SPIRENT ABACUS

T3 AND E3 TRAFFIC GENERATOR WITH CHANNEL ASSOCIATED SIGNALING

ABACUS 5000—TCG3 SUBSYSTEM

The Abacus 5000™ TCG3 subsystem provides PCM circuits to emulate a telephone exchange (central office) or terminal. The TCG3 subsystem simulates T3 and E3 call generation and switching.

APPLICATIONS

VoIP Convergence

- Test convergence to VoIP devices (in combination with ICG3 subsystem)
- 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 from a T3, or E3 link into an IP network through a media gateway

PBXs, Switches, Central Offices

- Create traffic
- Determine capacity

Transmission Equipment, Channel Banks, Multiplexers

- Verify transmission quality

Voicemail, IVR

- Transmit and receive account codes
- Create traffic to leave messages
- Replay messages

Switching

- Switch TCG3 interface to TCG3, PCG3 interfaces

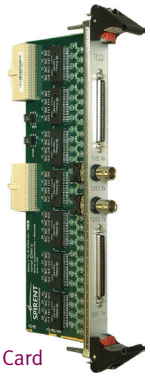
The TCG3 subsystem provides a full-duplex T3 or E3 circuit for TDM call generation and switching, with channel associated signaling (CAS).

In call generation mode, the TCG3 subsystem executes a call setup/teardown for each channel and executes a media script that includes transmission and reception of audio signals and files. When performing switching, it routes a call from one channel to another channel on the Abacus 5000 system, based on the number dialed by the SUT.

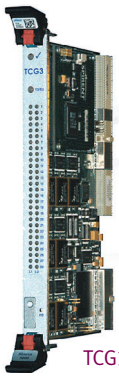
Spirent has the most complete TDM, VoIP and analog solution in one platform using the same GUI. For TDM: TCG3, PCG3. For VoIP: ICG3. For analog: ECG3, and Abacus 100 Analog.

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 diagrams
- Achieve overall cost savings by giving the user full flexibility in convergence testing with synchronized IP, TDM and analog measurements with the same user interface



TCI3 Rear Card



TCG3 Front Card

ABACUS 5000—TCG3 SUBSYSTEM

FEATURES

- One full duplex circuit
- Channelized T3 (45 Mbps) or E3 (34 Mbps) output
- T3: 28 T1 links
- E3: 16 E1 links
- 672 channels in T3 mode
- 480 channels in E3 mode
- Supports SS7 (ANSI, ETSI, ITU-T, CAS, MF R1/R2, ISDN PRI, GR-303, SLC-96/TR08 (Mode 1)
- Call generation and switching
- Built-in protocol analyzers
- Custom protocols
- Sends and receives tones, PRBS, speech
- Performs voice quality measurements on each call using PSQM, PSQM+
- PSQM, PSQM+ to MOS conversion
- MOS-LQO, R-factor (P.834), J-MOS calculations from PESQ measurements
- Performs voice quality measurements on 392 channels using PESQ
- Performs fax measurements on 336 channels
- Programmable call progress tones
- Detects and forwards DTMF, MF R1/R2 and pulse dialing
- Flexible call sequences
- Generates over 900,000 (T3 PRI-ISDN) calls per hour per subsystem
- Switches over 450,000 (T3 PRI-ISDN) calls per hour per subsystem
- Program test duration to be random or fixed (minimum 1 second)
- 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 14.4 kbps
- Up to V.90 data modem (112 channels max)
- 16 groups of SS7, PRI, GR-303 or V5 trunks per subsystem
- Voice quality measurements and fax within one script
- 2048 logical channels with GR-303 concentration
- SS7 CIC phone book
- 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
- Perform QoS validation using the Scripting for Voice Pattern Matching

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 any 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
- 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/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 timer for tone on and tone off
- Programmable make interval, break interval, and inter digit pause for pulse dialing
- Number of digits selectable between fixed or automatically detected

Call Progress Tones

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

Audio Monitor

- Listen to any 2 channels from the controlling PC

VOICE PATH MEASUREMENT SPECIFICATIONS

Measurements Performed on Each Channel

- Delays: Dial tone, Single tone, Dual tone, Call acknowledgement, Call setup, Round trip delay, 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)

SWITCHING SPECIFICATIONS

- Number dialed into a switching channel can comprise called and calling party numbers
- Number forwarded from Abacus 5000 can comprise called and calling party numbers
- Called and calling party numbers can be received and forwarded with prefix and suffix
- A single number can be allocated to any number of channels on Abacus 5000
- Maximum of 5 subsystems that stand in the most left shelf slots

PROTOCOL SPECIFICATIONS

- CAS, MF R1, MF R2, and pulse dialing
- Primary rate ISDN (US, ETSI, Lucent, Nortel Japan)
- GR-303 (IDT and RDT)
- SS7 (ANSI, ETSI, ITU-T)
- SLC-96/TR08 (Mode 1)

STANDARD FRAMING, LINE CODING AND SIGNALING SPECIFICATIONS

- 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
- T3 line code: B3ZS
- E3 line code: HDB3
- Signaling included with T1 option: loop start, ground start, E&M, FGD
- Signaling included with E1 option: R2, China R1, T1097, T0466, E&M, Q.50

CUSTOM TDM PROTOCOL SPECIFICATIONS

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

ABACUS 5000—TCG3 SUBSYSTEM

INTERFACES

- TCG3 subsystem for call generation, T3 and E3

PHYSICAL CONNECTIONS

- TCG3 front card with active components fits into one Abacus 5000 slot
- TCI3 rear card provides BNC and 68-pin connectors

ELECTRICAL SPECIFICATIONS

- T3 transmit level: 800 mVb-p
- E3 transmit level: 1000 mVb-p
- Transmit timing: recovered (loop) or derived from internal system clock
- Receive level: 0 to -6 dB from transmit level
- Line impedance: 75 ohms
- Isolation: 500 VAC rms between line and electronic

ORDERING INFORMATION

- TCG3 subsystem for call generation: T3, with TCI3 (P/N TCG-3000B)
- TCG3 subsystem for call generation: E3, with TCI3 (P/N TCG-3001B)
- TCG3 subsystem with switching: T3 emulates exchange, with TCI3 (P/N TCG-3002B)
- TCG3 subsystem with switching: E3 emulates exchange, with TCI3 (P/N TCG-3003B)
- TDM Bundles
  - T3 Call Gen—Abacus 5000, with CAS, PRI and SS7 (P/N TCG-3010B)
  - E3 Call Gen—Abacus 5000, with CAS, PRI and ETSI+ITU-T SS7 (P/N TCG-3011B)

Firmware Options

- Call generation (P/N SWF-3092)
- Switching (P/N SWF-3093)
- E3 (P/N SWF-3094)
- T3 (P/N SWF-3095)
- PRI: NI, ETSI, ITU-T (P/N SWF-3096)
- GR-303 (P/N SWF-3097)
- ANSI SS7 (P/N SWF-3099)
- ETSI + ITU-T SS7 (P/N SWF-3100)
- SS7 Virtual Trunks (P/N SWF-3103)
- PSQM, PSQM+ (P/N SWF-3104)
- PESQ (P/N SWF-3105)
- T.30 FAX (up to V.17) (P/N SWF-3106)
- V.90 data modem (P/N SWF-3108)
- T.30 fax and V.90 data modem combo (P/N SWF-3109)
- SLC-96/TR-08 (Mode 1) (P/N SWF-3118)

FOR MORE INFORMATION

Visit Spirent Communications' Web site at www.spirent.com/go/voice where you can learn about Spirent IP Telephony test systems and services, download product literature, white papers and test methodologies. Contact your local sales representative for 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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