VMEDIA™ V/S24T1™

24-CHANNEL VOICE/TONE PROCESSING BOARD with integrated T-1 ISDN NETWORK interface for VMEbus-based scalable systems

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Features & Benefits

- Optimal channel-per-slot granularity: 1.544 Mb/s digital service access (T-1) plus 24 channels of voice processing in a single VME slot enables system integrators, call processing software developers, large service bureaus, and PTTs to lower the cost of large platforms (IP, SN) by incorporating multiple boards per VME chassis and by eliminating cabling between boards, easing configuration, simplifying maintenance, and minimizing installation time
- Supports ISDN Primary Rate with extensive approvals (see *ISDN PRI Support* section)
- Optimized firmware and drivers for the VME environment give you better performance
- The V/S24T1 can be ordered with the I/O out the front or out the rear of the board
- Access to the SCSA SCbus through the P2/J2 VME backplane connector (ANSI/VITA 6-1994) lets you build higher density systems without interconnecting cables between boards
- SpringWare™ downloadable signal and call processing firmware gives you easy feature enhancement and field-proven performance based on over two million installed ports worldwide
- PerfectDigit™ DTMF (touchtone) provides reliable detection during voice playback and allows callers to “type-ahead” through menus
- Three independent Motorola DSP56002 digital signal processors, (40 MHz, each with private,

high-speed SRAM), let you execute high-performance SpringWare signal processing algorithms

- Two fast 386DE microprocessors off-load call processing tasks from the host CPU and host operating system
- SCSA geographic addressing simplifies installation by eliminating confusing DIP switch and jumper settings
- C language application program interfaces (APIs) compatible with several UNIX® implementations let you choose the open UNIX environment best suited to your application. Extend ISA/PCI based applications to the powerful VME environment and get your applications to market on a new VME platform with minimum effort.
- Configure multiple boards in a single VMEbus for easy and cost-effective system expansion while choosing the best computing platform for your needs

Applications

- Intelligent Network (IN) applications on scalable IPs
- Audiotex
- Automated directory services
- Cable television services
- Government telecommunication networks and services
- Interactive voice response
- Notification systems
- On-line database entry/query
- Operator services
- Service bureau applications
- Telemarketing, 700, 800, and 900 services

- Voice mail
- Voice messaging and paging
- Voice/audio response systems
- Wireless to wireline applications for PTTs
- Paging applications

The V/S24T1 digital T-1 voice board provides T-1 (1.544 Mb/s) service termination and call processing for up to 24 voice or data channels as well as ISDN Primary Rate Interface (PRI) functionality in a single VME slot. The V/S24T1 can be ordered with the I/O out the front or out the rear of the board. A unique dual-processor architecture comprising powerful digital signal processors (DSPs) and general-purpose microprocessors provides the power to handle all telephony signaling, DTMF, MF, and audio/voice signal processing tasks. The V/S24T1 board supports robbed bit signaling.

The V/S24T1 board, as a member of the VMEDIA family, is based on the Signal Computing System Architecture™ (SCSA™). SCSA is an open architecture that lets you use products from multiple vendors to build a unified computer telephony solution. SCSA features include distributed switching, clock fallback, logical addressing, and resource management.

The ANSI-approved SCSA SCbus™, available on the VME backplane, provides up to 2048 bidirectional time slots. SCSA resource switching makes building large, multiservice systems a reality. SCbus bandwidth in the VME chassis can be set to one of the following three rates at download time:

Data Bus Rate in MHz	Bidirectional Time Slots	Throughput in Mb/s
8.192	2048	131.072
4.096	1024	65.536
2.048	512	32.768

Downloaded SpringWare™ firmware algorithms, executed by the on-board DSPs, provide variable voice coding at 24 and 32 Kbps ADPCM (Adaptive Differential Pulse Code Modulation) and 48 and 64 Kbps µ-law or A-law PCM (Pulse Code Modulation). 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. SpringWare also provides reliable DTMF detection, DTMF cut-through, and talk off/play off suppression over a wide variety of telephone line conditions.

The V/S24T1 voice/tone processing board with T-1/ISDN network interface provides the following functionality *real time* on all 24 channels:

- connects to 24 telephone channels via DSX-1 termination
- automatically answers calls
- detects touchtones
- digitizes, compresses, and records voice signals
- plays voice messages to a caller
- places outbound calls and automatically tracks call progress

Configurations

You can use the V/S24T1 board to develop sophisticated, multifunction computer telephony systems that include voice processing, speech recognition, text-to-speech, fax, and SS7. Since the V/S24T1 board shares a common hardware and firmware architecture with other Dialogic SCbus-based boards, it offers maximum flexibility and scalability. You can add features or expand your system while protecting investment in hardware and application code. With only minimum modifications, you can port applications from lower density ISA/PCI bus-based platforms.

The V/S24T1 board occupies a single VME slot and all V/S24T1 boards can share the same interrupt level. The maximum number of lines that can be supported depends on the application, the amount of disk I/O required, and the host computer CPU.

The V/S24T1 board can operate in either terminate or drop and insert configurations. In a terminate configuration, the V/S24T1 board handles the call processing of the digital audio and telephony signaling. Additional system resources can access calls via the VME backplane-resident SCbus.

In a drop and insert configuration, two V/S24T1 boards are connected via the SCbus and can continuously pass all T-1 time slots through to each other. This configuration can join two separate T-1 lines, or it can be placed in-line between a T-1 line and a switch. Calls on individual T-1 channels can either terminate at a call processing resource on a V/S24T1 board, or “flow through” transparently from one V/S24T1 board to the other.

The V/S24T1 board supports D3/D4 trunks or ESF trunks when using ISDN signaling.

ISDN PRI Support

The Dialogic ISDN PRI Access firmware is a feature enhancement to the VMEDIA product series. The Dialogic PRI firmware is approved for use with many popular switches and in major telecommunications markets worldwide.

Features and benefits of ISDN PRI include:

- ISDN Primary Rate connectivity to Dialogic computer telephony systems
- Downloadable firmware yields complete 24-port voice processing plus PRI D-channel access in a single slot
- Dialed Number Identification Service (DNIS) enables application to automatically identify the purpose of the incoming call
- Automatic Number Identification (ANI) enables application to identify the calling party
- ANI-on-Demand feature saves money by selectively requesting ANI information only when needed
- ISDN offers inherent benefits to call center applications with its fast call setup and fast retrieval of DNIS and ANI information on inbound calls
- Vari-A-Bill® enables service bureaus to change the billing rate of a 900 call, on the fly
- Call-By-Call Service Selection allows an application to select the most efficient bearer channel service on a call-by-call basis
- Subaddressing allows direct connection to individual extensions or devices sharing the same phone number or as a proprietary messaging mechanism
- Powerful and universal software API simplifies access for developers who are unfamiliar with ISDN, yet enables sophisticated control of features
- Multinational approvals with all popular switches
- User-to-User Information allows an application to send proprietary messages to remote systems during call establishment
- Facility, Notify, and optional information elements allow applications to work with network-

specific supplementary services

Software Support

Dialogic VME products are supported by off-the-shelf Dialogic System Software and SDKs. Contact your Dialogic sales engineer for additional information on currently available operating system software support and porting options. The API library and software documentation is consistent with the ISA/PCI UNIX API library.

Functional Descripton

The V/S24T1 board emulates a T-1 channel bank and processes the digital on-hook/off-hook signaling information and digital voice signals from the telephone network. Digital T-1 signals from the telephone network enter the V/S24T1 board via a T1XC line interface. The line interface contains software switchable clock circuits which can be set to

- loop (clocking is slaved to the external network)
- independent (clocking is derived from an on-board oscillator)
- expansion (clocking is slaved to the SCbus)

The incoming T-1 bit stream is directed to the T-1 interface SC2000 chip that acts as the traffic coordinator to route the digital data received for each channel to the SCbus. This serial bit stream contains the digitized voice data and the signaling information for the incoming call.

The SC2000 chips incorporate matrix switching capabilities. Under control of the on-board 386DE control processor, the SC2000 chip can connect either a call being processed or an available external resource on any SCbus time slot. This enables the application to reroute calls to any other resource, such as facsimile, speech recognition, text-to-speech, or video.

Each of the DSP resources receive digital data via a dedicated SC2000 chip. The DSP processes the digitized voice data using downloaded SpringWare firmware. Each DSP can perform the following signal analysis and operations:

On the incoming data

- automatic gain control, which compensates for variations in the level of the incoming audio signal

- ADPCM or PCM algorithms, which 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

The V/S24T1 T1XC line interface extracts or inserts telephony signaling information, which an on-board control processor handles. The DSPs process the digitized voice and tone data.

When recording speech, the DSP can use different digitizing rates from 24 to 64 Kbps selectable 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. 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 DSP dedicated SC2000 chip and network interface.

An SC2000 chip can bundle time slots to carry high bandwidth data and can broadcast to multiple resources over the SCbus.

A pair of High-level Data Link Controllers (HDLC) control access to the SCbus message bus. This message bus is a separate SCbus channel that carries messages and control information such as out-of-band signaling and event control among SCSA devices. With out-of-band signaling, boards can respond quickly to interboard messages and control information.

The on-board control processors interpret and execute commands from the host VME CPU. These processors handle real-time events, manage data flow to the host CPU to provide faster system response time, reduce VME host processing demands, process DTMF and telephony signaling before passing them to the application, and free the DSPs to perform signal processing. Communications between a 386DE processor and the host VME CPU is via shared RAM, which acts as an input/output buffer and thus increases the efficiency of disk file transfers. All operations are interrupt-driven to meet the demands of real-time systems. When the system is initialized, SpringWare firmware, which controls all board operations, is downloaded from the host CPU. This downloadable firmware gives the board all of its intelligence and enables future enhancements and upgrades.

V/S24T1 TECHNICAL SPECIFICATIONS*

Number of ports	24
Max. boards/system	Application- and CPU-dependent
Digital network interface	On-board DSX-1 interface
Resource sharing bus	SCbus - up to 2048 bidirectional time slots
Control microprocessor	Two 386DE @ 25 MHz, 0 wait state
Digital signal processors	Three Motorola DSP56002 @ 40 MHz, each with 32 K word private, 0 wait state SRAM
HOST INTERFACE:	
Bus compatibility	VME64 (ANSI/VITA 1-1994)
VMEbus compliance	SA32, D16
Shared memory	512 KB
Base addresses	A24: 080000h to 0A0000h A32: 00080000h to 000A0000h Base address set by SCbus interface or manual jumper
Interrupt level	IRQ 1-7, software selectable. One IRQ line may be shared by all boards.
TELEPHONE INTERFACE:	
Clock rate	1.544 Mb/s ±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 62411 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
Connectors	RJ-48C or P2 (order option)
Loopback	Supports remote analog loopback, switch selectable local analog loopback, and software selectable local digital loopback

PHYSICAL REQUIREMENTS:

+5 VDC	3.0 A, typical; (measured)
+12 VDC	20 mA max.
-12 VDC	20 mA max.
Operating temperature	0° C to +50° C
Storage temperature	-20° C to +70° C
Humidity	8% to 80% noncondensing
Form factor	VME 6 U × 160 mm

REGULATORY CERTIFICATIONS:

United States	FCC part 68 ID# EBZUSA-22560-XD-N USOC: 6.0P 04DU9-BN/DN/1KN/1SN/1ZN UL: 1950 FCC Part 15 Class A
Canada	DOC: CS-03 885 6729 A CA81A D1/D1E/D2/D2E/D4/D4E/D4A/D4B/D4C ULC: CSA 950
Warranty	3 years standard

SPRINGWARE 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 dBm0 results in full scale recording, configurable by parameter**
Silence detection	-40 dBm0 nominal, software adjustable**
Transmit level (weighted average)	-12.5 dBm0 nominal, configurable by parameter**
Transmit volume control	40 dB adjustment range, with application definable increments and legal limit cap

Frequency response

24 Kb/s	300 Hz to 2600 Hz ±3 dB

32 Kb/s	300 Hz to 3400 Hz ±3 dB
48 Kb/s	300 Hz to 2600 Hz ±3 dB
64 Kb/s	300 Hz to 3400 Hz ±3 dB

AUDIO DIGITIZING:

24 Kb/s	OKI ADPCM @ 6 kHz sampling
32 Kb/s	OKI ADPCM @ 8 kHz sampling
48 Kb/s	Raw PCM @ 6 kHz sampling
64 Kb/s	Raw 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 dBm0 to +3 dBm0 per tone, configurable by parameter**
Minimum tone duration	50 ms, can be increased or decreased with software configuration
Interdigit timing	Detects like digits with a >50 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	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 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	Less than 5 Hz. - Note: Certain limitations exist for dual tones closer than 125 Hz apart.
Timing	Programmable cadence qualifier, in 10 ms increments
Dynamic range	Programmable, default set at -39 dBm0 to +0 dBm0 per tone

GLOBAL TONE GENERATION™:

Tone type	Generate single or dual tones
Frequency range	Programmable within 200 to 4000 Hz
Frequency resolution	1 Hz
Duration	10 msec increments
Amplitude	-40 dBm0 to 0 dBm0 per tone nominal, programmable

MF SIGNALING:

MF digits	0 to 9, KP, ST, ST1, ST2, ST3 per Bellcore LSSGR Sec 6, TR-NWT-000506 and ITU-T Q.321
Transmit level	Complies with Bellcore LSSGR Sec 6, TR-NWT-00506
Signaling mechanism	Complies with Bellcore LSSGR Sec 6, TR-NWT-00506
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 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 ringback tones used in 96 countries as specified by ITU-T Rec E., Suppl #2. Uses both frequency and cadence detection.
Positive Voice Detection™ accuracy	>99% based on tests on a database of real world calls
Positive Answering	
Machine Detection™	Detects recorded voice in as little as 1/10th of a second
Fax/modem detection	Preprogrammed
Intercept detection	Detects entire sequence of the North American tri-tone. Other intercept tone sequences can be programmed.

Dial tone detection before dialing	Application enable/disable; Supports up to three different user-definable dial tones; Programmable dial tone drop out debouncing.
TONE DIALING:	
DTMF digits	0 to 9, *, #, A, B, C, D per Bellcore LSSGR Sec 6, TR-NWT-000506
Frequency variation	Less than ±1 Hz
Rate	5 digits/s default, configurable by parameter**
Level	-7.5 dBm0 per tone, nominal, configurable by parameter**
PULSE DIALING:	
10 digits	0 to 9
Pulsing rate	10 pulses/s, nominal, configurable by parameter**
Break ratio	60% nominal, configurable by parameter**
ANALOG DISPLAY SERVICES INTERFACE (ADSI):	
Interface	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.

Host System Requirements

For a complete and up-to-date listing of partners providing system requirements support, contact your Dialogic Sales Engineer.

Additional Components

To install the V/S24T1 board in a VME chassis you need to make certain that the VME backplane is capable of carrying SCbus traffic on the P2/J2 connector. This is usually accomplished by backplane add-on options or via the ANSI-approved SCbus/VME backplane. To install the V/S24T1 board with rear I/O, additional I/O components will usually be required to bring the I/O from the rear of the backplane to the rear of the chassis. See the *VMEDIA VME Accessories* data sheet for detailed information. Contact Dialogic if you require additional assistance concerning the V/S24T1 board.

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