

PBX and Central Office Simulator



Switch, Gateway, Transmission Testing



Digital Switching with Protocol Conversion



Digital T1/E1 Interfaces (s/w selectable)



Analog/BRI Telephone Interfaces



R1, MFC-R2, PRI-ISDN, SS7 Signaling Protocols (includes user-defined CAS State Machine)



Bulk Call Generation with Scripting Capability



Load Testing with Real Time Fax Calls, Modem Traffic, Voice Files



Call Quality Assessment using PAMS, PSQM, PESQ



Remote Access Capability



Generate Call Records



Digital Central Office Switch Simulator (DCOSS)



The Digital Central Office Switch Simulator (DCOSS) converts a Pentium PC (portable, tower, rack-mount) into a digital central office switch simulator, PBX and switch, complete with T1, E1, and POTS Interfaces. A user-friendly graphical interface (GUI), through which complex switching, signaling, and digital transmission functions are easily controlled, provides the ease of operation as well as the flexibility required from telephony test equipment. DCOSS is ideal for simulating and testing advanced telecom networks and products, including switches, gateways, and transmission systems. The DCOSS can also be used for verifying T1/E1 signaling protocols of new systems. These protocols include R1, MFC-R2, PRI ISDN, SS7 and SS5.

DCOSS Main Features

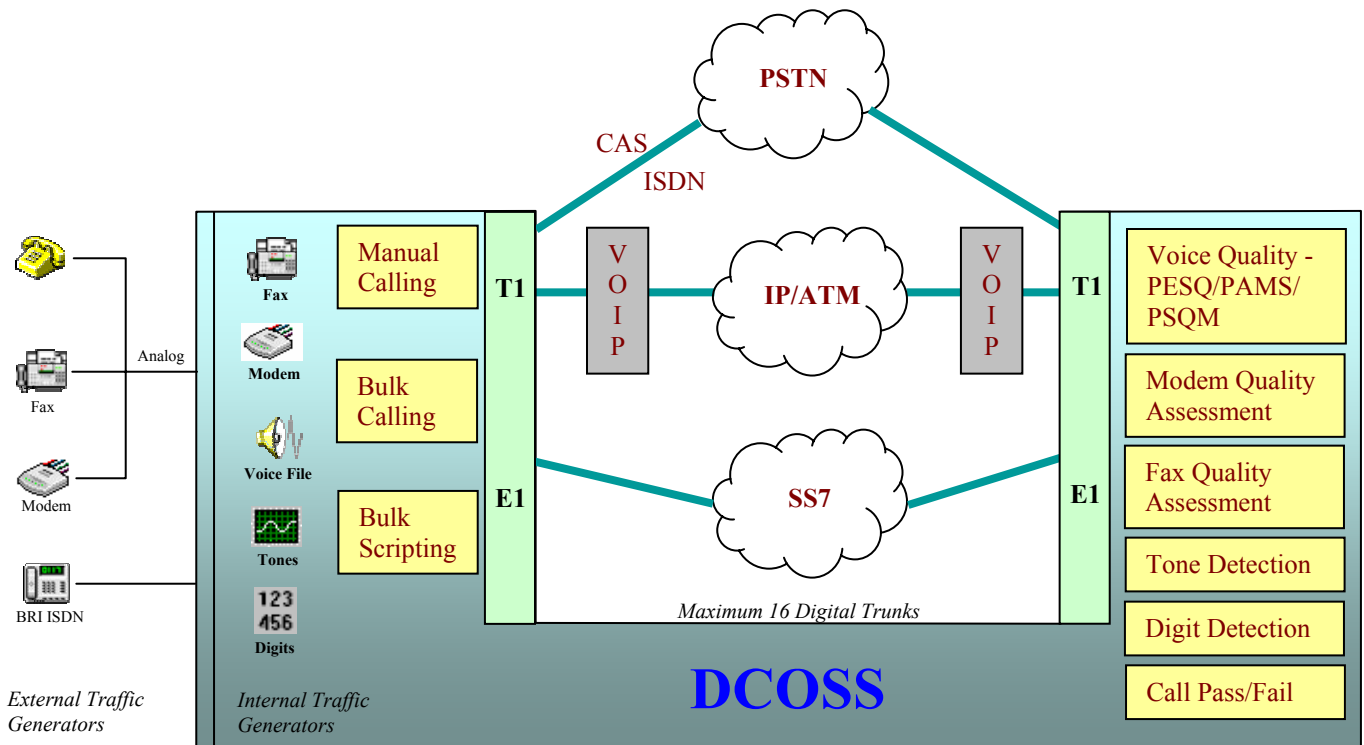
- Win XP Operating System with User-Friendly GUI.
- Portable system with digital T1/E1 trunks, BRI ISDN and POTS (Plain Old Telephone Set) Interfaces.
- Variety of Signaling Protocols including R1 (robbed-bit), MFC-R2, PRI ISDN (includes NFAS), SS7, SS5, and others.
- CAS State Machine with user-defined scripting. Works with DCOSS APS.
- Analog (FXO) and Digital (BRI) phone simulation.
- Generates and Receives Manual or Automatic Calls Simultaneously on any or all Timeslots.
- Record and Playback of PCM Voice Files simultaneously over all timeslots. Includes Voice Quality Assessment using PESQ, PAMS and PSQM Plus.
- Real-Time FAX Call Generation/Reception (V.29, V.27, V.17, V.33) simultaneously over all timeslots. Includes Fax Quality Assessment.
- Send/Receive Modem Traffic (V.21, V.23, V.34, V.90, V.92) simultaneously over all timeslots. Includes Modem Quality Assessment.
- Transmit and Detect DTMF/MF digits simultaneously over all timeslots.
- Transmit and Detect Single- or Dual-Frequency Tones simultaneously over all timeslots.
- Switches Calls among Timeslots/Trunks with Protocol Conversion capability.
- Pass/Fail Calls using User-Defined Pass Criteria.
- Real Time System Statistics with Hourly Information as well as Real Time Status.
- Multiple 2-Wire (RJ11) Standard Telephone Interfaces (16, 32).
- BRI ISDN European ST-Interface (16 port).
- Manual or Automatic ANI (Caller ID) Generation.
- Remote Access Capability (Client/Server) using GUI or Command Prompt. All DCOSS functions supported remotely. Access to OCX/DLL to develop user-defined client.
- Automatic Bulk Call Generation/Reception for load testing digital/analog trunks.
- Bulk Call Scripting with simple point-and-click script setup. Allows for conditional commands as well as script looping.

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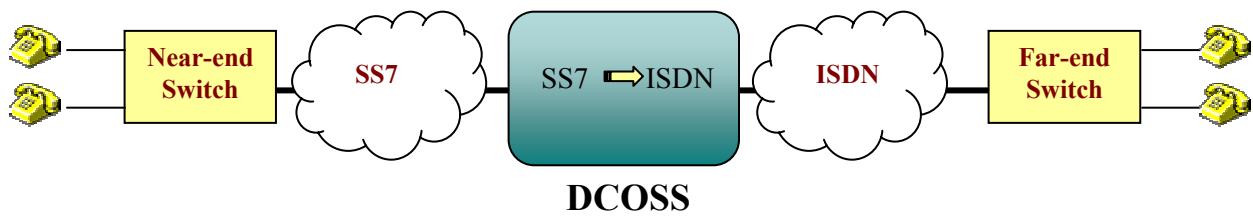
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DCOSS provides the solution required for testing switches, gateways, and networks using a variety of signaling protocols. As depicted below, DCOSS can simulate both ends (or one end) of a complex switching network with a variety of traffic generators (both internal and external). DCOSS can also be controlled remotely using the optional Client/Server application. The Remote Capability allows full control of many DCOSS's from a single client as well as many clients accessing a single DCOSS.

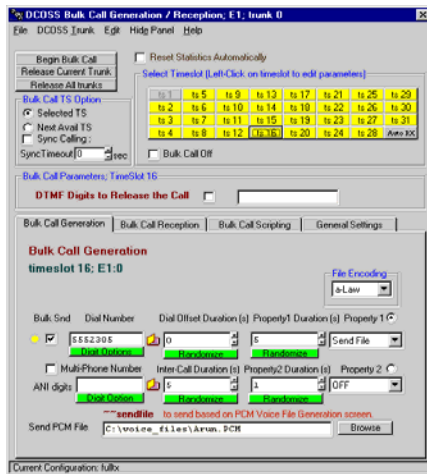


Using the DCOSS graphical interface, complex switching solutions can easily be configured while providing total voice/data throughput. As depicted below, protocol conversion is accomplished during call switching.

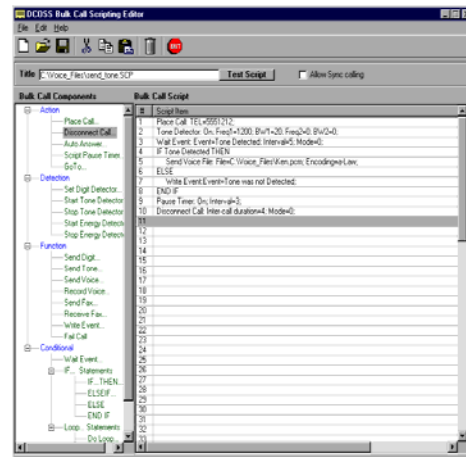


Simulating traffic and analyzing traffic are integral parts of a central office simulator test tool. DCOSS provides the functionality of bulk calling and utilizes a user-friendly GUI for configuring simplistic as well as complex scenarios on a per-timeslot basis. When more complex configurations are desired, including conditional statements and Do Loops, DCOSS provides Bulk Call Scripting, which utilizes a point-and-click interface for creating and editing bulk call scripts.

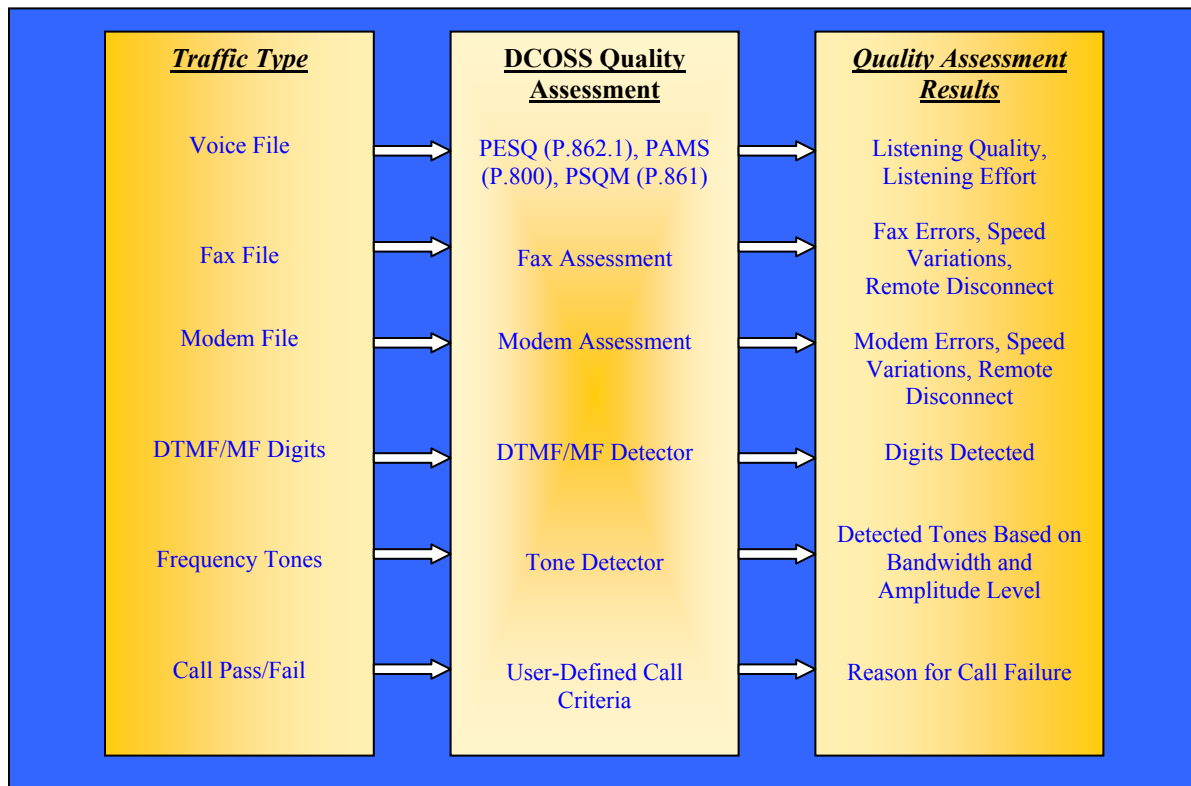
Bulk Call Generation/Reception



Bulk Call Scripting



During bulk calling and manual calling, DCOSS has the ability to analyze the generated traffic over the digital T1/E1 interfaces. As a result, DCOSS provides a quality assessment of the traffic.



DCOSS Specifications

T1/E1 Line Interface

T1 Signal	RJ48c (ANSI T1.102, T1.403)
T1 Line Code Format	AMI or B8ZS
T1 Framing Format	ESF or D4
E1 Signal	RJ48c (120 ohm)
E1 Line Code Format	HDB3 or AMI
E1 Framing Format	CEPT G.703/G.704 CAS
Onboard DSPs	100 MIPS each
Trunks Avail per System	2 to 16 T1/E1 (s/w selectable)

Audio Signal Processing

Receive Range	-68 to +3 dBm
Transmit	-28 to 0 dBm
Sampling Rates	8 ksamples/sec (telephone industry standard)

DTMF/MF Tone Dialing/Detection

DTMF Digits	0-9, *, # and ABCD per ITU-T Q.23 and Q.24
Dynamic Range	-40 dBm to 0 dBm nominal
Tone Duration	40 ms (minimum)
MF Digits	0-9, KP, ST, ST1, ST2, ST3 per US (R1) and ITU-T Q.321
Dynamic Range	-35 dBm to -5 dBm nominal

POTS (Plain Old Telephone Set) Interface

Line interface	600 ohms
Return loss	ERL: 20 dB, SRL HI:30 dB, SRL LO: 11 dB
Transmit Loss	0 dB, +1 dB (min), -1.3 dB (max)
Receive Loss	0 dB, +1 dB (min), -1.3 dB (max)
Attenuation	0 dB to -9 dB (3 dB increments)
PCM encoding	u-law or A-law

Standards Compliance

T1 and E1 Standards	G.703, G704, G.706, G.732, G.823
DS1 metallic interface	ANSI T1E1/88-00
European EMI/EMC	EN55022, EN50082-1
PRI ISDN	Q.931
ANSI T1 PRI Interface	T1E1.4/8868 April 88
PRI User-Net Layer 1	ITU-T I4.31, June 88
ISDN PRI Interface	AT&T Pub. TR41449 and TR41459
ITU SS7	Q.701-Q.704, Q.761-Q.764
ANSI SS7	ANSI T1.111, 113

Optional Software/Hardware

Voice Quality Testing using PESQ, PAMS, PSQM
Modem Capability with Modem Quality Assessment
Fax Capability with Fax Quality Assessment
Bulk Call Scripting
Remote Access
Analog Phone Simulator (APS)

Available Protocols

- R1 (WINK)
- MFC-R2
Certified Countries Available: Argentina, Bahrain, Bolivia, Brazil, Chile, China, Columbia, Czech Republic, Honduras, Indonesia, Korea, Malaysia, Mexico, Panama, Philippines, Singapore, Thailand, and CCITT.
- E1: Digital E&M, Digital CAS
- T1: Loop Start, Ground Start, Immediate Start
- SS5 Signaling
- SS7 Signaling
Available Protocol Layers: MTP 1-3, ISUP, TUP, TCAP, and SCCP
Interface: Dual T1/E1 (s/w selectable)
Support for 4 Links or 16 Links
User Friendly GUI for SS7 Configuration
- PRI ISDN (Europe, USA, Asia)
Europe
France Telecom VN6 (ETSI)
EuroISDN (Available Countries: Austria, Denmark, Finland, Greece, Iceland, Ireland, Italy, Liechtenstein, Luxembourg, Netherlands, Norway, Portugal, and Spain)
Belgium, China, Great Britain, Sweden, Germany, Singapore
European QSIG (ETSI)
USA
Northern Telecom DMS 100
AT&T 5ESS10
AT&T 4ESS
US National ISDN2 (BellCore)
USA QSIG
Asia
Australian Telecom 1
Hong Kong Telephone
Nippon Telegraph Telephone
Korean Telecom
- No Call Control

Ordering Information

- Portable Lunchbox Computer, Rackmount Computer, or Desktop computer
- Dual and Quad T1/E1 Interfaces
- Two-Wire Station Interfaces
- CAS, PRI ISDN, SS7, SS5 Protocols
- Fax Capability
- Modem Capability (V.21, V.23, V.34, V.90, V.92)
- Voice Quality Assessment (PESQ, PAMS, PSQM)

Specifications are subject to change without notice.