DEGEM SYSTEMS

Com & Telecom

MDC-3281

Telecommunication Networks Training System
Analog, digital Telephony & VoIP

Modern Communication

Fiber Optic Communication

Antennas

Radar

Cellular Communication

Global Position Systems

Satellite Communication

Microwaves

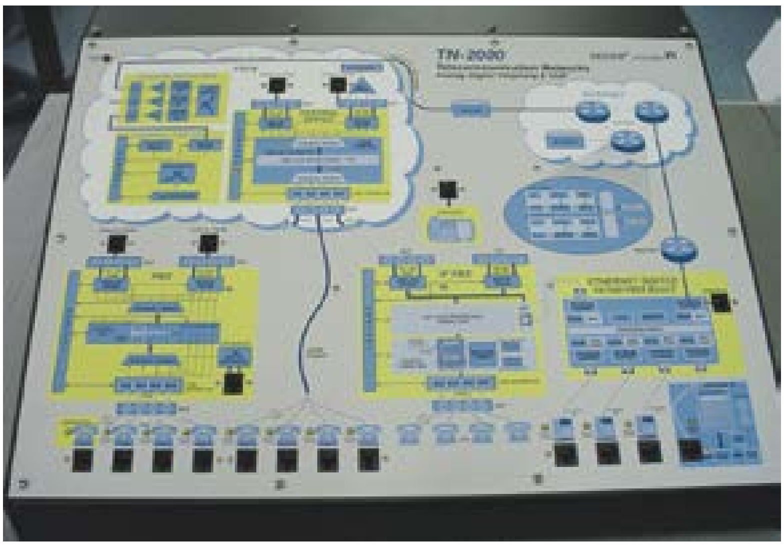
Telecommunication Networks

MDC-3281 is a unique training system that integrates state-of-the-art hardware with interactive courseware interactive courseware for learning fundamentals of modern electronic switching for both conventional Public Switched Telephone Networks (PSTN) and Voice over Internet Protocol (VoIP).

A real electronic exchange, which is controlled by the student's PC, provides practical exercises to reinforce the theory learned. Each experiment is accompanied by detailed procedures to guide the student through each experiment with a minimum of instructor intervention.

This course is suitable for technicians and engineers who wish to learn the basic principles of PSTN and VoIP switching in the classroom or laboratory.

- Computer-controlled switch
- 8 standard PSTN telephones
- 4 VoIP telephones
- Lesson modularity affords easy adaptation to various training programs
- Several vivid animations and graphics to illustrate PSTN and VoIP concepts
- Student theory comprehension is assessed by multiple choice questions
- Clear, step-by-step procedure for each experiment
- No subscriber line necessary



Specifications

TECHNICAL CHARACTERISTICS

- Operating voltage (100-250V)
- Main frequency (50-60Hz)
- Telephone ports (8 legacy, 4 IP)
- Off hook detect (0-80mA)
- Loop start threshold (54-140mA)
- Ring frequency (10-100Hz)
- Ring detect (22V)
- PBX features:
  - Call transfer
  - Follow me
  - Call forward
  - Call waiting
  - Call parking
  - Music on hold
- PC control link (USB)

SWITCHING EXCHANGE THEORY

Telephone network overview
- Brief history
- PSTN components
- Transmission media
- Cellular network
- IP network and VoIP

Telephony interfaces
- Analog interfaces
- FXS and FXO
- Digital interfaces
- E1/T1 and BRI, PRI

Legacy network signaling
- Analog line signaling
- Loop start, ground start
- Ring, E & M
- Address signaling: pulse, DTMF
- Information signaling
- Tones and cadences

Signaling protocols – IP network
- SIP, H.323
- RTP & RTCP media transport
- RSVP & DNS support services

Voice handling
- A/D, D/A:
- PCM encoding
- Echo encoding

Voice handling – IP network
- Voice compression algorithms
- DPCM, CELP, VCELP
- MGCP (media gateway control protocol)

Switching and multiplexing technologies
- Switching: space, time (TSI)
- Packet switching
- Multiplexing
- FDM
- TDM: E1, T1, 24 & 30 channels

Switching technology – IP network
- Router
- IP-IP telephone connection

Call management – legacy networks
- Call setup flowcharts
- Incoming call, outgoing call
- Numbering plan
- Local call
- Long distance (intercity) call
- International call

Call management – IP network
- Call setup flowcharts
- IP phone - IP phone call management

Private branch exchange (PBX)
- Architecture and configurations
- Interfaces
- User features
- Automated call distribution
- (ACD, hunt group)
- Operator station

EXPERIMENTS

- Analog interface (POTS)
- FXS (foreign exchange subscriber)
- Analog line signaling
- Command line interface (CLI)
- Tones and cadences
- Voice digitization, PCM encoding
- IP voice compression algorithms
- Time switching
- Packet switching
- Legacy call management: incoming, outgoing, local, long distance and international calls
- IP – IP call management
- PBX user features
- PBX operator station functions
- IP PBX features: call transfer, follow me, call forward, call waiting, parking

REQUIRED ACCESSORIES

Personal computer with MS-Windows

Specifications are subject to change without prior notice