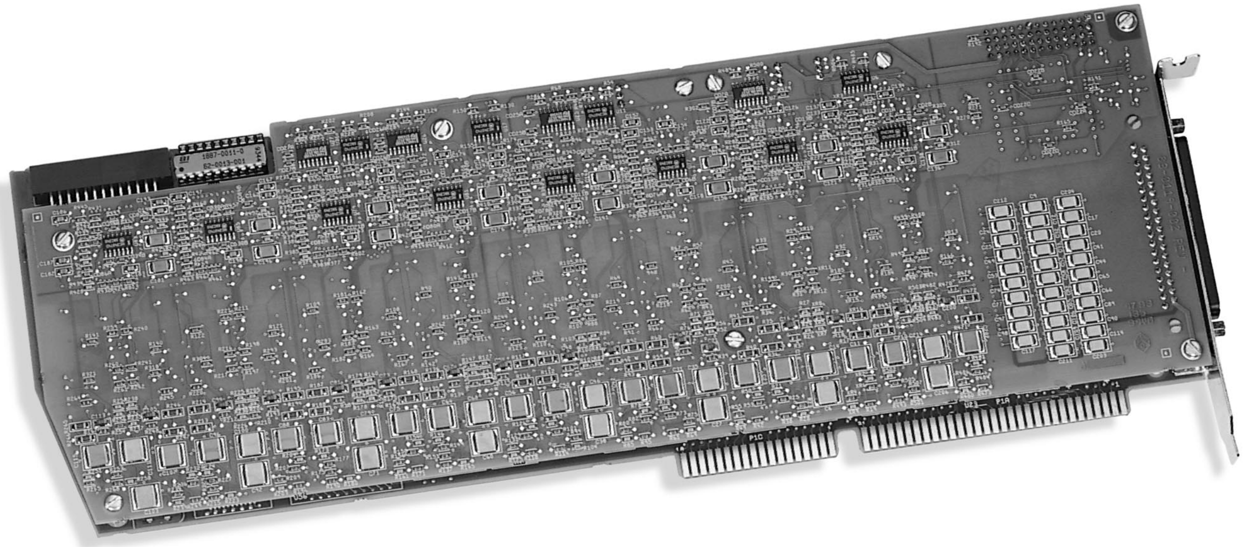

SCSA HARDWARE MODEL | **AEB HARDWARE MODEL** | **PEB HARDWARE MODEL**

D/160SC-LS

16-Port Voice Processing and Analog Interface Board

Dialogic voice 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 computer telephony system. With industry-standard ISA bus expansion boards and a variety of channel densities to choose from, you can integrate Dialogic voice products easily into exactly the type of system you require at a price and performance level unmatched in the computer telephony industry.

The D/160SC-LS™ board provides 16 channels of call processing and loop start interfaces in a single PC slot. A unique dual-processor architecture comprised of digital signal processors (DSPs) and a general purpose microprocessor handles all telephony signaling and performs all DTMF (touchtone) and audio/voice signal processing tasks. The D/160SC-LS board uses the Signal Computing System Architecture™ (SCSA). SCSA provides an open architecture that enables developers to use products from multiple vendors to build a unified computer telephony solution. SCSA provides features such as distributed switching, logical addressing, and location-independent resource management.

Downloaded firmware algorithms such as SpringWare, provide variable voice coding at 24 and 32 Kb/s ADPCM, and 48 and 64 Kb/s PCM μ-law or A-law. 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. SpringWare 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, Dialogic Global Dial Pulse Detection (GDPD) algorithm converts rotary pulses to DTMF in countries that have limited touchtone telephone service. Global DPD is also optimized in several countries, providing superior dial pulse detection and conversion.

SCSA HARDWARE COMPATIBLE

FEATURES AND BENEFITS

- Highest density analog interface voice processing platform in the industry enables system integrators and developers to lower costs by incorporating more ports per chassis, by using less expensive desktop-style machines, and by easing configuration/installation effort
- 16 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 SCSA™ SCbus™ with its 1024 time slot capability; SCxbus™ interbox communications provides the capability to build higher density systems and large, multinode systems
- Downloadable signal and call processing firmware by Dialogic, called SpringWare™, provides easy feature enhancement and field-proven performance based on over three million installed ports

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

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fax: 32-2-712-4300
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(Singapore)
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fax: 65-339-9211
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FEATURES & BENEFITS (cont.)

- PerfectDigit™ DTMF (touchtone) provides reliable detection during voice playback — allows callers to "type-ahead" through menus
- Optional Global Dial Pulse Detection™ (Global DPD™) feature enables callers without touchtone phones to access applications. No additional "pulse to tone converter" hardware is needed.
- Two independent Motorola DSP56002 digital signal processors, clocked at 65 MHz; each with private, high-speed SRAM, permit execution of high performance SpringWare signal processing algorithms
- Intel 486 GX microprocessor off-loads 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 application program interfaces (APIs) for MS-DOS®, UNIX®, Solaris®, and Windows NT® shorten your development cycle so you can get your applications to market faster
- High impedance, on-hook record capability enables high-density call logging and transaction record applications
- 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 on the best computing platform that fits your needs

APPLICATIONS

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

The D/160SC-LS voice board

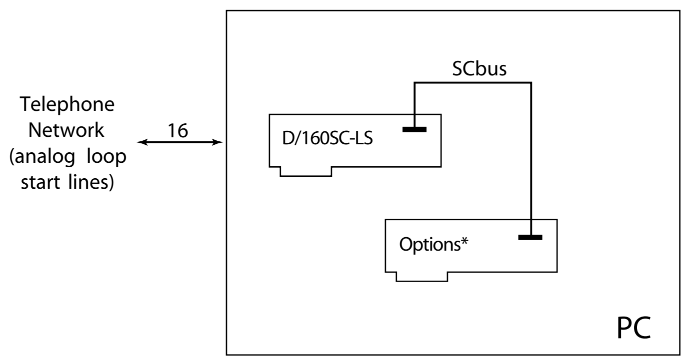
- connects to 16 analog loop start telephone channels
- automatically answers calls
- detects touchtone
- plays voice messages to a caller
- digitizes, compresses, and records voice signals
- places outbound calls and automatically reports the result

in real time on all channels.

CONFIGURATIONS

Use the D/160SC-LS board to develop sophisticated, multifunction computer telephony 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 Dialogic SCbus-based boards for maximum flexibility and scalability. Features can be added or systems can grow while protecting investment in hardware and application code. With only minimum modifications, applications can be easily ported to lower or higher line-density platforms.

The D/160SC-LS board installs in IBM® PC AT® (ISA bus) and compatible computers (80386, 80486, and Pentium™-based PC platforms). The D/160SC-LS board occupies a single expansion slot and up to 16 boards can be configured in a system with each board sharing the same interrupt level. 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.



*CP4/SC, CP6/SC, CP12/SC Facsimile
*ASR & TTS Resources

SOFTWARE SUPPORT

The D/160SC-LS board is supported by Dialogic System Software and SDK for Windows NT®, UNIX®, Solaris®, and MS-DOS®. These packages contain a set of tools for developing complex multichannel applications.

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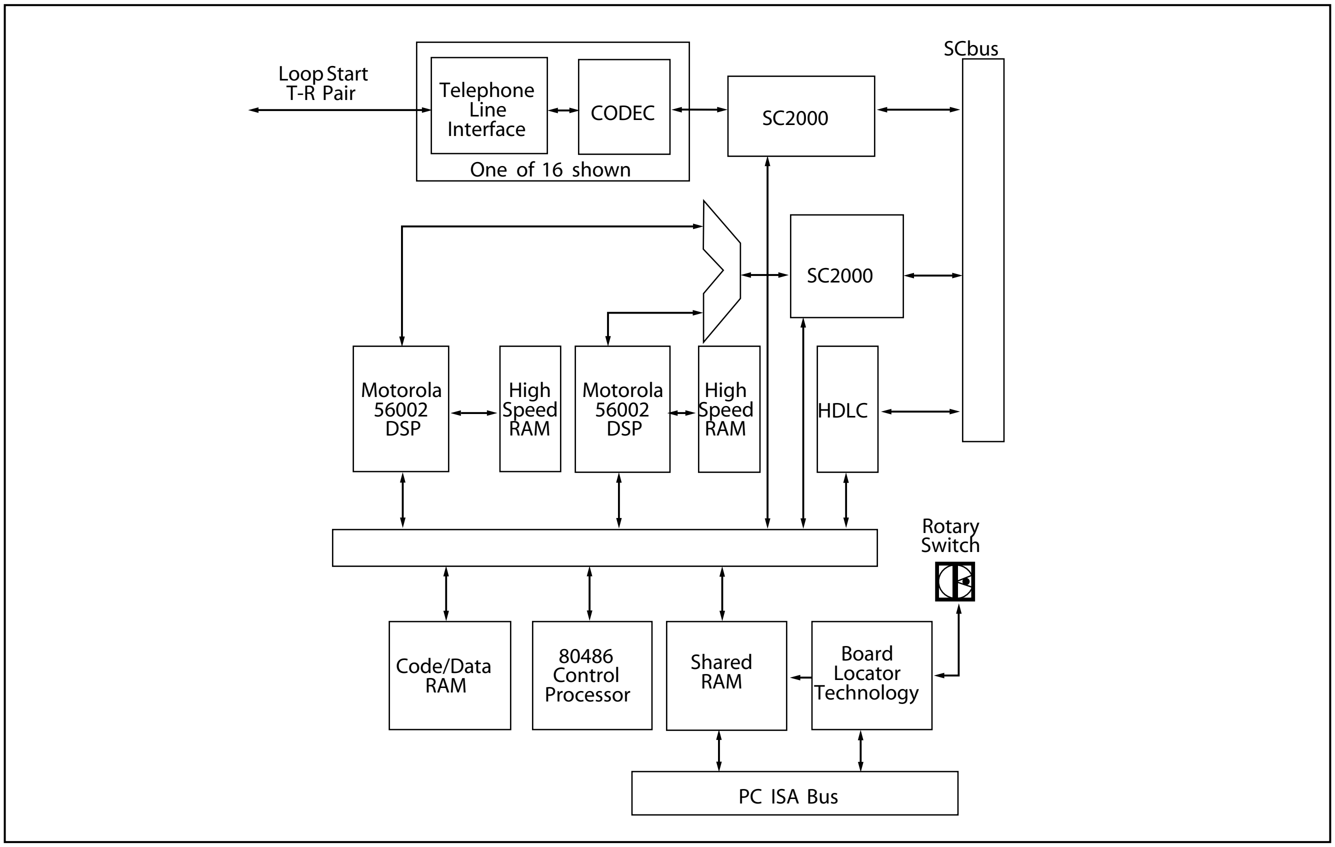
FUNCTIONAL DESCRIPTION



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
- digital-to-analog conversion
- ring and on-hook/off-hook signaling control
- tone detection and generation
- digitization and play back 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 EMI (electromagnetic interference) 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 and on-hook/off-hook signals to analog/digital inputs of a signal converter called a COder/DECoder (CODEC) that samples and digitizes these signals. These digitized signals are sent to an SC2000™ chip where they are routed via the SCbus either to an on-board 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 (in some enhanced caller ID environments) the name of the person calling. The on-hook audio path can also detect touchtones while the line is on-hook. This capability lets you use the D/160SC-LS board behind PBXs that require on-hook touchtone detection for their signaling.

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

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The DSP processes the digitized voice data based on SpringWare firmware loaded in code/data RAM. Each DSP performs the following signal analysis and operations.

On the incoming data:

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

- 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 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 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 sends the voice response to the caller via the telephone line interface.

When the system is initialized, SpringWare firmware to control all board operations is downloaded from the host PC to the board. This downloadable firmware gives the board all of its intelligence and enables easy feature enhancement and upgrades.

The on-board 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. Communication 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.

The Board Locator Technology circuit operates in conjunction with a rotary switch that eliminates the need to set confusing jumpers or DIP switches.

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Technical Specifications*

Number of ports	16
Max. boards/system	6 (MS-DOS); 16 (UNIX, Windows NT). Number may be limited by application and system performance.
Analog network interface	On-board loop start interface
Resource sharing bus	SCbus
Control microprocessor	Intel 80486 GX @ 28.5 MHz, 0 wait state
Digital signal processors	Two Motorola DSP56002 @ 49 - 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 Kbytes page
Base addresses	8000h to E800h, on 3 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
Impedance	600 Ohms nominal
Loop current range	20 to 120 mA
Ring detection	40 to 130 Vrms, 15.3 to 68.0 Hz
Echo return loss	20 dB min.
SNR	–40 dB
Cross talk coupling	–70 dB
Speech digitization	64 Kb/s, μ-law PCM (companding to ADPCM performed in SpringWare)
Freq. response	300 to 3400 Hz ±3 dB
Connector	DB-37
POWER REQUIREMENTS:	
+5 VDC	1.5 A max.
+12 VDC	375 mA 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. long. 0.793 in. wide (total “envelope”), 4.5 in. high (excluding edge connector)
SAFETY & EMI CERTIFICATIONS:	
United States	UL: 1459, with optional adapter
Canada	CSA: 225 (by UL)
Estimated MTBF	105,000 hours per Bellcore Method I
Warranty	3 years standard

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SpringWare Technical Specifications*

AUDIO SIGNAL:

Receive range	–40 to +2.5 dBm0 nominal, configurable by parameter**
Automatic gain control	Application can enable/disable. Above –18 dBm0 results in full scale recording, configurable by parameter.**
Silence detection	–38 dBm nominal, software adjustable**
Transmit level (weighted average)	–9 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:

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 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 Bellcore LSSGR Sec 6
Dynamic range	–36 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 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.

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	Programmable, default set at –6 dBm0 to +3 dBm0 per tone

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SpringWare Technical Specifications* (cont.)

GLOBAL TONE GENERATION™:

Tone type	Generate single or dual tones
Frequency range	Programmable within 200 to 4000 Hz
Frequency resolution	1 Hz
Duration	10 msec increments
Amplitude	–43 dBm0 to –3 dBm0 per tone nominal, programmable
MF SIGNALING:	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

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 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	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 6TR-NWT-000506
Frequency variation	Less than ±1 Hz
Rate	10 digits/s, configurable by parameter**
Level	–4.0 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**

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SpringWare Technical Specifications* (cont.)

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

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

- 80386, 80486, or Pentium IBM PC AT (ISA) bus or compatible computer. Operating system hardware requirements vary according to the number of channels being used.

ADDITIONAL COMPONENTS

- Multidrop SCbus cable
- Required: Station Adapter, 37-pin to 50-pin cable
- Optional: UL 1459 compliance adapter

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