SPIRENT Communications Inspired Innovation

Abacus™ 50 T1 and E1
T1 AND E1 TRAFFIC GENERATOR TEST SYSTEM

The Abacus 50 T1/E1 test system provides PCM circuits to emulate a telephone exchange (central office) or a terminal in a cost-efficient standalone platform designed for laboratory testing as well as distributed testing on the network.

VoIP Convergence

- A single user interface with synchronized Abacus 50 Analog and Abacus 5000 Analog, TDM and VoIP measurements
- Measure one-way delay between TDM and VoIP devices
- Verify functionality of media and voice gateways
- Check dial-up connectivity of voice traffic
- Assess voice quality
- Generate calls to an IP network

Network Equipment Manufacturers (chips, IP-PBX, gateway, MSs and SSs)

- Characterize system before trial
- Validate system scalability
- Identify capacity limits
- Measure call performance
- Automate regression testing

Service Providers (NSPs, SPs, ITSPs and Enterprises)

- Facilitate vendor selection
- Identify performance ceilings
- Enable accurate capacity planning
- Provide end-to-end service assurance testing
- Network planning and deployment analysis

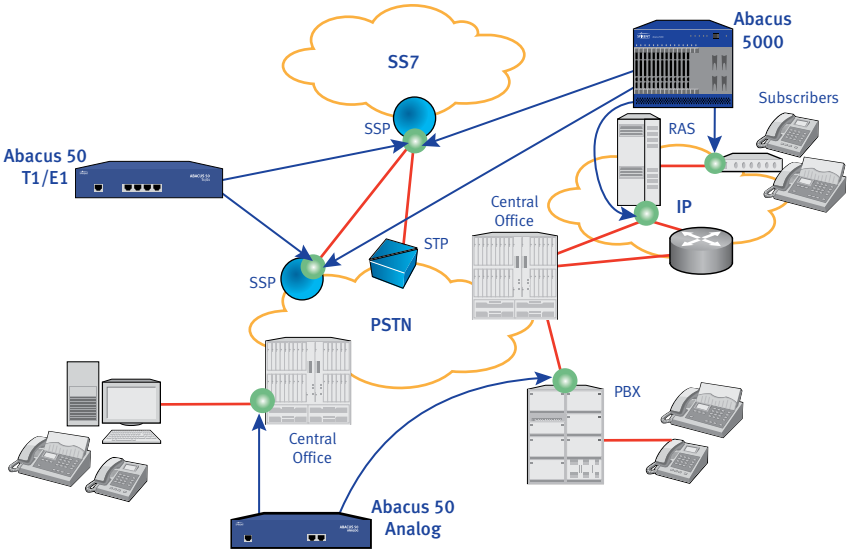
The Abacus 50 T1/E1 test system has the features of an Abacus 5000, but in a smaller form factor and supporting four T1/E1 ports. It is designed for lower port density laboratory test environments or to reduce the cost of deploying a remote testing solution. With the distributed testing firmware option, multiple systems can be viewed as one system for simplified management of multiple Abacus 50 systems (along with Abacus 5000 and Abacus 100 test systems).

The Abacus 50 T1/E1 system supports four T1 or E1 circuits with Channel Associated Signaling (CAS). It provides TDM call generation functionality to test PCM T1/E1 circuits with 24 or 30 channels per circuit. Each channel can be configured as an originating or terminating channel (calling or called party).

When generating calls, the Abacus 50 T1/E1 system executes a call setup/teardown for each channel and executes a media script that includes transmission and reception of audio signals and files.

BENEFITS

- Simplify the testing of converged IP telephony and PSTN networks and services with functional and performance testing for SS7, T.30 fax, V.90 data modem and PSTN/IP ladder diagram
- Enable service providers and enterprises to reduce time to market of services, while assuring they meet the quality requirements as perceived by users
- Achieve overall cost savings by giving the user full flexibility for convergence testing with synchronized Abacus 5000 IP, TDM, analog measurements and the same user interface



FEATURES

- Globally compliant with FCC, NET4, CTR21, JATE, and country-specific PTT specifications
- Programmable protocol state machine
- Compact flash memory to store application programs
- 10/100Base-T Ethernet controlled (RJ-45 front access)
- DSPs provide tone generation, tone recognition, DTMF, PSQM, PESQ, fax, and modem functionality
- On-board TCXO used for TDM clock and 1 PPS signal generation
- Programmable call progress tones
- Supports SS7 (ANSI, ETSI, ITU-T, China and Japan), CAS, MF R1/R1.5/R2, ISDN PRI, GR-303, V5.1/V5.2 and SLC-96/TR08 (Mode 1)
- 4 full-duplex T1 with 24 channels per circuit at the T1 rate (1.544 Mbps)
- 4 full-duplex E1 with 30 channels per circuit at the E1 rate (2.048 Mbps)
- 96 channels (T1) or 120 channels (E1)
- Call generation
- Built-in protocol analyzers
- Custom protocols
- Sends and receives tones, PRBS and speech
- Perform voice quality measurements on each call using PSQM, PSQM+
- PSQM, PSQM+ to MOS conversion
- Perform voice quality measurements on up to 48 channels using PESQ
- MOS-LQO, R-factor (P.834), J-MOS calculations from PESQ measurements
- Performs fax measurements on 96 channels
- Programmable call progress tones
- Detects and forwards DTMF and MF R1/R1.5/R2 pulse dialing
- Flexible call sequences
- Program test duration to be random or fixed from 1 second to indefinite
- Verifies speech path is established and retained for duration of call
- Results are automatically and continuously gathered and presented in tables and graphs
- SS7 event analyzer
- T.30 fax up to V.17 (up to 14.4 kbps)
- Up to V.90 data modem (14 channels max)
- Echo measurements
- QSIG basic call support on ISDN PRI
- BRI over V5
- SS7 COT CCR and advanced
- Voice quality measurements and fax within one script
- SS7 ISUP Configurability
- Call Tracer (ladder diagram for SS7)
- Load Profiling (Saw Tooth, Rectangle, Trapezoid and Poisson)
- Graphical Display of Measurements-over-Time
- Facility message support in PRI
- Idle bit pattern
- Pulse dialing on V5.1 Exchange
- Perform QoS validation using the Scripting for Voice Pattern Matching

PHYSICAL SPECIFICATIONS

Dimensions

- Height: 4.8cm (1.9”) with user-installed feet, 4.4cm (1.8”) without feet
- Width: 24cm (9.5”)
- Depth: 20cm (8”)

Weight

- 1.4Kg (3 Lbs)

Environment

- Operating temperature range: 0-40 °C at 20%-80% non-condensing humidity
- CE marked

LEDs

- Dual LEDs indicate status of channels and one status LED

TONE SPECIFICATIONS

- Send any two frequencies with 1 Hz resolution
- Send noise or silence
- Send with a resolution of 8 ms and an accuracy of ±20 ms
- Detect two frequencies with a minimum difference of 10 Hz for no noise
- Detect energy or silence
- Detect signals with a minimum duration of 40 ms at various thresholds, with an accuracy of ±20 ms

PATH CONFIRMATION SPECIFICATIONS

- 3-tone: Use series of three single frequencies
- Physical: Use series of dual frequencies to identify unique address of channel
- Resilient: Exchange tones with precise voice activation factor (VAF), and measure disturbances in the speech path
- PRBS: Send and receive 2^{11}-1 or 2^{15}-1, and perform full-duplex BERT
- Programmable cut through time

VOICE QUALITY SPECIFICATIONS

- PSQM, PSQM+ and PESQ measurements
- PSQM, PSQM+ to MOS conversion
- MOS-LQO, R-factor (P.834) and J-MOS calculations from PESQ measurements

SPECIFICATIONS FOR MAKING AND RECEIVING CALLS

Making and Receiving Calls, Sending and Receiving Digits

- Signaling: DTMF, MF R1/R1.5/R2, pulse, and custom digits; transmit level, receive level, and digit timing can be configured

DTMF or FSK Caller ID

- DTMF or FSK; send and receive with date and time
- Programmable times for tone on and tone off
- Programmable make interval, break interval, and inter digit pause for pulse dialing
- Number of digits is fixed or automatically detected

Call Progress Tones

- Send and detect dial tone, ring back, busy, and congestion
- Programmable frequencies and cadences

Audio Monitor

- Listen to any 2 channels from the controlling PC

VOICE PATH MEASUREMENT SPECIFICATIONS

Perform measurements on each channel:

- Delays: Dial tone, Single tone, Dual tone, Call acknowledgement, Call setup, Round trip, One way delay
- Hits and Clips
  - Measure up to 1 second of interruptions in speech path (with resilient path confirmation)
- Bit error rate (with PRBS path confirmation)

PROTOCOL SPECIFICATIONS

- CAS, MF R1/R1.5/R2, and pulse dialing
- Primary rate ISDN (US, ETSI, Lucent, Nortel Japan)
- GR-303 (IDT and RDT)
- V5.1 and V5.2
- SS7 (ANSI, ETSI, ITU-T, China, Japan)
- SLC-96/TR08 (Mode 1)
- QSIG basic call support on ISDN PRI

Standard TDM Protocols

- T1 frame format: D4 and ESF
- E1 frame format: 2 Frame, 16 Frame, or 16 Frame with CRC
- T1 line code: AMI, B8ZS
- E1 line code: AMI, HDB3
- Standard T1 protocols: loop start, ground start, E and M
- Standard E1 protocols: R2, China R1, T1097, T0466

Custom TDM Protocols

- Create protocols for T1 and E1
- Create any CAS state machine with unlimited number of states
- Each state sends any signaling bit
- Each state has 16 exit conditions
- Incorporate MF R2 state machine
- Send and detect caller ID and meter pulses

ECHO MEASUREMENT SPECIFICATIONS

- Echo cancellation on/off
- Echo delay
- ERL (Echo Return Loss)
- ERLE measurement (Echo Return Loss Enhancement)
- TELR measurements (Talk Echo Loudness Rating)
- Support echo measurements on 2 channels

COMPONENTS

- Stand alone 1U high 19” rack mountable with included brackets

CAPACITY

- Four T1 or E1 circuits

CONNECTIONS

- Front panel with four RJ-48 connectors for T1 or E1 and one RJ-45 10/100 Ethernet connector
- Back panel with one -48 VDC blocking power connector, grounding post, DB15 connector for future GPS/CDMA time synchronization (currently Ethernet support), and one DB9 connector communication port for configuration

ELECTRICAL

Power supplied through external -48 VDC desktop power supply with locking connector or external -48 VDC source

- 90 to 264 VAC (47 to 63 Hz) or -36 to -72 VDC
- Power draw: Maximum of 25W, 15W typical
- Power switch on back panel with fuse
- T1 transmit level: 3 Vb-p
- E1 transmit level: 2.4 Vb-p for E1 75 ohms; 3 Vb-p for 120 ohms
- Transmit timing: recovered (loop) or derived from internal system clock
- Receive level: 0 to -6 dB from transmit level
- T1 line impedance: 100 ohms
- E1 line impedance: software selectable between 75 ohms and 120 ohms
- Isolation: 500 VAC rms between line and electronics

ORDERING INFORMATION

Abacus 50 T1/E1

- Abacus 50 – T1 with 4 ports, call generation (P/N A-50-002)
- Abacus 50 – E1 with 4 ports, call generation (P/N A-50-004)

Multi-System/Distributed Firmware Option

Required when using Abacus 50 T1/E1, with any Abacus 50, or Abacus 100, or Abacus 5000 systems. Note: Abacus 5000 must also have multi-system/distributed testing option enabled (P/N SWF-3210).

- Enable Abacus system for distributed testing (P/N SWF-3510)

Distributed Abacus 50 Bundles

Includes Abacus 50 with PESQ, PRI, SS7, call generation and distributed firmware options:

- Distributed Abacus 50 – T1 bundle with 4 ports (P/N DA-50-002)
- Distributed Abacus 50 – E1 bundle with 4 ports (P/N DA-50-004)

Firmware Options

- E1 (P/N SWF-3532)
- T1 (P/N SWF-3533)
- PRI (NI, Lucent, Nortel, ETSI) (P/N SWF-3534)
- GR-303 (TMC only), for T1 (P/N SWF-3535)
- V5.1 and V5.2, for E1 (P/N SWF-3536)
- ANSI SS7 (P/N SWF-3537)
- ETSI + ITU-T SS7 (P/N SWF-3538)
- Chinese SS7 (P/N SWF-3539)
- Japanese SS7 (P/N SWF-3540)
- SS7 virtual trunks (P/N SWF-3541)
- PSQM/PSQM+ (P/N SWF-3542)
- PESQ (P/N SWF-3543)
- T.30 fax up to V.17 (P/N SWF-3544)
- V.90 data modem (P/N SWF-3546)
- T.30 FAX/V.90 analog data modem combo (P/N SWF-3549)
- BRI over V5, (requires SWF-3536) (P/N SWF-3551)
- SS7 COT CCR and Advanced (P/N SWF-3553)
- MF R1.5 signaling (P/N SWF-3554)
- SLC-96/TR-08 (Mode 1) (P/N SWF-3555)
- Scripting for Voice Pattern Matching (P/N SWF-3556)
- Echo measurements (P/N SWF-3522)

FOR MORE INFORMATION

Access Abacus 50 information at Spirent Communications' Web site www.spirent.com/go/voice to learn more about Spirent IP Telephony test systems and services, download product literature, white papers and test methodologies. Contact your local sales representative for more details.

SPIRENT GLOBAL SERVICES

Spirent Global Services provides a variety of professional services, support services and education services - all focused on helping customers meet their complex testing and service assurance requirements. For more information, visit the Global Services website at www.spirent.com/gs or contact your Spirent sales representative.

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 P/N 79-000122 Rev. E 11/07