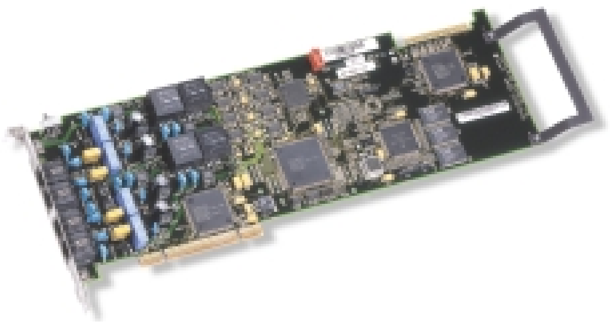
Intel in
Communications

Intel® Dialogic® VFX/41JCT-LS Four-Port Analog Converged Communications Board

The Intel® Dialogic® VFX/41JCT-LS is a four-port analog converged communications board used for developing global enterprise applications including unified messaging, interactive voice response (IVR), and contact centers.



Features and Benefits

Supports continuous speech processing, a flexible speech processing technology that couples with efficient drivers to offload crucial real-time signal processing in speech-enabled applications to onboard DSPs. This reduces system latency, increases recognition accuracy, and improves overall system response time for high-density speech solutions.

With a variety of international approvals, the board expands an application's ability to serve several global market segments at no extra cost.

CT Bus connector increases the board's capacity to interoperate with other CT Bus/SCbus compatible boards.

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

Configure multiple boards in a single chassis (PCI bus or mixed PCI/ISA bus) for easy and cost-effective system expansion up to 32 analog ports.

Intel® Dialogic® firmware, downloadable signal and call processing firmware, provides field-proven performance based on over four million installed ports with access to future feature enhancements.

PerfectDigit DTMF (touchtone) provides reliable detection during voice playback — lets callers type ahead through menus.

A-law or µ-law voice coding at dynamically selectable data rates, 24 Kb/s to 64 Kb/s, selectable on a channel-by- channel basis for optimal tradeoff between disk storage and voice quality.

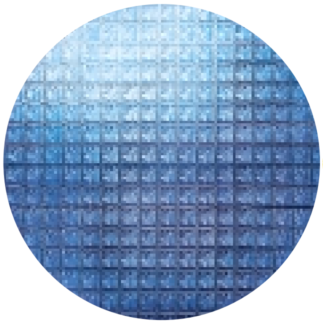
International Caller ID capability via on-hook audio path. Supports Bellcore CLASS*, UK CLI, Japanese Caller ID, and other international protocols.

Intel® NetMerge™ Converged Communications Server Software support facilitates multi-application development.

Patented outbound call progress analysis quickly and accurately monitors outgoing call status.

Software development kits (SDKs) for Windows NT*, Windows* 2000, and Linux* operating systems shorten the development cycle for faster time to market.

Optional onboard global dial pulse detection (global DPD) feature enables callers with non-touchtone phones to access applications without additional pulse-to-tone conversion equipment.



The VFX/41JCT-LS supports voice, enhanced fax, and software-based speech recognition processing in a single PCI slot, providing four analog telephone interface circuits for direct connection to analog loop start lines.

The VFX/41JCT-LS board is ideal for developers building enterprise unified messaging and interactive voice/fax response (IVR) applications for the global market segment. It provides four telephone line interface circuits for direct connection to analog loop start lines. Its dual-processor architecture, consisting of a digital signal processor (DSP) and a general-purpose microprocessor, handles all telephony signaling and performs DTMF (touchtone) and audio/voice and fax signal processing tasks.

Part of the Intel® Dialogic® PCI product family, the VFX/41JCT-LS board conforms to the H.100 CT Bus standard. The open architecture lets developers build converged communications solutions using products from multiple vendors. And since multiple VFX/41JCT-LS boards can be installed in a single PC chassis, developers can build systems scaling up to 32 ports.

Downloaded Spring Ware firmware algorithms, executed by the onboard DSP, provide variable voice coding at 24 and 32 Kb/s Adaptive Differential Pulse Code Modulation (ADPCM), and 48 and 64 Kb/s µ-law or A-law Pulse Code Modulation (PCM). Sampling rates and coding methods are selectable on a channel-by-channel basis. Applications may dynamically switch sampling rate and coding method to optimize data storage or voice quality as the need arises. Additional coding algorithms (such as GSM and G.726) are available for use in applications that support the Voice Profile for Internet Mail (VPIM) standard.

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.

The Intel® Dialogic® global dial pulse detection (global DPD) algorithm, available as a software option for the VFX/41JCT-LS board, lets you use the board in countries that have limited touchtone telephone service. The global DPD product can be optimized on a | country-by-country basis to provide superior dial pulse detection.

With all these advanced features in a half-size PCI board footprint, the VFX/41JCT-LS board is ideal for client or small server system development. The board offers enhanced DSP power and memory capacity that provide a base level of performance for today's requirements as well as the head room for future application expansion via software-based technologies.

The VFX/41JCT-LS board features onboard DSP-based fax, a rich set of fax features that lets developers add fax functionality to their communications applications.

Intel voice and fax products offer a rich set of advanced features, including state-of-the-art DSP technology and signal processing algorithms, for building the core of any converged communications system. With industry-standard PCI bus expansion boards developers can easily and cost effectively integrate Intel® Dialogic®

Fax Features

Supports API-selectable MH, MR, or MMR image compression for send and receive.
MMR image coding option now available; if selected, ECM (per ITU-T T.6) shall be used to detect, replace, and report defective scan lines on both fax send and receive operations.
Supports TIFF/F files for fax send and receive operations.
Supports sending of 7-bit ASCII text files via a host-based ASCII to TIFF/F fax conversion library function.
Supports scaling of documents to three different widths.

Applications

- Voice mail/messaging
- Interactive voice/fax response
- Contact center
- Audiotex
- Operator services
- Dictation
- Auto dialers
- Unified messaging
- Online data entry/query
- Fax mail
- Integrated voice mail and fax mail
- Single call fax-on-demand
- Enhanced messaging

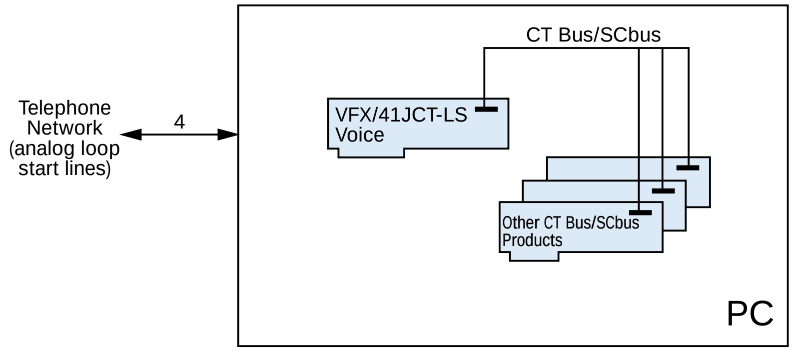


Figure 1: VFX/41JCT-LS Typical Configuration

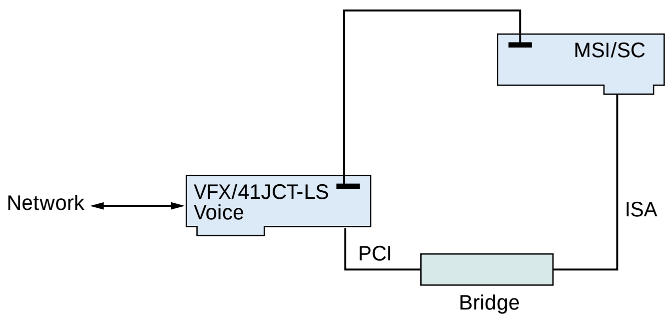


Figure 2: Mixed PCI/ISA Configuration Example

voice and fax products into exactly the type of system they require, with superior performance.

Configurations

Use the VFX/41JCT-LS board to build sophisticated computer telephony (CT) systems with capabilities like speech recognition, facsimile, and text-to-speech (TTS). The VFX/41JCT-LS board shares a common hardware and firmware architecture with other Intel boards based on the CT Bus and SCbus standards for maximum flexibility and scalability. Add features or grow the system while protecting your investment in hardware and application code. Easily port applications to lower- or higher-line-density platforms with minimal modifications.

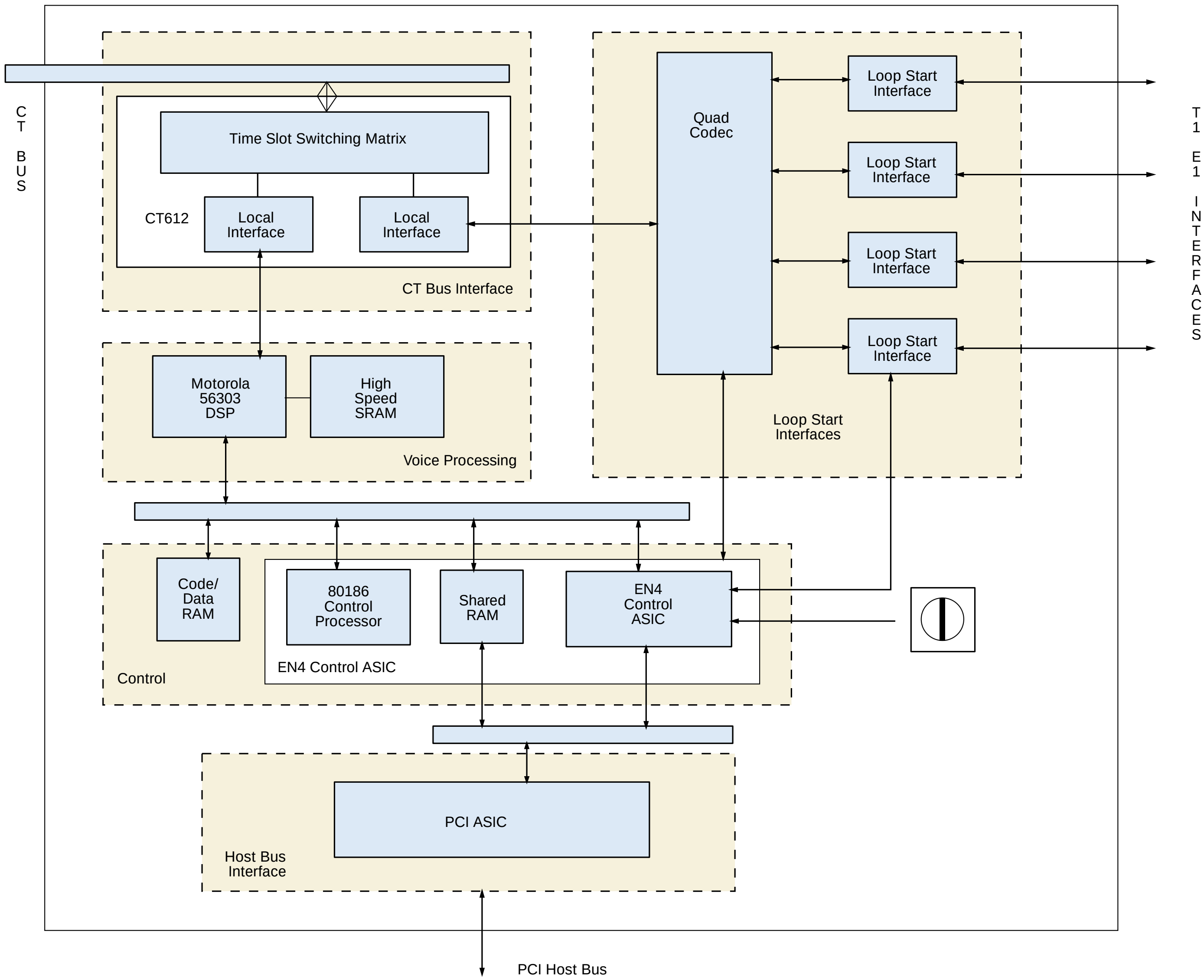
The VFX/41JCT-LS board installs in any PCI-based PC or server (PCI bus or mixed PCI/ISA) and compatible computers (Intel386™, Intel486™, or Pentium® processor-based PC platforms). The VFX/41JCT-LS board provides everything required for building integrated voice and fax solutions scalable from four ports to 32 ports. The maximum number of lines that can be supported is dependent on the application, the amount of disk I/O required, and the host computer CPU and power supply.

The VFX/41JCT-LS board can operate within a mixed chassis containing Intel PCI and ISA products. The forward-looking design of the board conforms to the new H.100 CT Bus to enable connection to next-generation CT Bus products. The VFX/41JCT-LS board can also connect to existing SCbus products through the use of an optional CT Bus/SCbus adapter. The adapter provides both SCbus and H.100 physical connectors required to link the board to current SCbus products.

Software Support

Intel® Dialogic® System Release (SR) software and Software Development Kit (SDK) for Windows NT* and Windows* 2000 and the SR software and SDK for Linux* operating systems support the VFX/41JCT-LS board. These packages contain a set of tools for developing complex multi-channel applications.

Applications developed on one platform can be easily ported to the other. A C++ language-based library/application programming interface (API) lets users develop a wide variety of applications. The same commands can be used to dial and receive voice and fax calls. The application can then play or record voice files, or send or receive fax files in the same session, providing the utmost application flexibility. An API with



Functional Block Diagram

logical addressing resource management eliminates the need to keep track of physical resource addresses such as time slots and board numbers.

Functional Description

The VFX/41JCT-LS board uses 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 80186 control microprocessor. This dual-processor approach offloads many low-level decision-making tasks from the host computer and thus enables 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 DTMF and telephony signaling, and frees the DSP to perform signal processing on the incoming call.

Each of four analog loop start telephone line interfaces on the VFX/41JCT-LS board receives analog voice, fax, and telephony signaling information from the telephone network (see block diagram). Each telephone line interface uses reliable, solid-state hook switches (no mechanical contacts) and FCC-part 68 class B ring detection circuitry. This FCC-approved ring detector is less susceptible to spurious rings created by random voltage fluctuations on the network. 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.

Inbound telephony signaling (ring detection, loop-current detection, and Caller ID information) and fax signals are detected 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 (and the fax) signal from the network is bandpass filtered and conditioned by the line interface and then applied to a codec (COder/DECoder) circuit. The codec filters, samples, and digitizes the inbound analog audio/fax signal and passes this signal to a Motorola* 56303 DSP.

Based on Spring Ware firmware loaded in DSP RAM, the DSP performs the following signal analysis and operations on this incoming data:

- Automatic gain control (AGC) to compensate for variations in the level of the incoming audio signal
- Applies an ADPCM or 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 tone
- Silence detection 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 play-back
- 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

The VFX/41JCT-LS board also supports optional Global DPD software that recognizes dial pulse digits even in the most difficult telephony environments.

When recording speech, the DSP can use different 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 via 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 digitized voice and appropriate signaling responses to the CODEC to be converted into analog format for transmission to the telephone network.

Signaling data (on-/off-hook, ringing, Caller ID, etc.) is passed to the onboard control processor and transmitted to the application via a dual-port shared RAM and the host PCI bus.

When using the VFX/41JCT-LS board and the CT Bus, digital voice and signaling information from a network board or other resource enter the board via the H.100 connector and CT Bus interface. A CT612 chip manages these signals and acts as the traffic coordinator and matrix switch to buffer the high-speed digital data from the bus until the data for each channel can be transmitted to the DSP.

The CT612 chip transmits several lower speed data streams over the CT Bus high-speed channel. The bus configuration is set when the firmware is downloaded at system initialization. This chip incorporates matrix switching capabilities. Under control of the onboard control processor, the CT612 chip can connect any call being processed to any of the four analog lines or to any of the 4096 CT Bus time slots. This enables the application to switch calls to or from other resources, such as facsimile or speech recognition, as they are needed, or to reroute calls.

The onboard control processor controls all operations of the VFX/41JCT-LS board via a local bus and interprets and executes commands from the host PC. The processor 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 a processor 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. 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 downloadable firmware gives the board all of its intelligence and enables easy feature enhancement and upgrades.

With the rotary switch on the VFX/41JCT-LS board set to 0, the board is Plug and Play enabled. Configuration is handled exclusively by software. Alternatively, developers can set the rotary switch to another value to manually control board location for ease of cabling or backwards compatibility with Intel board locator technology (BLT) installation.

Technical Specifications**

	Number of ports	4
	Max. boards/system	8
	Analog network interface	Onboard loop-start interface circuits
	Control processor	80C186 @ 34.8 MHz
	Digital signal processor	Motorola* DSP56303 @100 MHz, with 128Kx24 private SRAM
Host Interface		
	Bus compatibility	PCI (complies with PCISIG Bus Specification, Rev. 2.1)
	Bus speed	33 MHz maximum
	Bus mode	Target mode operation only
	Shared memory	32 KB page
	I/O ports	None
Telephone Interface		
	Trunk type	Loop start
	Loop current range	20 mA to 120 mA
	Impedance	600 Ohm nominal
	Ring detection	15 Vrms min., 13 Hz to 68 Hz, (configurable by parameter[†])
	Echo return loss	Configurable by software parameter
	Crosstalk coupling	Less than -70 dB at 1 KHz channel to channel
	Receive signal/noise ratio	70 dB referenced to -15 dBm
	Frequency response	200 Hz to 3400 Hz ±3 dB (transmit and receive)
	Connectors	Four RJ-11 type
Environmental Requirements		
	+5 VDC	750 mA max.
	+12 VDC	200 mA max.
	–12 VDC	100 mA max.
	Operating temperature	0°C to +50°C
	Storage temperature	–20°C to +70°C
	Humidity	8% to 80% non-condensing
	Form factor	Universal slot (5 V or 3.3 V) 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 15 class A; FCC Part 68 EBZUSA-75385-VM-T UL: E96804 UL1950
	Canada	DOC: 885-5542A For specific country approval designation, see the Intel Communications Systems Products Global Product Approvals listing on the Intel Web site or contact your Intel technical sales representative
	Estimated MTBF	274,000 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 T.4 (Group III), T.30 ETSI NET/30 compliant
	Data rate	Transmission: 14,400 b/s (v.17) (max)
	Variable speed selection	Automatic step-down to 12,000 b/s, 9600 b/s, and lower Reception: 9,600 b/s
	Transmit data modes	API-selectable Modified Huffman (MH) Modified Read (MR) Modified Modified Read (MMR) with Error Correction Mode (ECM)
	Receive data modes	API-selectable MH, MR, and MMR with ECM
	File data formats	Tagged Image File Format (TIFF) for transmit/receive MH, MMR, and ASCII text transmit
	ASCII-to-fax conversion	Performed on the host CPU rather than in the Intel firmware. Supports multiple fonts and language character sets, including all Windows fonts.
	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 among any of the three supported widths
	Polling modes	Normal and turnaround
	Image resolution	Normal (203 pels/in. × 98 lines/in.) Fine (203 pels/in. × 196 lines/in.)
	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 –18 dBm results in full-scale recording, configurable by parameter.†
	Silence detection	–38 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		
	13 Kb/s	GSM @ 8 kHz sampling
	24 Kb/s	ADPCM @ 6 kHz sampling
	32 Kb/s	ADPCM @ 8 kHz sampling
	32 Kb/s	G.726 @ 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 Bellcore LSSGR Sec 6
	Dynamic range	-45 dBm to +3 dBm 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	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	±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
	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 +3 dBm per tone
	Acceptable twist	6 dB
	Acceptable freq. variation	Less than ±1 Hz

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	>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	80% to 90% based on application and environment
	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 dial tone drop out debouncing
Tone Dialing		
	DTMF digits	0 to 9, *, #, A, B, C, D per Bellcore LSSGR Sec 6, TR-NWT-000506
	MF digits	0 to 9, KP, ST, ST1, ST2, ST3
	Frequency variation	Less than ±1 Hz
	Rate	10 digits/s, configurable by parameter†
	Level	–4 dBm0 per tone, nominal, configurable by parameter†
Pulse Dialing		
	10 digits	0 to 9
	Pulsing rate	10 pulses/s, nominal, configurable by parameter† 20 pulses/s for Japan, 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.	

† 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.
‡ Average speech mandates +16 dB peaks above average and preserves -13 dB valleys below average.

Hardware System Requirements

- Intel386™, Intel486™, 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

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

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