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 DSP-processed speech is transmitted by the control processor to the host PC for disk storage. When replaying a stored file, the



D/320-JCT Typical Configuration

processor retrieves the voice information from the host PC and passes it to the DSP, which converts the file into digitized voice. For CT Bus/SCbus configurations, the internal local buses operate at 2.048 Mb/s. The onboard control processor(s) controls all operations of the board via local buses and interprets and executes commands from the host PC. These processors:

- Handle real-time events
- Manage data flow to the host PC to provide faster system response time
- Reduce PC host processing demands
- Process DTMF before passing them to the application
- Free the DSPs to perform signal processing

Communications between a processor and the host PC is via the shared RAM, which acts as an input/ output buffer, increasing 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. When the system is i–nitialized, Spring Ware firmware is downloaded from the host PC to the onboard code/data RAM and DSP RAM to control all board operations. This firmware gives the board all of its intelligence and enables easy feature enhancement and upgrades. The Traffic Controller ASIC is the Intel486 processor interface that handles all peripheral devices (CT612 and DSPs) and host PC functions (programmable interrupts and shared RAM).

Technical Specifications**

D/160-JCT

No. ports	16
Maximum boards/system	Six. Number may be limited by application, system performance, and the number of CT Bus loads per board.
CT Bus loads per board	1.0
Maximum CT Bus loads/system	20 (See the CT Bus specification for details.)
Resource sharing bus	H.100, CT Bus
Control microprocessor	One Intel Intel486® GX processor at 32.7 MHz, 0 wait state
Digital signal processors	Two Motorola* DSP56303 processors at 100 MHz, each with 256 K word private, 2 wait state SRAM

Host Interface

Bus compatibility	PCI (Complies with PCISIG Bus Specification, Rev. 2.2.)
Bus speed	33 MHz maximum
Bus mode	32- to 16-bit conversion in target mode
Shared memory	2 x 64 KB page
I/O ports	None
Support	3.3 V or 5 V signaling environment (universal connectivity)

Power Requirements

+5 VDC	1.5 A maximum
+12 VDC	3.0 mA maximum
–12 VDC	Not required
Operating temperature	0 to +50°C
Storage temperature	-20 to +70°C
Humidity	8 to 80% noncondensing
Form factor	PCI (universal connector) 12.3 in. long (without edge retainer) or 13.3 in. long (with edge retainer) 0.79 in. wide (total envelope) 3.87 in. high (excluding edge connector)

Safety and EMI Certifications

United States	FCC part 68 ID#: EBZUSA-23891-CE-E UL: 1950 (E96804)
Estimated MTBF	367,000 hours per Bellcore Method
Warranty	Intel® Telecom Products Warranty Information at http://www.intel.com/network/csp/products/3144web.htm

Technical Specifications** (cont.)

D/320-JCT

No. ports	32
Maximum boards/system	Six. Number may be limited by application, system performance, and the number of CT Bus loads per board.
CT Bus loads per board	1.5
Maximum CT Bus loads/system	20 (See the CT Bus specification for details.)
Resource sharing bus	H.100, CT Bus
Control microprocessor	Two Intel Intel486® GX processor at 32.7 MHz, 0 wait state
Digital signal processors	Four Motorola* DSP56303 processors at 100 MHz, each with 256 K word private, 2 wait state SRAM

Host Interface

Bus compatibility	PCI (Complies with PCISIG Bus Specification, Rev. 2.2.)
Bus speed	33 MHz maximum
Bus mode	32- to 16-bit conversion in target mode
Shared memory	2 x 64 KB page
I/O ports	None
Support	3.3 V or 5 V signaling environment (universal connectivity)

Power Requirements

+5 VDC	2.0 A maximum
+12 VDC	7.0 mA maximum
–12 VDC	Not required
Operating temperature	0 to +50°C
Storage temperature	-20 to +70°C
Humidity	8 to 80% noncondensing
Form factor	PCI (universal connector) 12.3 in. long (without edge retainer) or 13.3 in. long (with edge retainer) 0.79 in. wide (total envelope) 3.87 in. high (excluding edge connector)

Safety and EMI Certifications

United States	FCC part 68 ID#: EBZUSA-23891-CE-E UL: 1950 (E96804)
Estimated MTBF	244,000 hours per Bellcore Method I
Warranty	Intel® Telecom Products Warranty Information at http://www.intel.com/network/csp/products/3144web.htm

Spring Ware Firmware Technical Specifications**

Facsimile

Fax compatibility	ITU-T G3 compliant (T.4, T.30)
	ETSI NET/30 compliant
Data rate	14,400 b/s (v.17) send
	9,600 b/s receive
Variable speed selection	Automatic step-down to 12,000 b/s, 9,600 b/s, 7,200 b/s, 4,800 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, M
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	215 mm (8.5 in.) 255 mm (10.0 in.) 303 mm (11.9 in.)
Image scaling	Automatic horizontal and vertical scaling between page sizes
Polling modes	Normal and turnaround
Image resolution	Normal (203 pels/in. x 98 lines/in.) Fine (203 pels/in. x 196 lines/in.)
Fill minimization	Automatic fill bit insertion and stripping

Audio Signal

Receive range	(T-1) –40 to +2.5 dBm0 nominal, configurable by parameter† (E-1) –43 to +2.5 dBm0 nominal, configurable by parameter†
Automatic gain control	Application can enable/disable. Above –18 dBm0 (T-1) or –21 dBm0 (E-1) results in full-scale recording, configurable by parameter.†
Silence detection	–38 dBm0 nominal, software adjustable†
Transmit level (weighted average)	(T-1) –9 dBm0 nominal, configurable by parameter† (E-1) –12.5 dBm0 nominal, configurable by parameter†
Transmit volume control	40 dB adjustment range, with application-definable increments and legal limit cap

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

13 Kb/s	GSM @ 8 kHz sampling
24 Kb/s	OKI ADPCM @ 6 kHz sampling
32 Kb/s	OKI ADPCM @ 8 kHz sampling
32 Kb/s	G.726 @ 8 kHz sampling
48 Kb/s	A-law PCM @ 6 kHz sampling
64 Kb/s	A-law PCM @ 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

Spring Ware Firmware Technical Specifications** (cont.)

DTMF Tone Detection

DTMF digits	0 to 9, *, #, A, B, C, D per CCITT Q.23
Dynamic range	–36 dBm0 to –3 dBm0 (T-1) or –39 dBm0 to 0 dBm0 (E-1) 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
Acceptable twist	(T-1) Meets Bellcore LSSGR Sec 6 and EIA 464 requirements
	(E-1) Meets appropriate CCITT specifications†
Noise tolerance	Meets Bellcore LSSGR Sec 6 and EIA 464 requirements for Gaussian, impulse, and power line noise tolerance
Cut-through	(T-1) Local echo cancellation permits 100% detection with a >4.5 dB return loss line
	(E-1) Digital trunks use separate transmit and receive paths to network. Performance dependent on far-end handset's match to local analog loop
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
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	(T-1) Programmable, default set at –36 dBm0 to –0 dBm0 (single tone), –3 dBm0 (dual tone) (E-1) Programmable, default set at –39 dBm0 to +0 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
	(T-1) –43 dBm0 to –3 dBm0 per tone nominal, programmable
	(E-1) –40 dBm0 to +0 dBm0 per tone nominal, programmable

Spring Ware Firmware 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.
	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	>99% based on tests on a database of real world calls in North America. Performance in other markets may vary.
	Positive voice detection speed	Detects voice in as little as 1/10th of a second
	Positive answering machine detection accuracy	85% based on tests on a database of real-world calls in North America. Performance in other markets may vary.
	Fax/modem detection	Preprogrammed
	Intercept detection	Detects entire sequence of the North American tri-tone. Other intercept tone 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 per Bellcore LSSGR Sec 6, TR-NWT-000506
	Frequency variation	Less than ±1 Hz
	Rate	10 digits/s, configurable by parameter†
	Level	–7.5 dBm0 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 Display Services Interface (ADSI)		
	FSK generation per Bellcore TR-NWT-000030.	
	Programmable dial tone drop out debouncing	

Hardware System Requirements

D/160-JCT and D/320-JCT

- Intel386™, Intel486™, or Intel® Pentium® processor PCI bus or mixed PCI/ISA bus computer
- Operating system hardware requirements vary according to the number of channels being used
- System must comply with PCISIG Bus Specification Rev. 2.1 or later

Additional Components

D/160-JCT and D/320-JCT

- Multidrop CT Bus cables(CBLCTB68C3DROP, CBLCTB68C4DROP, CBLCTB68C8DROP, CBLCTB68C12DROP, CBLCTB68C16DROP)
- CT Bus/SCbus Adapter (CTBUSTOSCBUSADP)
- SC bus terminator kits (1SCBUS1TERMKIT, 2SCBUS1TERMKIT, 3SCBUS1TERMKIT)

†Configurable to meet country-specific PTT requirements. Actual specification may vary from country to country for approved products

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1515 Route Ten
Parsippany, NJ 07054
Phone: 1-973-993-3000
Fax: 1-973-993-3093

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