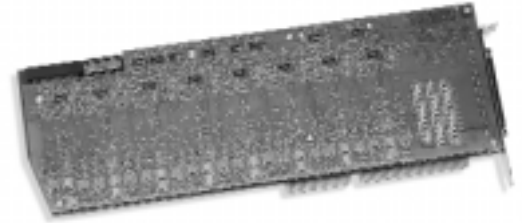




# D/160SC-LS Voice Board

## 16-Port Voice Processing and Analog Interface Board

The D/160SC-LS board from Intel provides 16 channels of call processing and loop start interfaces in a single PC slot. A dual-processor architecture, comprising a digital signal processor (DSP) and a general-purpose microprocessor, handles all telephony signaling and performs all DTMF (touch-tone) and audio/voice signal processing tasks.



### Features and Benefits

High-density analog interface voice processing platform helps system integrators and developers lower costs by incorporating more ports per chassis, by using less expensive desktop-style machines, and by easing configuration/installation effort

Sixteen independent loop start telephone interfaces, combined with 16 channels of voice processing in one ISA slot, provide effective solutions for building high-density applications

Create more cost-effective switching solutions via access to the SCbus with its 1024 time slot capability; SCx bus interbox communications provide the capability to build higher density systems and large, multinode systems

Downloadable signal and call processing firmware provides field-proven performance based on over ten million installed ports

DTMF (touch-tone) provides reliable detection during voice playback — lets callers “type-ahead” through menus

Optional Global Dial Pulse Detection feature lets callers without touch-tone phones access applications. No additional “pulse-to-tone converter” hardware is needed.

Two independent Motorola\* 56002 DSPs, clocked at 65 MHz; each with private, high-speed SRAM, permit execution of Spring Ware signal processing algorithms

Intel486™ GX microprocessor offloads call processing tasks from host PC, giving more power to the application

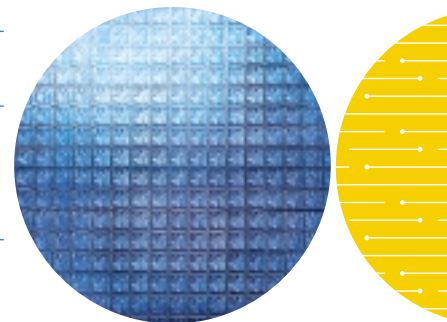
Board locator technology eliminates confusing DIP switch or jumper settings and simplifies installation

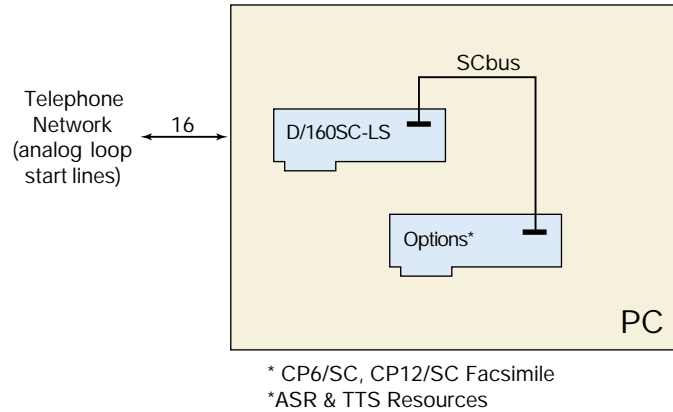
C language APIs for Windows NT\*, Windows\* 2000, and Linux\* operating systems shorten development cycles so you can get your applications into the marketplace faster

Caller ID capability for “screen pop” applications (supports Bellcore\* CLASS\* Protocols)

Configure multiple boards in a single PC (ISA bus) for easy and cost-effective system expansion

Supported by Intel® NetMerge™ Converged Communications Software, a standards-based development platform for building scalable applications that can run with other vendors' products on converged communications servers





**Configuration Diagram**

Downloaded firmware algorithms provide variable voice coding at 24 Kb/s and 32 Kb/s Adaptive Differential Pulse Code Modulation (ADPCM), and 48 Kb/s and 64 Kb/s Pulse Code Modulation (PCM)  $\mu$ -law or A-law. 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. Firmware also provides reliable DTMF detection, DTMF cut-through, and talk off/play off suppression over a wide variety of telephone line conditions.

Offered as a software option, Global Dial Pulse Detection converts rotary pulses to DTMF in countries that have limited touch-tone telephone service. The Global DPD product can be optimized on a country-by-country basis to provide superior dial pulse detection and conversion.

The D/160SC-LS voice board

- Connects to 16 analog loop start telephone channels
- Answers calls
- Detects touch-tone
- Plays voice messages to a caller

- Digitizes, compresses, and records voice signals
- Places outbound calls and automatically reports the result

And it does these things in real time on all channels.

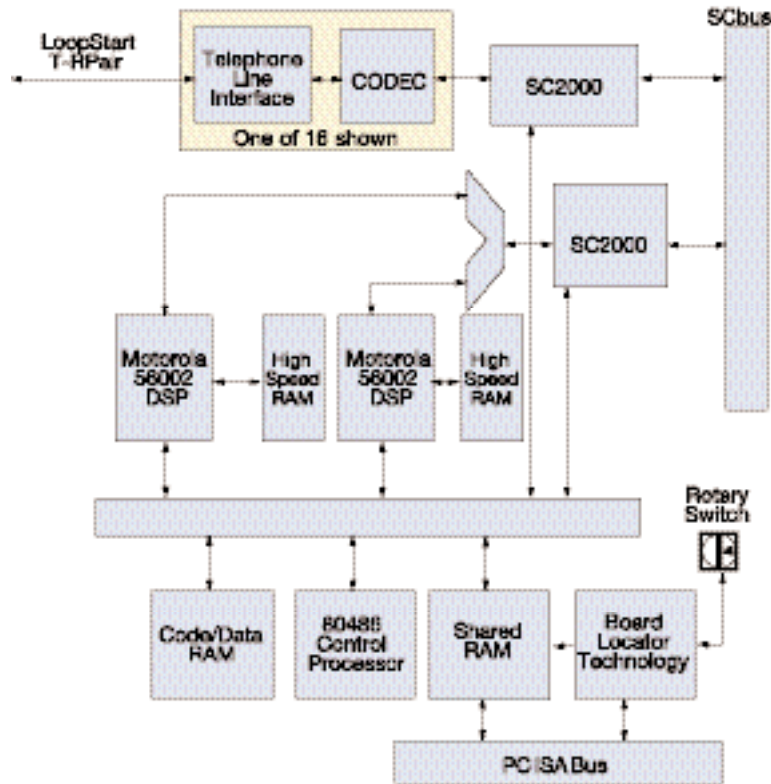
## Configurations

Use the D/160SC-LS board to develop sophisticated, multifunction CT systems incorporating capabilities such as voice processing, speech recognition, and text-to-speech (TTS). The D/160SC-LS board shares a common hardware and firmware architecture with other SCbus-based Intel® Dialogic® boards for maximum flexibility and scalability. Add features or grow the system while protecting your investment in hardware and application code. Applications can be easily ported to lower or higher line-density platforms with only minimal modifications

The D/160SC-LS board installs in IBM PC AT\* (ISA bus) and compatible computers (PC platforms based on Intel386™, Intel486™, and Pentium® processors). The D/160SC-LS board occupies a single expansion slot and up to 16 boards can be configured in a system

## Applications

- Voice messaging
- Interactive voice response
- Voice/audio response systems
- Audiotex
- Operator services
- Telemarketing/call center
- Call logging
- Dictation
- Auto dialers
- Notification systems
- Online data entry/query



#### Functional Description

with each board sharing the same interrupt level. The maximum number of lines that can be supported is dependent on the application type, the amount of disk I/O required, and the host computer CPU and power supply.

### Software Support

The D/160SC-LS board is supported by System Software and Software Development Kits (SDKs) for Windows NT\*, Windows 2000\*, and Linux\* operating systems. These packages contain a set of tools for developing complex multichannel applications.

For added flexibility, the D/160SC-LS board is also supported in Intel® NetMerge™ Converged Communications Software. This resource management software makes application development easier and helps applications from different suppliers written to standard application program interfaces (APIs) like ECTF S.100\* work together for converged communications. The server software makes this possible by managing technology resources (boards and host-based technologies) within the server and by providing basic switching functions to multiple client applications.

A minimum of 35 MB should be used in any field-deployable system, with 64 MB or more recommended. A processor (like the Pentium processor) is recommended, as well as a fast (i.e., SCSI II) disk I/O system.

### Functional Description

The D/160SC-LS board connects 16 analog (loop start) telephone lines to 16 on-board call processing resources or to other resources via the SCbus. This board provides

- Interference suppression
- Ring and on-hook/off-hook signaling control
- Tone detection and generation
- Digitization and playback of voice files

The signals from the 16 loop start telephone lines connected to the D/160SC-LS board first pass through a telephone line interface that provides transient protection and electromagnetic interference (EMI) suppression (see block diagram). These telephone line interfaces use 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 allows applications to go off-hook any time during ring cadence without damaging the board.

The telephone line interface applies the inbound signal including the ring or other in-band signaling to analog/digital inputs of a signal converter called a COder/DECOder (CODEC) for sampling and digitization. These digitized signals are sent to an SC2000 chip where they are routed via the SCbus either to an onboard DSP or to an external resource on any of the 1024 SCbus time slots. This enables the application to reroute calls to any added resource, such as speech recognition, facsimile, or TTS.

Part of the D/160SC-LS board's telephone interface includes an on-hook audio path that detects Caller ID information. Depending on the level of service offered by the local telephone provider, Caller ID can include the date, time, caller's telephone number, and the name of the person calling (in some enhanced Caller ID environments). The on-hook audio path can also detect touch-tones while the line is on-hook. This capability lets you use the D/160SC-LS board behind private branch exchanges (PBXs) that require on-hook touch-tone detection for their signaling.

When onboard call processing resources are used, the network signals are extracted and passed to the onboard control processor, which can change channel status and communicate channel events to the application via a shared RAM and the host PC ISA bus.

Based on Spring Ware firmware loaded in code/data 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 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
- 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:

- Transmits an off-hook signal to the telephone network
- Dials out (makes 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, or a modem

When recording speech, the DSP can use different digitizing rates from 24 Kb/s 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. The D/160SC-LS board can record incoming signals with the telephony interface in either the high-impedance on-hook state or the normal off-hook state. 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 CODEC, which is controlled by a pair of SC2000 chips. The CODEC converts the digitized voice into analog voice and transmits the voice response to the caller via the telephone line interface. (Although this product is capable of recording incoming signals in an on-hook state, applications such as call logging should use the Intel® Dialogic® D/160SC-LS-HiZ line-tapping board, which is specifically designed for analog high-impedance recording.)

The onboard control processor controls all operations of the D/160SC-LS board via a local bus and interprets and executes commands from the host PC. This 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 the 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 ISA 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 board to control all board operations. This downloadable firmware gives the board all of its intelligence and enables upgrades.

The board locator technology circuit operates in conjunction with a rotary switch that eliminates the need to set confusing jumpers or DIP switches.

## Technical Specifications\*\*

Number of ports	16
Max. boards/system	16 (Windows NT, Windows 2000, Linux). Number may be limited by application and system performance.
Analog network interface	Onboard loop start interface
Resource sharing bus	SCbus or PEB
Control microprocessor	Intel486™ GX @ 32.768 MHz, 0 wait state
Digital signal processors	Two Motorola DSP56002 @ 49 MHz to 66 MHz, each with 32 K word private, 0 wait state SRAM

### Host Interface

Bus compatibility	IEEE P996 ISA compatible (IBM PC AT)
Bus speed	8 MHz typical
Bus mode	Automatically configures to 8- or 16-bit transfer mode
Shared memory	32 KB page
Base addresses	8000h to E800h, on 32K boundaries. All D/SC boards share the same base address. Shared memory is page mapped in/out dynamically as needed.
Interrupt level	IRQ 2/9, 3, 4, 5, 6, 7, 10, 11, 12, 14, 15, software selectable. One IRQ line must be shared by all D/SC boards.
I/O ports	None

### Telephone Interface†

Trunk type	Loop start; also works with ground start for inbound applications
Loop current range	20 mA to 120 mA
Impedance	600 Ohms nominal
Ring detection	40 Vrms to 130 Vrms, 15.3 Hz to 68.0 Hz
Echo return loss	20 dB min.
SNR	-40 dB
Cross talk coupling	-70 dB
Speech digitization	64 Kb/s, $\mu$ -law PCM (companding to ADPCM performed in Spring Ware)
Freq. response	300 Hz to 3400 Hz $\pm$ 3 dB
Connector	DB-37

### Power Requirements

+5 VDC	1.5 A max.
-12 VDC	250 mA max.
Operating temperature	0°C to +50°C
Storage temperature	-20°C to +70°C
Humidity	8% to 80% noncondensing
Form factor	PC AT 13.3 in. (33.25 cm) long 4.5 in. (11.25 cm) high (excluding edge connector)

### Safety and EMI Certifications

United States	UL: 1459, with optional adapter
Canada	CSA: 225 (by UL)
Estimated MTBF	153,000 Hours per Bellcore* Method IIIA
Warranty	Intel® Telecom Products Warranty Information: 3 years standard, see <a href="http://www.intel.com/network/csp/products/3144web.htm">http://www.intel.com/network/csp/products/3144web.htm</a>

## Spring Ware Technical Specifications\*\*

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### Audio Signal

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

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### 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

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### Audio Digitizing

24 Kb/s	OKI* ADPCM @ 6 kHz sampling
32 Kb/s	OKI ADPCM @ 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 Kb/s and 32 Kb/s data rates Adjustment range: ±50% Adjustable through application or programmable DTMF control

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### DTMF Tone Detection

DTMF digits	0 to 9, *, #, A, B, C, D per Bellcore LSSGR Sec 6
Dynamic range	-36 dBm to +3 dBm per tone, configurable by parameter <sup>†</sup>
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	Meets Bellcore LSSGR Sec 6 and EIA 464 requirements
Noise tolerance	Meets Bellcore 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 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.

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### Global Tone Detection

Tone type	Programmable for single or dual
Max. number of tones	Application-dependent
Frequency range	Programmable within 200 Hz to 4000 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	Programmable, default set at -6 dBm0 to +3 dBm0 per tone

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## Spring Ware Technical Specifications\*\* (cont.)

### 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 dBm0 to -3 dBm0 per tone nominal, programmable

### MF Signaling

<b>R1</b>	
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 $\pm 1$ 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 utilizes 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. Utilizes 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 market segments 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	Pre-programmed
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 $\pm 1$ Hz
Rate	10 digits/s, configurable by parameter <sup>†</sup>
Level	-4.0 dBm0 per tone, nominal, configurable by parameter <sup>†</sup>

### Pulse Dialing

10 digits	0 to 9
Pulsing rate	10 pulses/s, nominal, configurable by parameter <sup>†</sup>
Break ratio	60% nominal, configurable by parameter <sup>†</sup>

## Spring Ware Technical Specifications\*\* (cont.)

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### Analog Caller Identification

Applicable standards	Bellcore TR-TSY-000030 Bellcore TR-TSY-000031 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 for error-free performance
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

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### Analog Display Services Interface (ADSI)

FSK generation per Bellcore TR-NWT-000030  
CAS tone generation and DTMF detection per Bellcore TR-NWT-001273

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## Hardware System Requirements

- Intel386, Intel486, or Pentium processor, IBM PC AT (ISA) bus or compatible computer

## Additional Components

- Multidrop SCbus cable
- The D/160SC-LS board is sold without cables and patch panel: SA160CONNKIT (includes cable and mini patch panel)
- For systems that connect directly to the PSTN, the LSI/1459 kit is necessary to comply with the UL 1950 directive. The LSI/1459 kit consists of a small box and cable that installs in line between the mini patch panel and the board: LSI160TOLSI1459.

To learn more, visit our site on the World Wide Web at <http://www.intel.com>.

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

