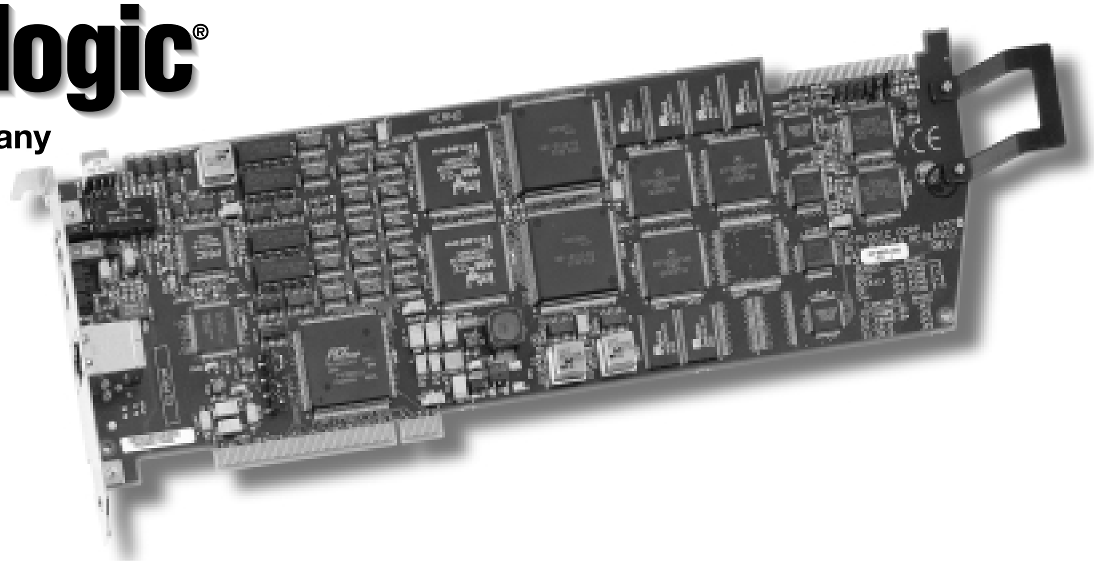
Dialogic®
an Intel company



SCSA Hardware Model | PEB Hardware Model | CT Bus/SCbus

SingleSpan™-JCT Series

24- and 30-Port Voice Processing with Digital Network Interface

The D/240JCT-T1™ and D/300JCT-E1™ boards are the next generation of SpringWare™ based SingleSpan™ products. They are ideal for developers seeking to provide cost-effective, highly scalable, high-density communications applications requiring multimedia resources such as voice, software-based speech recognition, fax, and digital network interface in a single personal computer (PC) slot. These boards offer a rich set of advanced features and support state-of-the-art digital signal processing (DSP) technology and industry-standard PCI bus and CT Bus™ technologies. Onboard DSP-based fax and support for software-based speech recognition lets developers maximize the number of boards in the system for multimedia communications applications such as Web-enabled call centers, unified messaging, or speech-enabled interactive voice response (IVR). 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 leveraging existing legacy messaging systems. In addition, support under GlobalCall™ and CT Media™ software facilitate global deployment and add the flexibility to scale systems to meet the growing needs of your business.

Configurations

Use SingleSpan™-JCT boards to develop sophisticated, multimedia communications systems incorporating capabilities such as voice processing, facsimile, text-to-speech (TTS), and automatic speech recognition (ASR). These boards share a common hardware and software architecture with other Dialogic 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 ported easily to lower- or higher-density platforms, with only minimum modifications.

Features and Benefits

- Supports G.726 bit exact and GSM coders, letting developers implement unified messaging applications that meet Voice Profile for Internet Messaging (VPIM) standards
- Offered in industry-standard 32-bit PCI form factor for increased performance
- High channel-per-slot density: one T-1 ISDN span with 24 channels of voice processing or one E-1 ISDN PRI span with 30 channels of voice processing
- Supports DSP-based onboard fax and host-based speech recognition to maximize the number of boards in the system*
- Enables system integrators and developers to lower costs by incorporating more ports per chassis, using less expensive desktop-style machines, and easing configuration/ installation effort
- H.100 connector lets developers take advantage of the industry-standard CT Bus and increases the board's - capacity to interoperate with other CT Bus compatible boards
- SCSA™ SCbus™ connectivity through a simple cable adapter enables applications to access additional re-sources such as TTS and ASR

Dialogic Corporation
1515 Route Ten
Parsippany, NJ 07054
1-973-993-3000
fax: 1-973-993-3093
http://www.dialogic.com

Dialogic North American Sales
1-800-755-4444
or 1-973-993-3030
fax orders: 1-973-631-9631
sales@dialogic.com

Dialogic Latin America & Caribbean
(Buenos Aires)
54-11-4328-1531
fax: 54-11-4328-5425
dlac.sales@dialogic.com

Dialogic Europe
(Brussels)
32-2-712-4311
fax: 32-2-712-4300
info.europe@dialogic.com

Dialogic Japan
(Tokyo)
81-3-5430-3252
fax: 81-3-5430-3373
japan.sales@dialogic.com

Dialogic China
(Beijing)
86-10-6590-0055
fax: 86-10-6590-7989
china.sales@dialogic.com

Dialogic Asia
(Singapore)
65-339-9833
fax: 65-339-9211
asia.sales@dialogic.com

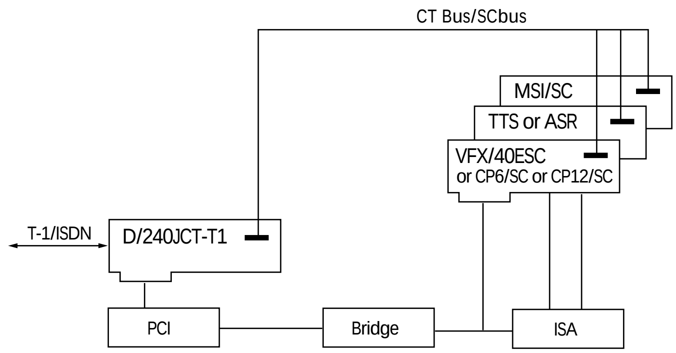
Features and Benefits (cont.)

- Dialogic® SpringWare™ downloadable signal and call processing firmware provides easy feature enhancement and field-proven performance based on over four million installed ports
- PerfectDigit™ DTMF (touchtone) provides reliable detection during voice playback — lets caller "type-ahead" through menus
- Three (T-1) or four (E-1) independent Motorola 56303 DSPs, clocked at up to 100 MHz; each with private, high-speed SRAM, permit execution of high-performance SpringWare signal processing algorithms
- Two Intel486® GX microprocessors offload call processing tasks from host PC, providing more power to the application
- Configure multiple boards in a single PC for easy and cost-effective system expansion on the computing platform that best fits your needs
- Support under Dialogic GlobalCall™ software lets the same application work on multiple signaling systems worldwide (e.g., ISDN, T-1 robbed-bit, R2/MF, pulsed, MF Socotel)
- Supports ISDN Primary Rate Interface (PRI)
- Supports the BoardWatch™ tool, the SNMP-compatible software for remote CT board management

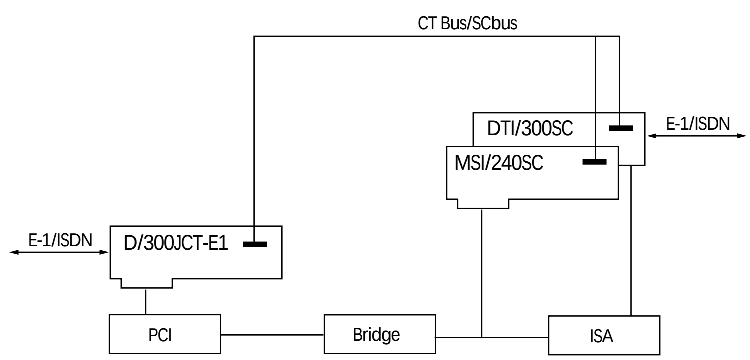
Fax and host-based speech recognition are mutually exclusive and are available through a software upgrade in SR 5.01. Host-based speech-recognition is available through Continuous Speech Processing (CSP) feature.

SingleSpan-JCT boards install in any PCI-based PC or server (PCI bus or mixed PCI/ISA bus) and compatible computers (Intel® 386 and 486, Intel® Pentium Digital AlphaServer™, or Sun UltraSparc). Each board occupies a single expansion slot and up to 10 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.

SingleSpan-JCT boards can operate in either terminate or drop-and-insert configurations. In a terminate configuration, the board handles the call processing of the digital audio and telephony signaling, facsimile, and the host-based speech recognition. If additional resources are required, such as TTS, these resources can be switched to the call via the CT Bus/SCbus. A D/240JCT-T1 or D/300JCT-E1 board installed as a terminating device eliminates the need for a channel bank. The system operates as a standalone call-processing node.



In a drop-and-insert configuration, use SingleSpan-JCT boards and a DTI™ board connected via the CT Bus/SCbus to pass T-1 or E-1 time slots through to each other. This configuration joins two separate T-1 or E-1 lines, or it can be placed in-line between a T-1 or E-1 line and a switch (a PBX, for example). Calls on individual channels can either terminate at a call processing resource on a SingleSpan-JCT series board, or "flow through" transparently to the DTI board.



Applications

- Voice-enabled e-commerce or voice portal solutions
- Unified messaging
- Follow-me/one-number service
- Voice messaging
- Speech-enabled interactive voice response
- Web-enabled call center

ISDN-PRI Support

The Dialogic ISDN Primary Rate Interface (PRI) firmware is a feature enhancement to the DIALOG/HD™ Voice and Switching Products Series. The Dialogic PRI firmware is approved for use with many popular protocols in major markets, based on both T-1 (1.544 Mb/s) and E-1 (2.048 Mb/s) physical interfaces.

Features and benefits of ISDN PRI include

- ISDN PRI connectivity to Dialogic CT systems
- Dialed Number Identification Service (DNIS) enables the application to route incoming calls by automatically identifying the number the caller dialed

- Automatic Number Identification (ANI) enables the application to identify the calling party
- ANI-on-Demand feature saves money by selectively requesting ANI information only when needed
- ISDN offers inherent benefits to call center applications with its fast call setup and fast retrieval of DNIS and ANI information on inbound calls
- Call-By-Call Service Selection lets an application select the most efficient bearer channel service on a call-by-call basis
- Subaddressing allows direct connection to individual extensions or devices sharing the same phone number, or as a proprietary messaging mechanism
- Powerful and universal software interface simplifies access for developers who are unfamiliar with ISDN, yet enables sophisticated control of features
- Multinational approvals with many popular protocols
- User-to-User Information lets an application send proprietary messages to remote systems during call establishment
- Facility, Notify, and optional Information Elements (IEs) let applications work with network-specific supplementary services
- Layer 2 access empowers developers to build customized Layer 3 protocol.

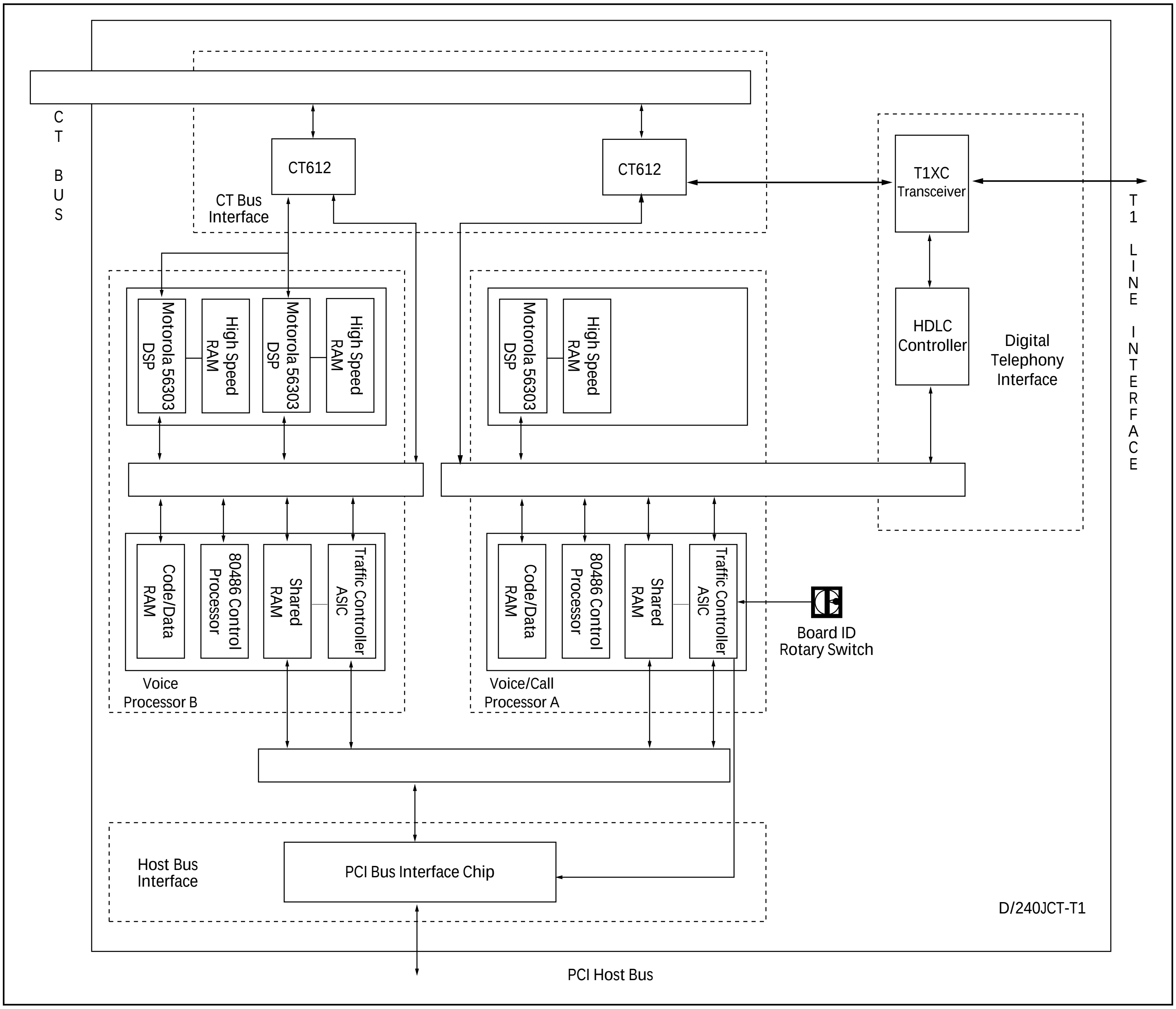
Software Support

SingleSpan-JCT boards are supported by the Dialogic System Software and Software Development Kits for UNIX, Solaris, and Windows NT. These packages contain a set of tools for developing sophisticated, multimedia communications applications.

SingleSpan-JCT boards can use GlobalCall™ software, a call control interface that simplifies the development and use of compelled R2 and other special signaling protocols.

These boards also support the BoardWatch™ tool, the SNMP-compatible software for remote CT board management. BoardWatch 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.

Functional Description



D/240JCT-T1

The D/240JCT-T1 board connects directly to a channel service unit (CSU), digital service unit (DSU), or to other network terminating equipment. The CSU chosen must support the D4 or ESF (within ISDN) superframe format. Most functions traditionally performed by a DSU (such as unipolar to bipolar format conversion, framing, etc.) are performed by the D/240JCT-T1 board. (The only exception is the ability to interpret certain bipolar violation patterns such as loopback start and stop commands from the T-1 network.)

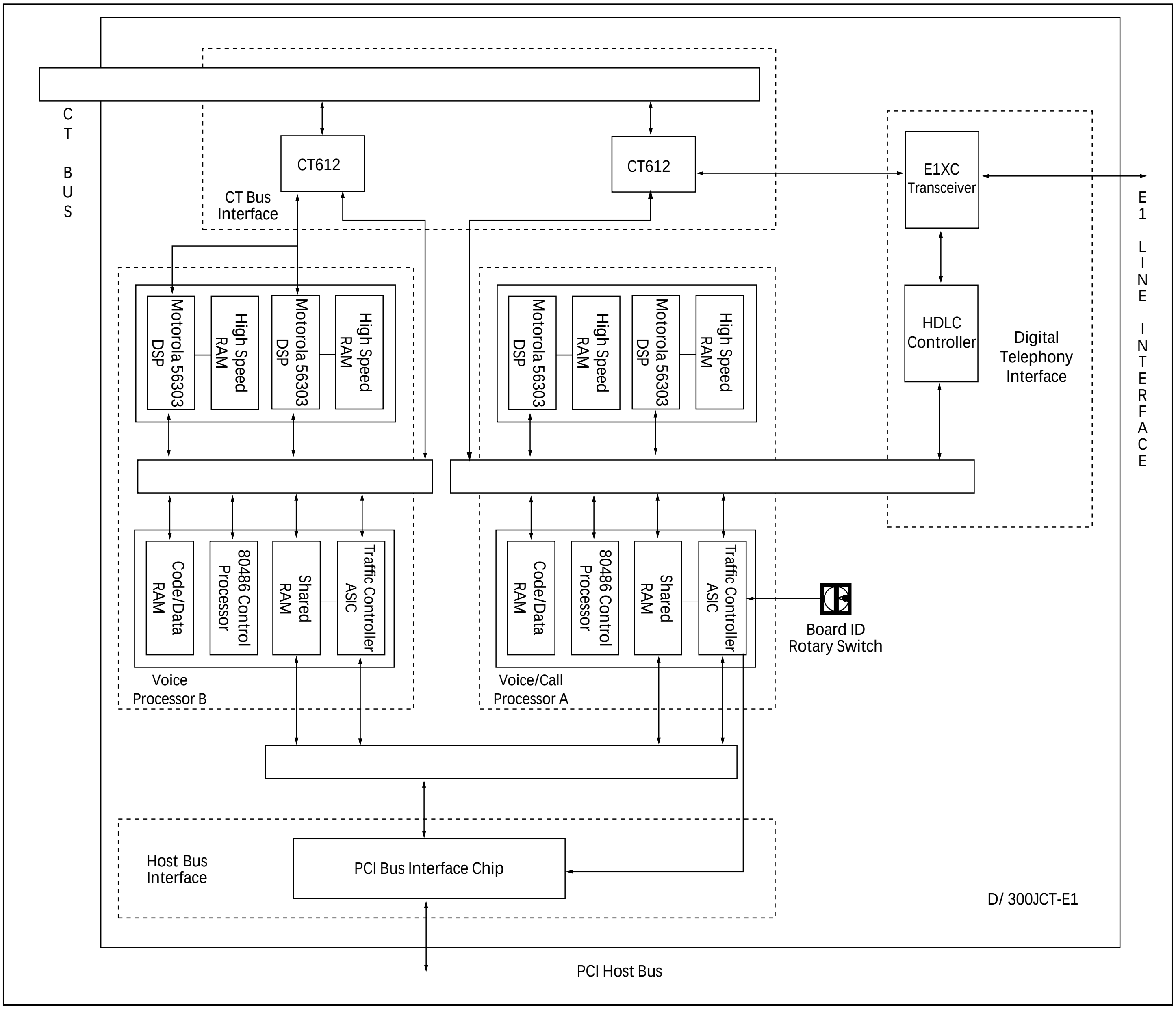
The board processes the digital on-hook/off-hook signaling information and digital voice signals from the telephone network. Digital T-1 signals enter the board via a T1XC line interface (see block diagram). The line interface contains a software-switchable clock that can be set to any of the following settings:

- Loop (clocking is slaved to the external network)
- Independent (clocking is derived from an onboard oscillator)
- Expansion (clocking is slaved to another bus clock master board)

The incoming T-1 bit stream is applied to a CT612 chip, which acts as a traffic coordinator for each channel and as an interface to the CT Bus. This serial bit stream contains the digitized voice data and the signaling information for the incoming call.

Each of two CT612 functional modules on the D/240JCT-T1 board transmits several lower speed data streams over a single high-speed channel. The bus configuration is set when the firmware is downloaded at system initialization. These chips incorporate matrix switching capabilities. Under control of an onboard control processor, a CT612 functional module can connect a call being processed or an available external resource to any of the 1024 CT Bus time slots. This lets the application route calls to any added resources such as fax, TTS, or ASR.

Functional Description



A DSP resource receives digital voice data via a CT612 module. The DSP processes the data based on SpringWare firmware loaded in its high-speed RAM. Each DSP performs the following signal analysis and operations on this incoming data:

- applies automatic gain control to compensate for variations in the level of the incoming audio signal
- applies an Adaptive Differential Pulse Code Modulation (ADPCM), Pulse Code Modulation (PCM), GSM, or G.726 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
- 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

The dual processor combination also performs the following outbound dialing and call progress monitoring functions:

- transmits an off-hook signal to the telephone network
- dials out (makes an outbound call)
- 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, an answering machine, a facsimile machine, or a modem

The board's line interface extracts or inserts telephony information, which is processed by an onboard control processor. The DSPs only process the digitized voice data.

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 processor retrieves the voice information from the host PC and passes it to the DSP, which converts the file into digitized voice. The DSP uses the CT Bus circuitry to send the digitized voice responses to the caller via the T1XC line interface.

For CT Bus/SCbus configurations, the internal local buses operate at 2.048 Mb/s. A High-Level Data Link Controller (HDLC) formats ISDN data. The HDLC receives ISDN signaling data from the T1XC interface and CT612 ASIC and makes it available to the control processor. It also formats and sends outbound signaling data from the control processor to the network interface through the CT612 ASIC and T1XC transceiver chip.

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 and telephony signaling 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 initialized, SpringWare 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 80486 control processor interface that handles all peripheral devices (CT612, HDLC, DSPs, T1XC) and host PC functions (Board Locator Technology™, programmable interrupts, and shared RAM). The Board Locator Technology circuit inside the Traffic Controller ASIC operates in conjunction with a rotary switch, eliminating the need to set confusing jumpers or DIP switches.

D/300JCT-E1

The D/300JCT-E1 board processes the digital on-hook/off-hook signaling information and digital voice signals from the telephone network. Digital E-1 signals enter the board via an E1XC line interface (see block diagram). The line interface supports CRC4 error detection (Cyclic Redundancy Check) and contains a software-switchable clock that can be set to any of the following settings:

- Loop (clocking is slaved to the external network)
- Independent (clocking is derived from an onboard oscillator)
- Expansion (clocking is slaved to another bus clock master board)

Each of two CT612 functional modules on the D/300JCT-E1 board transmits several lower speed data streams over a single high-speed channel. The bus configuration is set when the firmware is downloaded at system initialization. These chips incorporate matrix switching capabilities. Under control of an onboard control processor, a CT612 functional module can connect a call being processed or an available external resource to any of the 1024 CT Bus time slots. This lets the application route calls to any added resources such as fax, TTS, or ASR.

A DSP resource receives digital voice data via a CT612 module. The DSP processes the data based on SpringWare firmware loaded in its high-speed RAM. Each DSP performs the following signal analysis and operations on this incoming data:

- applies automatic gain control 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 — 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

The dual processor combination also performs the following outbound dialing and call progress monitoring functions:

- transmits an off-hook signal to the telephone network
- dials out (makes an outbound call)
- 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, an answering machine, a facsimile machine, or a modem

The board’s line interface extracts or inserts telephony signaling information, which is processed by an onboard control processor. The DSPs only process the digitized voice data.

When recording speech, the DSP can use 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 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. The DSP sends the digitized voice responses to the caller via the CT612 functional modules, the CT Bus, and the E1XC line interface.

For CT Bus/SCbus configurations, the internal local buses operate at 2.048 Mb/s. A High-Level Data Link Controller (HDLC) formats ISDN data. The HDLC receives ISDN signaling data from the E1XC interface and the CT612 and makes it available to the control processor. It also formats and sends outbound signaling data from the control processor to the network interface

Functional Description

through the CT612 ASIC and E1XC transceiver chip.

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 and telephony signaling 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 initialized, SpringWare 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 80486 control processor interface that handles all peripheral devices (CT612, HDLC, DSPs, E1XC) and host PC functions (Board Locator Technology, programmable interrrupts, and shared RAM). The Board Locator Technology circuit inside the Traffic Controller ASIC operates in conjunction with a rotary switch, eliminating the need to set confusing jumpers or DIP switches.

Technical Specifications

D/240JCT-T1

Number of ports	24
Max. boards/system	10 (UNIX, Windows NT). Number may be limited by application, system performance, and the number of CT Bus loads per board.
CT Bus loads per board	2
Maximum CT Bus loads per system	20 (see CT Bus specification for further details)
Digital network interface	Onboard DSX-1 interface
Resource sharing bus	H.100 CT Bus
Control microprocessor	Two Intel 80486 GX @ 32.7 MHz, 0 wait state
Digital signal processors	Three Motorola DSP56303 @ 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 max.
Bus mode	32- to 16-bit conversion in target mode
Shared memory	2 x 64 KB page
I/O ports	None

TELEPHONE INTERFACE:

Clock rate	1.544 Mb/s ±32 ppm
Level	3.0 V (nominal)
Pulse width	323.85 ns (nominal)
Line impedance	100 Ohm ±10%
Other electrical characteristics	Complies with AT&T TR62411 and ANSI T1.403-1989
Framing	SF (D3/D4), ESF for ISDN
Line coding	AMI, AMI with B7 stuffing, B8ZS
Clock and data recovery	Complies with AT&T TR62411 and Bellcore TA-TSY-000170
Jitter tolerance	Complies with AT&T TR62411 and ANSI T1.403-1989
Connectors	RJ-48C
Telephony bus connector	H.100-style 68-pin fine pitch card edge connector
Loopback	Supports switch-selectable local analog loopback and software selectable local digital loopback

Technical Specifications (cont.)

POWER REQUIREMENTS:

+5 VDC	2.0 A typical; 2.2 A max.
+12 VDC	6 mA typical; 6.6 mA max.
–12 VDC	Not required
Operating temperature	0°C to +50°C
Storage temperature	–20°C to +70°C
Humidity	8% to 80% noncondensing
Form factor	PCI long card, 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-20078-XD-N UL: 1950 (E96804)
Canada	IC: 885 5959 A CSA: 950 (LR 84340)
Estimated MTBF	150,000 hours per Bellcore Method I
Warranty	3 years standard

D/300JCT-E1

Number of ports	30
Max. boards/system	10 (UNIX, Windows NT). Number may be limited by application, system performance, and the number of CT Bus loads per board.
CT Bus loads per board	2
Maximum CT Bus loads per system	20 (see CT Bus specification for further details)
Digital network interface	Onboard E-1 interface
Resource sharing bus	H.100 CT Bus
Control microprocessor	Two Intel 80486 GX @ 32.7 MHz, 0 wait state
Digital signal processors	Four Motorola DSP56303 @ 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 max.
Bus mode	32- to 16-bit conversion in target mode
Shared memory	2 x 64 KB page
I/O ports	None

TELEPHONE INTERFACE:

Network clock rate	2.048 Mb/s ±50 ppm
Internal clock rate	2.048 Mb/s ±32 ppm
Level	2.37 V (nominal) for 75 Ohm or 3.0 V (nominal) for 120 Ohm lines
Pulse width	244 ns (nominal)
Line impedance	75 Ohm, unbalanced or 120 Ohm, balanced
Other electrical characteristics	Complies with CCITT Rec. G.703
Framing	CCITT G.704-1988 with CRC4
Line coding	HDB3
Clock and data recovery	Complies with CCITT Rec. G.823-1988
Jitter tolerance	Complies with CCITT Rec. G.823, G.737, G.739, G.742-1988
Connectors	BNC for 75 Ohm or RJ-48C for 120 Ohm lines
Telephony bus connector	H.100-style 68-pin fine pitch card edge connector
Loopback	Supports switch-selectable local analog loopback and software selectable local digital loopback

Technical Specifications (cont.)

POWER REQUIREMENTS:

+5 VDC	2.0 A typical; 2.2 A max.
+12 VDC	6 mA typical; 6.6 mA max.
–12 VDC	Not required
Operating temperature	0°C to +50°C
Storage temperature	–20°C to +70°C
Humidity	8% to 80% noncondensing
Form factor	PCI long card, 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-20078-XD-N UL: 1950 (E96804)
Canada	IC: 885 5959 A UL: CSA: 950 (E96804)
Estimated MTBF	150,000 hours per Bellcore Method I
Warranty	3 years standard

FACSIMILE:

Fax compatibility	ITU-T G3 compliant (T.4, T.30), ETSI NET/30 and T.6 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, 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, MR, and MMR
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.), and 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.

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 and frequency variation	(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 depend-ent 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
Max. number of tones	Application-dependent
Frequency range	Programmable within 300 to 3500 Hz
Max. 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 to 4000 Hz
Frequency resolution	1 Hz
Duration	10 ms increments
Amplitude	(T-1) –43 dBm0 to –3 dBm0 per tone nominal, programmable (E-1) –40 dBm0 to +0 dBm0 per tone nominal, programmable

MF SIGNALING (T-1):	**R1**
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 dBm0 to –3 dBm0 per tone
Acceptable twist	6 dB
Acceptable freq. variation	Less than ±1 Hz

SpringWareTechnical Specifications (cont.)

MF SIGNALING (E-1):	**R2**
MF digits	All 15 forward and backward signal tones per CCITT Q.441
Transmit level	–8 dBm0 per tone, nominal, per CCITT Q.454; programmable
Signaling mechanism	Supports the R2 compelled signaling cycle and non-compelled pulse requirements per CCITT Q.457 and Q.442
Dynamic range for detection	–35 dBm0 to –5 dBm0 per tone
Acceptable twist	6 dB
Acceptable freq. variation	Less than ±1 Hz
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. CAS tone generation and DTMF detection per Bellcore TR-NWT-001273

*All specifications are subject to change without notice.

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

Hardware System Requirements

D/240JCT-T1 and D/300JCT-E1

- 80386, 80486, or Pentium® microprocessor 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/240JCT-T1 and D/300JCT-E1

- Multidrop CT Bus cables
- CT Bus/SCbus Adapter