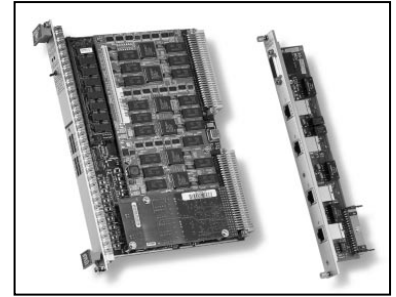
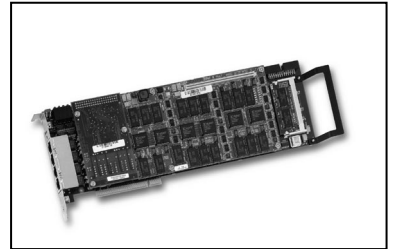


DM/V960-4T1-cPCI

RJ48-T1-cPCI Rear I/O



Also available in VME.



Also available in PCI.

CT Bus

SCbus

PEB

QuadSpan Voice Series

Quad T-1/E-1 ISDN Network Interface with
Up to 120 Ports of Voice Processing and Telephony Signaling

The QuadSpan™ Voice Series of products provides a powerful set of advanced voice processing and telephony networking features that developers can use to create large-scale intelligent peripherals for enhanced services. Offered on a single-slot PCI, CompactPCI® (cPCI) or VME format board, each QuadSpan provides access to four T-1 (1.544 Mb/s) or E-1 (2.048 Mb/s) digital network interfaces, and 48 to 120 ports of voice and telephony signal processing.

Powerful DSPs provide a rich set of voice processing features, including various rates of voice compression, recording and playback, telephony tone signaling, reliable DTMF detection using local echo cancellation, and automated out-bound call progress analysis with Positive Answering Machine Detection™.

The QuadSpan Series supports GlobalCall™ — a unified call control programming interface and protocol engine — which makes it easier for an application to access worldwide digital network interface protocols such as ISDN Primary Rate.

With access to the computer telephony industry standard ECTF H.100/H.110 CT Bus™ and ANSI/VITA SCbus™, applications using the QuadSpan can provide switching capability or expand to include other technologies such as automatic speech recognition (ASR), fax, ATM connectivity, SS7, and IP telephony.

The QuadSpan Series of products is based on the Dialogic DM3™ architecture, which provides a development environment that accelerates application development and provides a path for future growth. Software Development Kits are available for Windows NT®, Solaris® and UnixWare® environments, offering full interoperability with the broad Dialogic CT product line.



FEATURES AND BENEFITS

- Up to 120 ports of voice processing in a single slot enables development of large-scale intelligent peripherals for enhanced services
- Choice of T-1 or E-1 digital network interfaces with internationally approved CAS and ISDN Primary Rate access allows applications to connect to a variety of switches
- Unified call control access through GlobalCall™ interface provides worldwide application portability and shortens development time
- Choice of PCI, CompactPCI®, or VME formats provides a choice of platforms for system integrators
- Built on the DM3™ Mediastream architecture for unmatched performance and modular expandability, protecting developer's investment
- Built on the industry standard CT buses — ECTF H.100/H.110 CT Bus™ and ANSI/VITA SCbus™ allowing for application expansion through access to other CT boards, such as speech recognition and SS7

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The table below summarizes the features of each QuadSpan model.

QuadSpan Model	Telephony Signaling Channels	Voice Processing Channels	Network Interface	Form Factor	Resource Bus	OS Support
DM/V1200-4E1	120 ports	120 ports	Four E-1 / ISDN PRI	PCI, CompactPCI, VME	SCbus, CT Bus	Windows NT Solaris UnixWare
DM/V960-4T1	96 ports	96 ports	Four T-1 / ISDN PRI	PCI, CompactPCI, VME	SCbus, CT Bus	Windows NT Solaris UnixWare
DM/V600-4E1	120 ports	60 ports	Four E-1 / ISDN PRI	PCI, CompactPCI	SCbus, CT Bus	Windows NT Solaris UnixWare
DM/V480-4T1	96 ports	48 ports	Four T-1 / ISDN PRI	PCI, CompactPCI	SCbus, CT Bus	Windows NT Solaris UnixWare

OVERVIEW

These QuadSpan Voice products provide the following functionality in *real time* on all 96 (T-1) or 120 (E-1) channels:

- connect to 96 (T-1) or 120 (E-1) telephone channels via DSX-1 termination
- automatically answer calls using virtually any international telephony signaling protocol
- detect DTMF (touchtone)
- digitally compress and record voice signals
- play voice messages to a caller
- place outbound calls and automatically track call progress

Downloadable Firmware

The QuadSpan hardware consists of a baseboard with a RISC processor and four DS-1 digital network interfaces (different assemblies are used for T-1 and E-1). An array of digital signal processors (DSP) resides on a low-profile daughterboard. Telephony signaling protocols and voice processing features are downloaded to the board on power up, and reside as firmware on the various on-board processors. This downloadable firmware approach, combined with hardware daughterboards, enables easy feature upgrade and expansion. Individual firmware components, such as a network interface protocol, or a voice recording function, are referred to as *resources*.

Network Interface

The T-1 versions of QuadSpan supports all T-1 robbed-bit signaling protocols and are fully compatible with all resource devices that use or can be set to use 1.544-MHz clocking and μ -law pulse code modulation (PCM). All basic channel associated signaling functions such as pulse dialing and detection, winking, and hookflash are supported.

The E-1 versions of QuadSpan supports all CEPT channel associated signaling (CAS) protocols and are fully compatible with interface devices that use or can be set to use 2.048 MHz clocking and A-law PCM (ITU-T Recommendation G.703/704/711). Check the Dialogic *Worldview*[™] Web site for an up-to-date list of the growing number of R1 and R2/MF compelled protocols supported.

The QuadSpan Series also supports ISDN Primary Rate Interface (PRI) Access. PRI enables applications to take advantage of the speed, power, and flexibility of ISDN. Dialogic maintains an extensive number of approvals in international markets. Check the Dialogic *Worldview* Web site (<http://www.dialogic.com>) for an up-to-date list.

GlobalCall

GlobalCall is the Dialogic technology for unified call control on Dialogic SpringWare[™] and DM3 architectures. GlobalCall is software and firmware that enables Dialogic hardware to connect to the public switched telephone network (PSTN). GlobalCall is a comprehensive Dialogic resource that offers seamless access to a variety of network interfaces, and provides portability around the world. GlobalCall provides developers with international protocols for multiple network transports, including E-1, T-1, ISDN, and IP. It shields developers from the many variations of call control that exist around the world and allows them to concentrate on their applications and leverage them into new geographic markets.

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Voice Processing

Voice processing features, downloaded to the on-board DSPs at power up, allow the QuadSpan to play and record voice messages to and from callers through the digital network interface. Messages can be stored using G.711 μ -law or A-law PCM, at a rate of 64 Kb/s, as is used by the public switched telephone network; or to reduce storage requirements, voice coding algorithms can compress recordings to 24 or 32 Kb/s using adaptive differential pulse code modulation (ADPCM). Other coders, such as G.726 ADPCM, and 11KHz WAV are expected in future releases.

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 needed. 12 MB of on-board RAM can be used for local caching of voice prompts for optimal overall system performance.

Automatic gain control (AGC) is provided to automatically adjust the signal level of incoming calls for recording at normal levels, compensating for adverse line conditions, distance, and other factors. Playback volume can also be dynamically adjusted over a 40 dB range using DTMF input or directly from the application.

Reliable DTMF detection is provided because voice processing applications commonly require callers to control recording and playing using DTMF input. Local echo cancellation techniques are used to improve DTMF cut-through and talk off/play off suppression over a wide variety of telephone line conditions.

The voice player and recorder resources are linked with the DTMF detection resources using run time control (RTC) messages. This allows a play or record process to be quickly initiated or terminated using DTMF input from the caller. The RTC function off-loads the host application from involvement in every interaction, thereby enabling voice processing applications to scale to hundreds of ports per system.

Tone Signaling

In addition to the DTMF signaling, commonly used for voice processing, the QuadSpan also contains a robust set of features used for network tone signaling and control.

PerfectCall™ call progress analysis accurately monitors outbound calls, detects when calls are answered and distinguishes

- line ringing but not being answered
- line busy
- problem completing call (such as operator intercept)
- call answered by a human or answering machine
- call answered by a fax machine or modem

PerfectCall is intelligently tolerant of the wide variation in call progress signaling tones found in central offices and PBXs around the globe and offers accurate performance right out of the box. Unique, patented DSP-based algorithms are used to accurately discriminate human speech from recorded human voice and from network noise.

Form Factors

The QuadSpan is available in PCI, cPCI, and VME form factors enabling systems developers to choose the platform that is optimal for their intended deployment. All three platforms require only a single slot in the bus backplane. cPCI offers the advantage of sharing the same software and CPU availability as PCI. Developers can build one application, and deploy their system on PCI or cPCI, depending on the needs of the end user.

cPCI and VME both use the Eurocard 6U format and are especially suited for large-scale systems that require a high degree of availability and reliability. The combination of live-insertion hardware technology, flexible device drivers, callable loaders, and configuration utilities provides unsurpassed provisioning and main-

FEATURES AND BENEFITS

- Software development kits for Windows NT®, Solaris® and UnixWare® yield faster time to market

APPLICATIONS

- Enhanced services
- Intelligent peripherals
- CO voice messaging
- Prepaid/debit card
- International callback
- Network call center
- Next-generation telco switches

tenance support. Systems can be provisioned initially for minimum service levels so that resources can be added later. Operators can change the configuration at any time without disturbing connected calls or system operation. System slots can even be reconfigured for a different product as call patterns or service offerings change.

After a newly inserted board is recognized and identified, detailed diagnostic tests verify board functionality to prevent a defective or incorrect board from being installed in a functioning system. After downloading the specified feature modules, the device driver is notified to include the new capabilities in the active device tables. Applications can be notified of new resources upon completion of the download process or by periodic application query.

The QuadSpan Voice Series supports Simple Network Management Protocol (SNMP), which provides remote fault and performance monitoring and basic control of Dialogic components (hardware/firmware/software). SNMP makes it possible to continuously monitor idle channels to detect incipient failures, preventing active calls from being switched to failed resources. Suspect boards can be taken off-line without interrupting normal call flow.

SCbus and CT Bus Connectivity

The SCbus and CT Bus are time division multiplex buses for carrying isochronous data between multiple computer telephony boards within a single system. Both are implemented as a ribbon cable “mezzanine” bus for PCI systems, and connect directly into the computer backplane for cPCI and VME systems.

The SCbus and CT Bus provide the ability to switch calls from a network interface port on one board to a network interface port or mediastream processing resource located on another board. Having a separate bus for this purpose permits low-latency communications that is independent of the computer’s I/O and memory buses.

The SCbus provides up to 2,048 bi-directional, DS0 (64 Kb/s) time slots. The SCbus is the industry standard adopted by ANSI/VITA for VME. The CT Bus is a newly emerged standard, adopted by the ECTF as H.100 for PCI systems and H.110 for cPCI systems. The CT Bus interoperates with the SCbus while providing double the capacity. Both buses provide distributed switching for simplifying resource management and control. Separate clock master and synchronization fallback provide reliable system operation.

DM3 Architecture

The QuadSpan is based on Dialogic DM3 Mediastream architecture. DM3 delivers major improvements in performance and flexibility to meet system developers’ most stringent requirements. The key is a universal hardware platform with a selection of open, firmware-based technologies and improved embedded resource and application development tools. With versions for the most demanding computing platforms — like PCI, cPCI, and VME — and advances based on software and hardware standards, DM3 architecture makes it possible to build high-density, reliable, network-grade solutions with unmatched price and performance.

DM3 insulates firmware resources from the underlying hardware, and host device drivers are resource independent. This reduces development cycles and speeds time to market.

The DM3 modular hardware architecture enables DSPs, network interface modules, and memory to be added or replaced as required. Developers can add features to extend the life of the platform, or remove resources to lower cost for niche applications. In this way, developers can protect their investment in hardware and software.

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Software Support

The QuadSpan Series will be supported with Software Development Kits for Windows NT, Solaris UltraSparc, and UnixWare operating system environments. These kits contain a complete set of tools for developing complex multi-channel applications.

APPLICATIONS AND CONFIGURATIONS

Use the Dialogic QuadSpan products to develop sophisticated, multifunction CT systems that include voice processing, speech recognition, fax processing, and IP telephony. Since QuadSpan boards share a common hardware and firmware architecture with other Dialogic DM3 boards, you can add features or expand the number of ports into the hundreds while protecting your investment in hardware and application code.

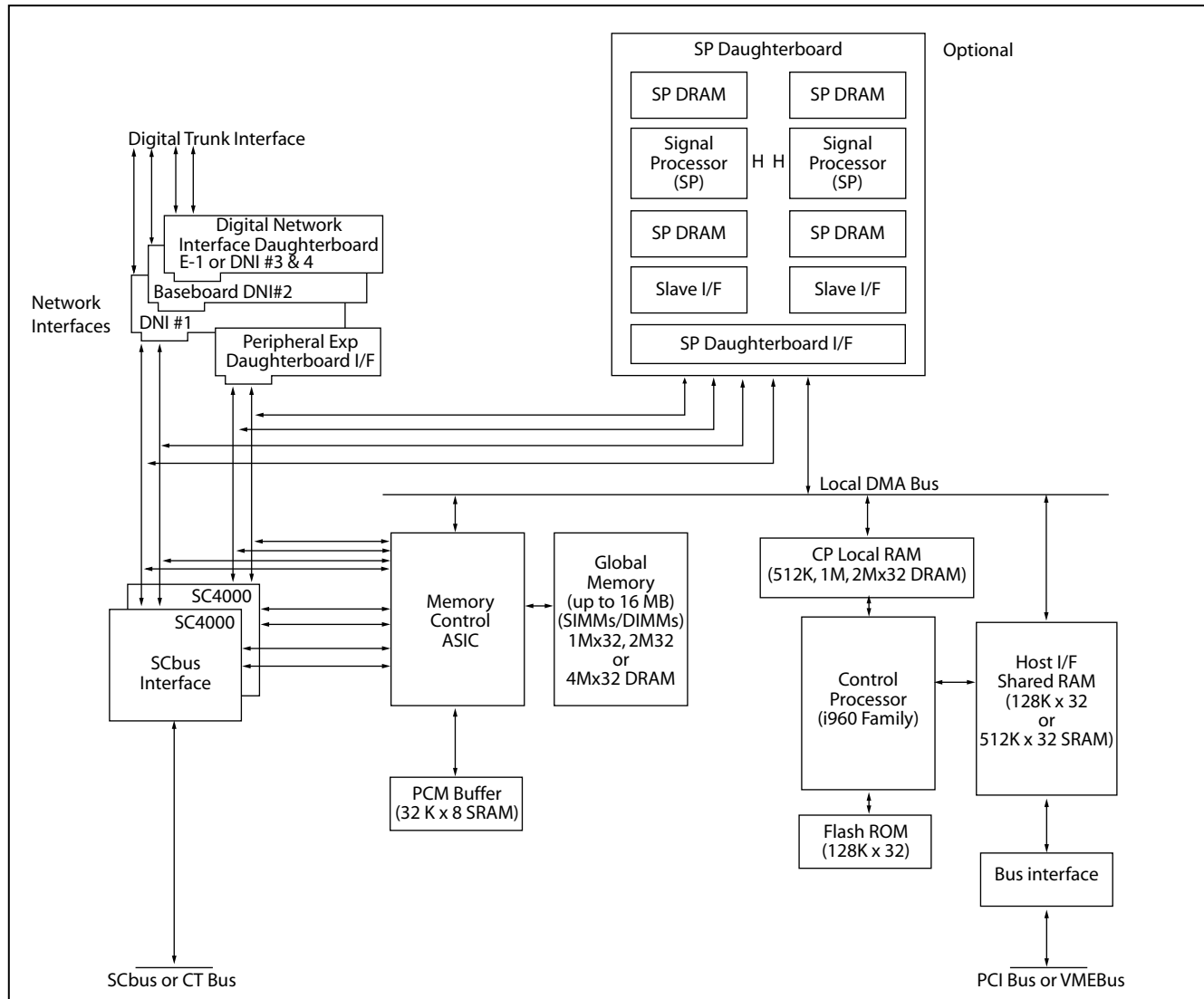
QuadSpan products occupy a single computer backplane slot and multiple QuadSpan products can be installed in a single computer. The maximum number of lines supported depends on the

- application
- intended call traffic
- host computer CPU
- power available from the chassis power supply.

QuadSpan voice products can operate in either terminate or drop-and-insert configurations. In a terminate configuration, they handle the call processing of the digital audio and telephony signaling. Additional system resources can access calls via the SCbus/CT Bus. This configuration is ideal for voice messaging, unified messaging, and interactive voice response application.

In a drop-and-insert configuration, the boards are connected via the SCbus/CT bus and can continuously pass all T-1/ E-1 time slots through to each other. This configuration can switch call traffic between separate T-1 or E-1 lines, or it can be placed in-line between a T-1/E-1 public network trunk and a digital switch. Calls on individual channels can either terminate at a call processing resource on a board, or “flow through” transparently from one QuadSpan product to the other. This configuration is ideal for call center, prepaid calling card, international call-back, and telecom resale applications. ■

FUNCTIONAL DESCRIPTION



The QuadSpan products above are based on the DM3 Mediastream architecture* (see the block diagram above). The architecture consists of a set of core specifications and firmware modules that are implemented on boards with various processors including

- RISC Processor for centralized control
- Digital Signal Processor(s) (DSP) for mediastream processing
- TDM Bus interface (SCbus/CT Bus)

- Digital Telephony Network Interfaces
 - Host computer bus interface

QuadSpan products support up to 96 (T-1) or 120 (E-1) channels of voice processing via a bank of DSPs and four E-1 or T-1 digital trunk interfaces (DTI) circuits. The DTI circuits contain signaling services (ISDN, CAS, and CCS) plus any alarm handling and line maintenance services required by the installed networks. Each DTI includes

software switchable clock circuits that can be set to

- loop mode (transmit clocking is slaved to the external network)
- independent mode (transmit clocking is derived from an on-board oscillator)
- expansion or system mode (transmit clocking is slaved to the SCbus/CT Bus; receive clocking is always slaved to the trunk interface)

* For up-to-date information about DM3 architecture, visit the Dialogic *WorldView* Web site at <http://www.dialogic.com>.

The Control Processor is a general purpose Intel i960™ RISC microprocessor. It is responsible for the initialization, configuration, and control of the various elements that make up the QuadSpan products. It controls the TDM bus interface, as well as the signaling protocols for the DTIs installed on the platform.

The QuadSpan Series of products support various DSP configurations for voice processing and call progress analysis capabilities. These features are provided by daughterboard configuration using up to six Motorola 5630x DSPs per card. The DM3 architecture will permit DSPs from other manufacturers to be supported in the future, extending the life of the product.

The DSP processes the digitized voice data using downloaded resource firmware. Each DSP can perform the following signal analysis and operations:

For incoming data

- automatic gain control, which compensates for variations in the level of the incoming audio signal
- ADPCM or PCM algorithms, that compress the digitized voice and

save disk storage space

- tone detection of DTMF, MF, or application-defined single or dual tones
- silence detection 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 pitch of playback upon application or user request
- generates tones — DTMF, MF, or any application-defined general-purpose tone
- performs outbound dialing
- monitors call progress functions, including
 - line busy
 - operator intercept
 - ring
 - no answer
 - answered; the DSP detects whether the answering party is a person, answering machine, a fax machine, or modem

When recording speech, the DSP can use different digitizing rates from 24 to 64 Kb/s selectable by the applica-

tion 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. DSP-processed speech is transmitted by the control processor to the host for disk storage. When playing back a stored file, the processor retrieves the voice information from the host CPU 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 network interface or TDM bus.

Shared RAM on the QuadSpan boards enables communication between the host system and the i960 control processor. A bank of global memory is also provided to facilitate communications between the control processor and the various DSPs. In addition to providing a data pathway between processors, the global memory can also serve as a repository for data, which is to be shared among processors, or which may not be storable within local memory associated with the processor. ■

■ DM3 Hardware Technical Specifications*

QUADSPAN CONFIGURATION:

Digital interfaces	4 T-1 or 4 E-1
Max. boards per system	Application, call traffic, and CPU dependent
Control processor	Intel i960CF at 33 MHz, 66 MIPS
Control processor memory	Up to 8 MB local to control processor
Digital signal processors	Motorola 5630x, 1 K word program cache
	DM/V1200: Six DSPs @ 80 MIPS each (100 MIPS in near future)
	DM/V960: Six DSPs @ 80 MIPS each (100 MIPS in near future)
	DM/V600: Three DSPs @ 80 MIPS each (100 MIPS in near future)
	DM/V480: Three DSPs @ 80 MIPS each (100 MIPS in near future)
DSP memory	256 K word DRAM local to each DSP
	128 K word SRAM local to each DSP
Baseboard global memory	32-bit wide DRAM accessible to all signal processors, control processor, and host
	DM/V1200 & DM/V960: 16 MB
	DM/V600 & DM/V480: 8 MB, upgradable

PCI PLATFORM:

Form factor	PCI long card, single-slot width
Host interface memory	512 KB
Bus compatibility	Rev 2.1 of PCI Bus Specification
Bus mode	Target and DMA master mode operation
Computer telephony bus	ECTF H.100 compliant CT Bus, offering: <ul style="list-style-type: none"> • On-board switching access to 4096 bi-directional 64 kb/s DS0 time slots • SCbus Interoperability through Dialogic provided adapter • 68-pin ribbon cable connector
	Some early versions of QuadSpan offer 1024 to 2048 time slot SCbus, with H.100 style connector
Network connectors	Four RJ-48C on rear bracket

CompactPCI PLATFORM:

Form factor	6U Eurocard form factor, single-slot width
Host interface memory	512 KB
Bus compatibility	Rev 2.1 of PCI Bus Specification
Bus mode	Target and DMA master mode operation
Computer telephony bus	ECTF H.110 Compliant CT Bus, offering: <ul style="list-style-type: none"> • On-board switching access to 4096 bi-directional 64 kb/s DS0 time slots
Network connectors	Provided through rear I/O transition modules BNC for 75 Ohms or RJ-48C for 100 and 120 Ohm lines

VME PLATFORM:

Form factor	6U Eurocard form factor, single slot width
Host interface memory	128 Kbytes
P1 bus compatibility	VME64 bus (ANSI/VITA 1-1994) VITA P1.1 VME64 bus extensions VITA High Availability VME Extensions
P2 (computer telephony bus)	ANSI/VITA 6-1994 SCSA VITA 6.1-1995 SCSA Extensions
VMEbus compliance	SA32, SD32, BLT Base addresses A24: 080000h to 0A0000h A32: 00080000h to 000A0000h Base address set by geographic addressing or manual jumper
Interrupt level	IRQ 1-7, software selectable. One IRQ line may be shared by all boards.
Network connectors	Provided through rear I/O transition modules BNC for 75 Ohms or RJ-48C for 100 and 120 Ohm lines

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■ DM3 Hardware Technical Specifications* (cont.)

TELEPHONE INTERFACE:

(T1):	
Clock rate	1.544 Mb/s \pm 32 ppm
Level	3.0 V (nominal)
Pulse width	325.85 ns (nominal)
Line impedance	100 Ohms, balanced
Other electrical characteristics	Complies with AT&T TR62411 and ANSI 403-1989
Framing	SF (D3/D4), ESF
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
Loopback	Software selectable local digital loopback and local analog loopback
(E1):	
Clock rate	2.048 Mb/s \pm 32 ppm
Level	2.37 V (nominal) for 75 Ohms or 3.0 V (nominal) for 120 Ohm lines
Pulse width	244 ns (nominal)
Line impedance	75 Ohms, unbalanced or 120 Ohms, balanced
Other electrical characteristics	Complies with ITU-T Rec. G.703
Framing	ITU-T Rec. G.704-1988 with CRC-4
Line coding	HDB3
Clock and data recovery	Complies with ITU-T Rec. G.823-1988
Jitter tolerance	Complies with ITU-T Rec. G.823, G.737, G.739, G.742-1988
Loopback	Software selectable local digital loopback and local analog loopback

REGULATORY CERTIFICATIONS:

See the Dialogic *WorldView* Web site (<http://www.dialogic.com>) for the current certification status

POWER REQUIREMENTS:

Configuration	+5 VDC	+12 VDC	-12VDC	+ 3.3 VDC
DM/V1200-4E1	17.4 W	0.6 W	0.6 W	0.0 W
DM/V960-4T1	17.4 W	0.6 W	0.6 W	0.0 W
DM/V600-4E1	17.4 W	0.6 W	0.6 W	0.0 W
DM/V480-4T1	17.4 W	0.6 W	0.6 W	0.0 W

COOLING REQUIREMENTS:

Operating temperature	0°C to +50°C
Storage temperature	-20°C to +70°C
Humidity	8% to 80% noncondensing
WARRANTY:	3 Years Standard

■ Resource Technical Specifications*

AUDIO SIGNAL:

Usable receive range	-40 dBm0 to 0 dBm0 nominal, configurable by parameter**
Automatic gain control	Application can enable/disable. Above -21 dBm results in full scale recording, configurable by parameter.**
Silence detection	-40 dBm nominal, software adjustable**
Transmit level (weighted average)	-12.5 dBm 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
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
64 Kb/s	G.711 PCM (μ-law for T-1 and A-law for E-1)
Digitization selection	Selectable by application on function call-by-call basis

DTMF TONE DETECTION™:

DTMF digits	0 to 9, *, #, A, B, C, D per Bellcore LSSGR Sec 6
Dynamic range	(T-1) -36 dBm to +3 dBm per tone, configurable by parameter** (E-1) -39 dBm to +0 dBm per tone, configurable by parameter**
Minimum tone duration	32 ms; can be increased with software configuration
Interdigit timing	Detects like digits with a >45 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 ITU-T Q.23 recommendations**
Noise tolerance	Meets Bellcore LSSGR Sec 6 and EIA 464 requirements for Gaussian, impulse, and power line noise tolerance
Cut through	(T-1) Local echo cancellation permits 100% detection with a >4.5 dB return loss line (E-1) Digital trunks use separate transmit and receive paths to network. Performance dependent on far end handset's match to local analog loop.
Talk off	Detects less than 10 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 300-3500 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) Default set at -36 dBm to +3 dBm per tone, programmable (E-1) Default set at -39 dBm to +0 dBm per tone, programmable

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	(T-1) -43 dBm to -3 dBm per tone nominal, programmable (E-1) -40 dBm to 0 dBm per tone nominal, programmable

■ Resource Technical Specifications* (cont.)

(T-1) 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 dBm to +3 dBm per tone
Acceptable twist	6 dB
Transmit frequency variation	Less than ± 1 Hz
(E-1) MF SIGNALING:	R2
MF digits	All 15 forward and backward signal tones per ITU-T Q.441
Transmit level	-8 dBm0 per tone nominal, per ITU-T Q.454; programmable
Signaling mechanism	Supports the R2 compelled signaling cycle and non-compelled pulse requirements per ITU-T Q.457 and Q.442
Dynamic range for detection	-35 dBm to -5 dBm per tone
Acceptable twist	7dB
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 ITU-T 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 ITU-T 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	>85% accurate based on application and environment
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 (when not part of regulatory approval).

TONE DIALING:

DTMF digits	0 to 9, *, #, A, B, C, D per Bellcore LSSGR Sec 6, TR-NWT-000506, ITU-T Q.23
Frequency Variation	Less than ± 1 Hz
Rate	10 digits/s, configurable by parameter**
Level	(T-1) -4.0 dBm per tone, nominal, configurable by parameter** (E-1) -7.0 dBm per tone, nominal, country-specific**

PULSE DIALING:

10 digits	0 to 9
Pulsing rate	10 pulses/s, nominal, configurable by parameter**
Break ratio	60% nominal, configurable by parameter**

* 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.