

Analog Phone Simulator (MAPS™ APS)

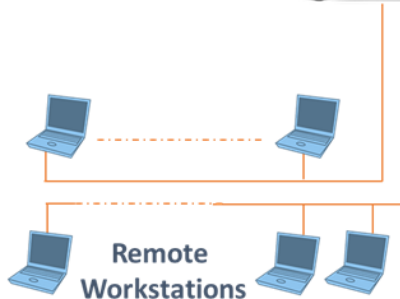
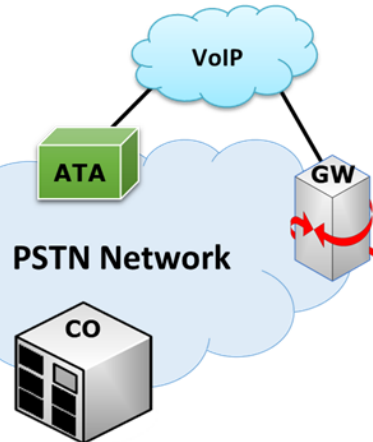
24-Port, 48-Port, 96-Port, 192-Port FXO and FXS

MAPS™ APS-24, 48, 96, or 192 Port

GL solution simulates these endpoint devices



RJ-11



- Bulk call generator in minutes
- FXO and FXS Capabilities
- Plug and play
- Assessment of Voice Quality, Fax Quality
- Tones and Digits Detection
- Scalable from hundreds to thousands of calls
- Remoting, Automation, Scripting, Reporting

Overview

MAPS™ APS is a high capacity analog 2-Wire FXO/FXS or 4-Wire E&M Bulk Call Generator that performs QA tests for Central Office, PBX, Analog Telephone Adapter, Gateway or other telecommunications equipment, which provide local loop interfaces. It includes server hardware, GL MAPS™ software, and channel bank(s), along with optional modules (Fax Emulation and Voice Quality Testing Analysis) in a compact rackmount system. MAPS™ APS system supports up to 96 independent FXO ports or FXS ports per 1U MAPS™ APS/ALS server. More can be achieved by simply scaling the system with a 4U MAPS™ APS Server connected with 2 Octal T1 E1 Cards which can then support up to 384 analog ports.

MAPS™ APS supports Supplementary Service Testing and [Interactive Voice Response \(IVR\)](#) depends on DTMF digits, or tones, as inputs from the users, based on which the system presents a menu for automating various services to users. Provides high-density connection to any 2-wire analog interface for fully automated custom testing.

MAPS™ platforms offers automated, scripted, multi-user, multi-protocol, and high capacity Bulk Call Generation. This platform is the basis for all signaling protocols and for traffic generation – whether voice, tones, digits, fax, data, or video, depending on the network support. MAPS™ covers legacy PSTN, TDM, SONET SDH, next generation VoIP, and wireless protocols, interfaces, and equipment. MAPS™ can support any of the following protocols in TDM networks for establishing signaling links and generate or receive traffic - CAS, FXO FXS, ISDN, SS7, PPP, GSM, INAP, CAP, and MAP.

For more information, refer to [2Wire Analog and 4Wire E and M Bulk Call Generator](#) webpage.

Supported Call Scenarios

- Caller ID
- Two-way Calling
- Three-way Conference Calling
- Three-way Calling with Calling Party Number ID
- VMWI – Voice Mail with MWI (message waiting indicator) and SDT (stutter dial tone)
- Call Waiting – Detect tone, Call ID, Flash to accept call
- Call Forwarding



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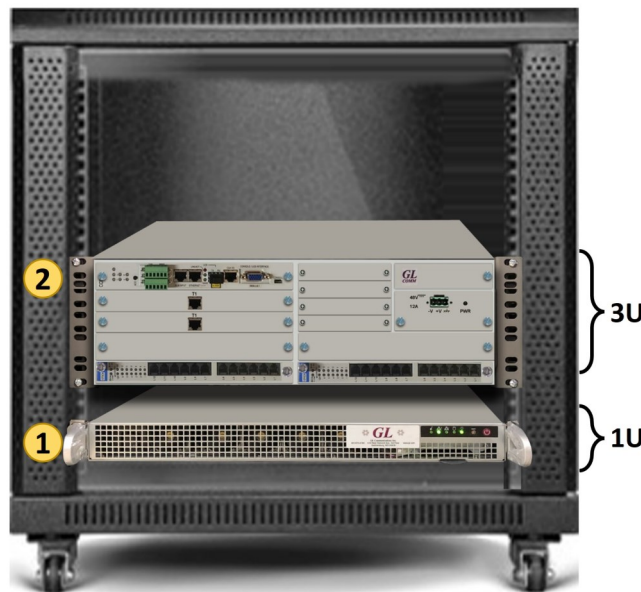
Main Features

- Up to 192 independent ports per 1U MAPS™ APS (More can be achieved by scaling)
- Test central office, PBX, Gateway, Analog/Digital/VoIP networks
- Call monitoring and call recording
- Concurrent users and tests per system
- Fully Automated with Command Line Interface (CLI) and external control
- Full FXO and FXS Functionality via flexible scripts—Refer to Functional Specifications for complete details
- Supports Interactive Voice Response (IVR) using GL's Speech Transcription Server
- API support (Python, Java) for integration with automation frameworks
- Supports E&M (Type I, II, III, IV, V) signaling – immediate start, wink start, delay start
- Voiceband Measurement Tests using T1E1 Ports and VF ports

Hardware Specifications

MAPS™ APS - 24 Port (FXO and FXS)

MAPS™ APS 24 Ports

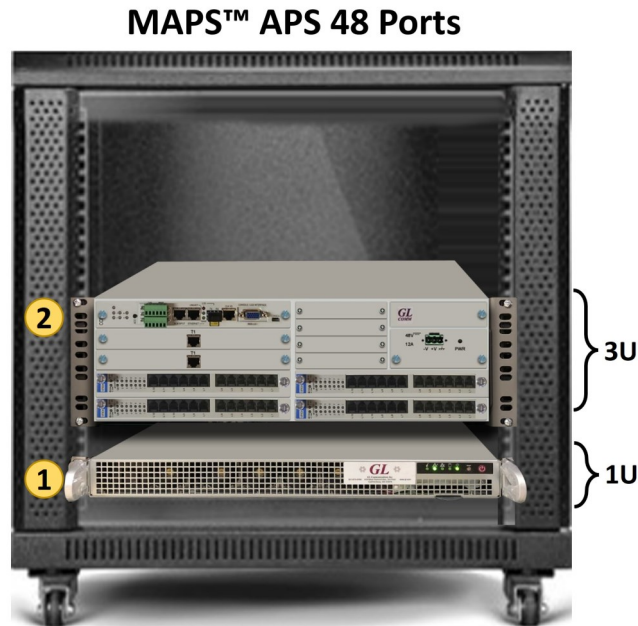


- 1** MAPS™ APS Server (Optional VQT Analysis)
- 2** APSCB-24 x 1

- 19" 1U Xeon Rack PC - MAPS™ APS
 - GL's Dual T1 (2 x T1) for Call Control
 - Automated APS Signaling emulation over TDM
 - T1 crossover cables to connect GL's Dual T1 to APSCB-24
- 19" 1U Xeon Rack PC - VQT (Optional)
- 1xAPSCB– Channel Bank
 - 24 Analog Channels using 2x12 channel FXO Voice Service Card (Each FXO card supports 12 Analog FXO ports)
 - Dual T1 Service Card

Hardware Specifications (Contd.)

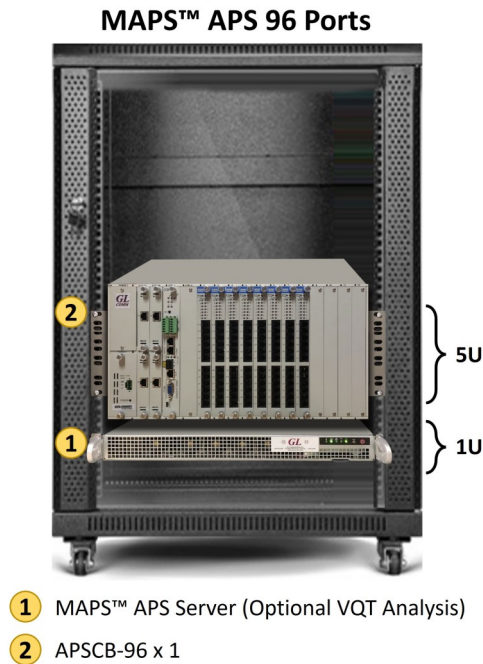
MAPS™ APS - 48 Port (FXO and FXS)



- 19" 1U Xeon Rack PC - MAPS™ APS
 - GL's Quad T1 (2x T1) - for Call Control
 - Automated APS Signaling emulation over TDM
 - T1 crossover cables to connect GL's Quad T1 to APSCB-48
- 19" 1U Xeon Rack PC - VQT (Optional)
- 1xAPSCB-48 Channel Bank
 - 48 Analog Channels using 4x12 channel FXO Voice Service Card (Each FXO card supports 12 Analog FXO ports)
 - 1x Dual T1 Service Card

Hardware Specifications (Contd.)

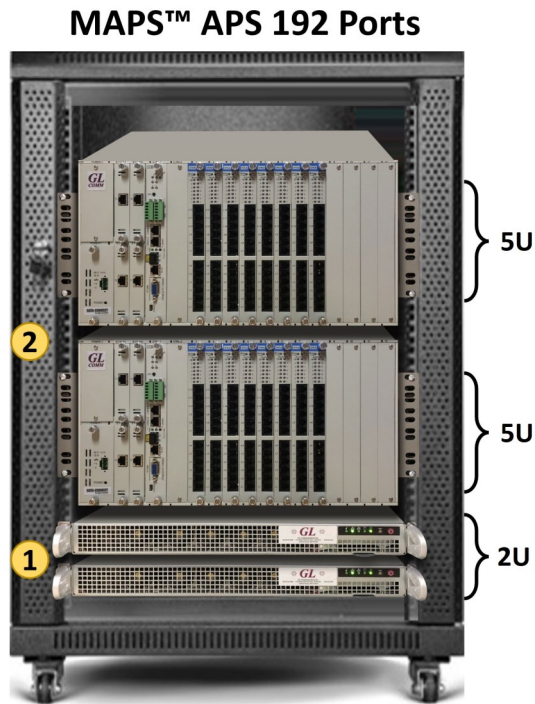
MAPS™ APS - 96 Port (FXO and FXS)



- 19" 1U Xeon Rack PC - MAPS™ APS
 - GL's Quad T1 (4x T1) for Call Control
 - Automated APS Signaling emulation over TDM
 - T1 cross-over cables to connect GL's Quad T1 to APSCB-96
- 19" 1U Xeon Rack PC - VQT (Optional)
- APSCB-96 Channel Bank
 - 96 Analog Channels using 8x12 channel FXO Voice Service Card (Each FXO card supports 12 Analog FXO ports)
 - 2x Dual T1 Service Card

Hardware Specifications (Contd.)

MAPS™ APS - 192 Port (FXO and FXS)



- ① MAPS™ APS Server (Optional VQT Analysis)
- ② APSCB-96 x 2

- 19" 1U Xeon Rack PC - MAPS™ APS
 - GL's Octal T1 (8x T1) for Call Control
 - Automated APS Signaling emulation over TDM
 - T1 crossover cables to connect GL's Octal T1 to 2xAPSCB-96
- 19" 1U Xeon Rack PC - VQT (Optional)
- 2xAPSCB-96 Channel Bank
 - 192 Analog Channels using 16x12 channel FXO Voice Service Card (Each FXO card supports 12 Analog FXO ports)
 - 4x Dual T1 Service Card

Hardware Specifications (Contd.)

VQuad™ HD 24-Port (WB FXO)

(Supports NB, WB)



- 1 VQuad™ System w/24 FXO HD Ports
- 2 VQT Central System
(WebViewer w/Oracle DB, PESQ, POLQA)

Hardware Specifications

- 19" 2U Xeon Rack PC - VQuad™ FXO
 - GL's Dual UTA HD (12x) for Call Control
 - VQuad™ for Automated CAS Signaling emulation over TDM
- 19" 1U Xeon Rack PC - VQT/WebViewer™
 - PESQ and/or POLQA Server License
 - WebViewer™ w/Oracle
- 24-port Patch Panel with 50pin Telco RJ-21 Male to 24xRJ-11 ports

VQuad™ 24-port HD FXO solution is an all-in-one 2U rack supporting both Wide Band (WB) and Narrow Band (NB) Audio. Multiple VQuad 24-Port FXO Racks can be connected virtually unlimited FXO ports. Similar to MAPS™ APS solutions, this solution also includes all the required functionalities of analog emulation—Refer to [Functional Specifications](#) for complete details.

The 2U VQuad™ HD FXO system incorporates two embedded SBCs (Single Board Computer), each running latest VQuad™ software, along with 12 Dual UTA HD units supporting a total 24 FXO Analog ports. All 24 FXO ports are independent and can be fully automated or remote controlled for a fully autonomous test solution. Access to the 24 Analog ports is via an Amphenol connector on the back of the system which can be broken out to 24 RJ11 connections. Each embedded SBC includes two Gigabit Ethernet connections, two USB connections, and a VGA monitor connection. LED lights on the front of the 2U Rack display status of each SBC along with each Dual UTA HD.

When combined with the [WebViewer™](#) Central System all results and events can be saved directly to a Central Database and accessed using the WebViewer (direct access to the database tables is also available). Analysis is provided using GL [Voice Quality Analysis](#) (both POLQA and PESQ supported), [Echo Measurement Utility](#) (EMU), [Fax Analysis](#), and [Voice Band Analyzer](#) (VBA) applications. Like MAPS™ APS solutions, VQuad™ 24-Port FXO solution also includes all the required functionalities of analog emulation.

- Supports 24 independent HD FXO ports per VQuad™ system (2U)
- Scalable solution for unlimited number of FXO ports
- Wide Band (WB) and Narrow Band (NB) support (for HD and SD Audio)
- Multiple Users and Tests per system
- Run tests between systems
- Fully automated and remote accessible via CLI
- Remote accessible via Central Database
- Full FXO Functionality and Analysis via flexible Scripts

Functional Specifications

FXO Capabilities

- Support for up to 96 independent FXO ports per 1U MAPS™ APS (more can be achieved by scaling)
- Full FXO functionality via flexible scripts
- Narrowband supported
- Supports Loop Start and Ground Start signaling
- **Supported call scenarios**
 - Caller ID
 - Two-way Calling
 - 3-way Conference Calling
 - 3-way Calling with Calling Party Number
 - VMWI – Voice Mail with MWI (message waiting indicator) , SDT (stutter dial tone) and SIT (special information tone)
 - Call Waiting – Detect tone, Call ID, Flash to accept call
 - Call Forwarding

FXS Capabilities

- Support Up to 96 independent FXS ports per 1U MAPS™ APS (more can be achieved by scaling, requires FXS voice cards)
- Central office emulation with two way calling
- Supports Loop Start and Ground Start signaling
- User-programmable call progress tone generation for different countries/regions:
 - Dial tone
 - Ringback tone
 - Busy tone
 - Reorder tone
 - Howler tone (extended off-hook signal)
- Ring generation with programmable ring cadence

Reporting

- Multi-User, Multi-Test reporting
- Reports Executed, Successful, and Failed test cases
- Call Failure, Completion, and Call Drop (sustain calls) events
- Voice Quality Test MOS Scores
- Delay Measurements (OWD, PDD)
- Summarization with Failure Details sufficient to determine root cause
- Report generation supports PDF and CSV file formats
- Central DB of events/results/errors

Functionalities

- Basic Telephony functions
 - Any Country signaling
 - On-hook, Off-hook, Flash hook
 - Ringing, Ring back, Dial tones, Digits
 - Path verification
 - Call Fail – No Ring-back, Busy, Fast Busy, SIT
- Traffic
 - Voice, DTMF/MF Digits, Single/Dual Tones, Fax, Voice Quality
- Digit Related Functions
 - Send digits, Detect digits
- File transfer functions
 - Send file, Receive file
- Tone Related Functions
 - Detect busy tone
 - Detect call waiting tone
 - Detect dial tone
 - Detect reorder tone
 - Detect ring-back tone
 - Detect special dial tone
 - Detect test tone
 - Detect tone
 - Send test tone
 - Send tone
- Fax Related Functions
 - Send fax, Receive fax
- FSK Related Functions
 - Detect Caller ID, Detect VMWI
- VQT (Optional)
 - MOS, E-Model, PESQ, POLQA Scores

Voiceband Measurements Capabilities for VG3 and VG6

- Voice Quality Test (VQT)
- Twenty Three Tone Test
- Signal-to-Noise Ratio and Level
- Three Tone Slope (Gain Slope)
- C Notched Noise (CNN) Test
- 1004 Hz Net Loss
- Attenuation Distortion
- Intermodulation Distortion (IMD)
- Impulse Noise

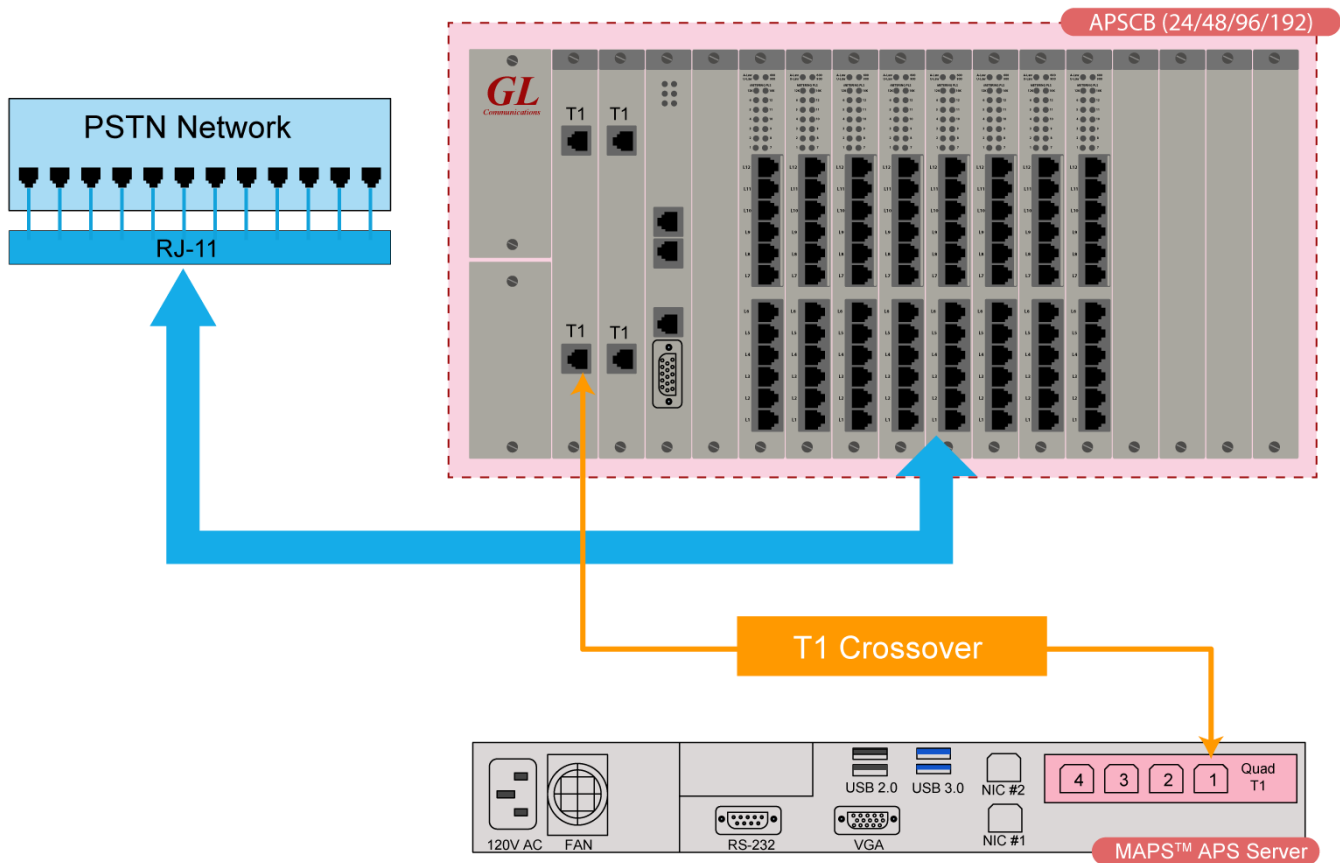
CLI Capabilities

- Client-server application (TCP/IP)
- Script-based call generation, call flow control and feature testing
- Comprehensive API
- Support for Python and Java clients
- Connect multiple clients simultaneously
- Independent execution available for all lines

Working Principle

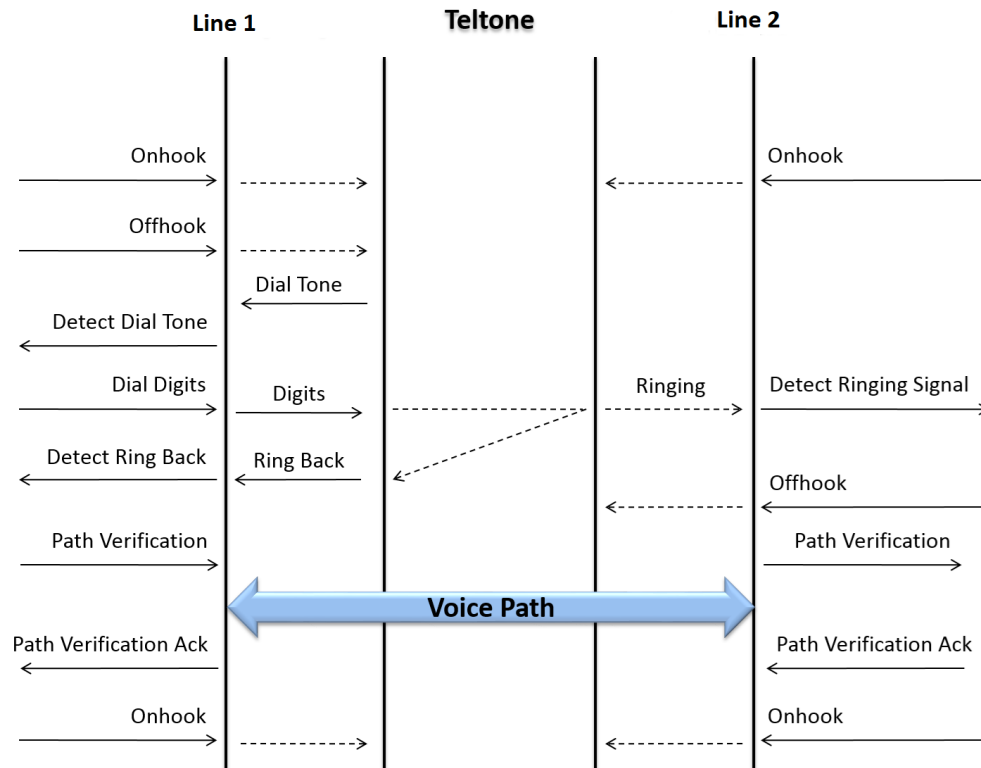
The MAPS™ APS server connects to the APSCB24/48 via a Quad T1 card and digital calls are generated to the APSCB24/48 using the MAPS™ CAS CLI or GUI. The APSCB24/48 converts the digital calls to analog and emulates standard analog telephones. Each digital call signals Analog off hook with DTMF dialed digits from the APSCB24/48. Both outgoing and incoming analog calls are supported. The connection to the APSCB24/48 includes a standard 50 pin Amphenol telco connector, which can be connected to a breakout box or patch panels that supports up to 24 2-wire interfaces. Each of the analog ports serves as an independent analog telephone, which can be individually configured and activated for various telephony tasks. The calls can be monitored and recorded for voice quality analysis.

Thus, MAPS™ APS uses a Quad T1 card and two sets of APSCB48 and patch panels to achieve a max of 96 analog ports. More can be achieved by simply scaling the system with a 4U MAPS™ CAS server sporting 2 Octal T1 Cards which can support up to 384 analog ports. Refer to the functional diagram of MAPS™ APS.

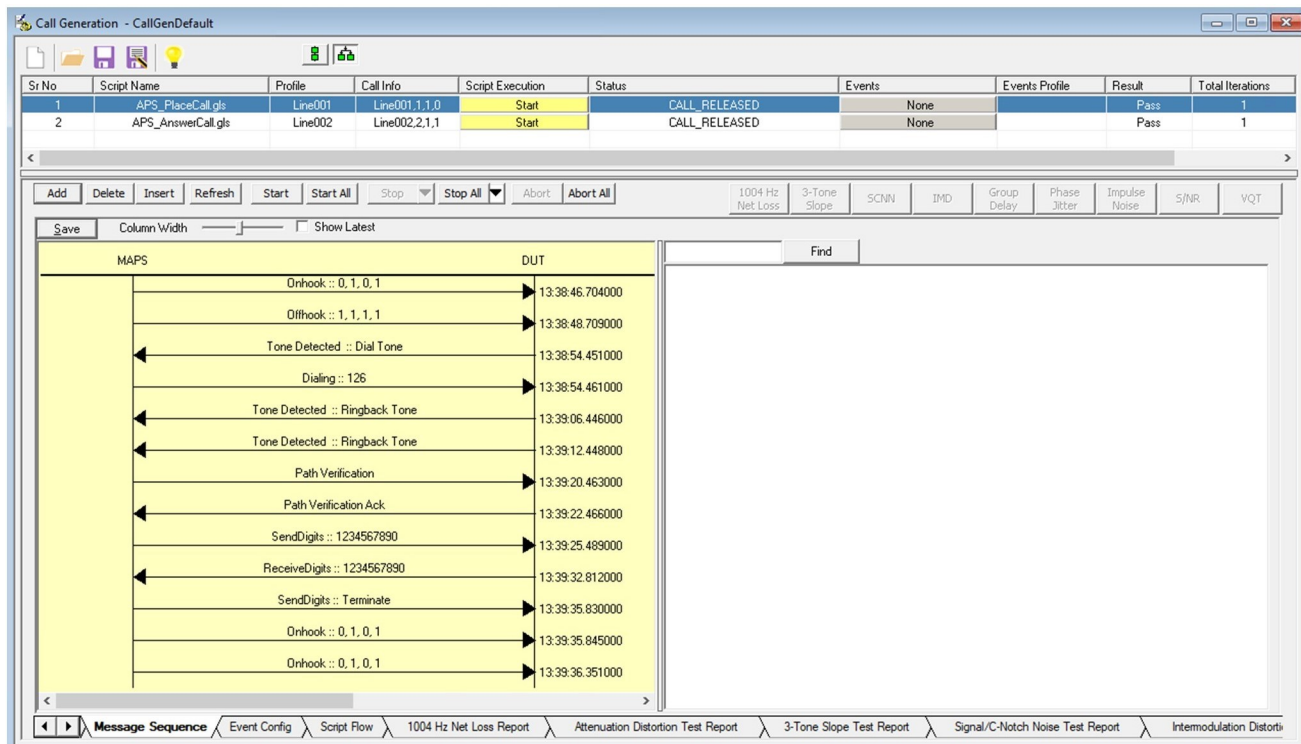


Loop Start signaling

Loop start is an analog signaling technique in which a calling party receives a dial tone on closing DC current loop. After detecting the dialed tone, the calling party will dial the digits and Central Office (CO) sends Ringback tone. The Ringback tone is detected at the calling party and then the call gets connected. Onhook is send to disconnect the call. The call scenario is as depicted in the below screenshot. The Line 1 and Line 2 are connected are connected to the Central Office under test.

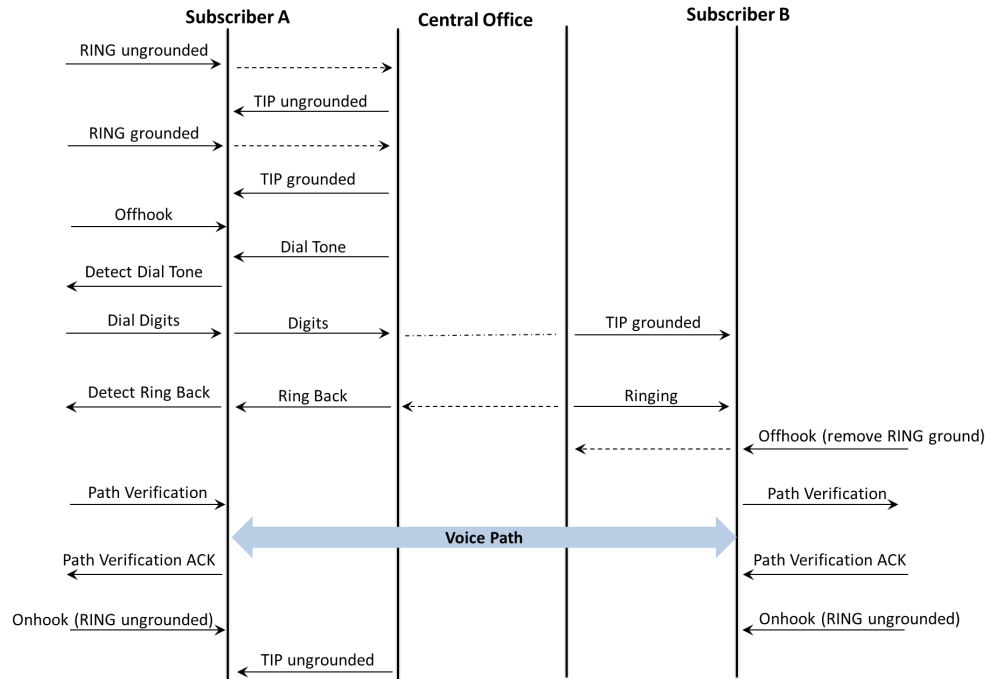


Refer to the below screenshot for Loop Start signaling call emulation.



Ground Start signaling

Ground Start signaling is an analog signaling technique typically used between PBX and Central Office. Initially TIP and RING both the leads are ungrounded, which indicates the Idle state. The RING lead changes to ground state to request service from Central Office i.e., Seize. When Central Office senses the RING ground, the TIP lead changes to ground state. When PBX (FXO) module senses the TIP grounded, it closes the loop, removes the RING ground, and goes in offhook state. FXS module sends dial tone, and the call continues as in loop start. FXO Subscriber A and Subscriber B are connected to the Central Office under test is as depicted in the below screenshot.



Refer to the below screenshot for **Ground Start signaling call emulation**.

The screenshot shows the MAPS (Message Automation Protocol Simulation) FXO (APS) - [Call Generation - CallGenDefault] interface. The interface includes a menu bar (Configurations, Emulator, Reports, Editor, Debug Tools, Windows, Help) and a toolbar with various icons. Below the toolbar is a table showing the call execution status for two scripts: APS_PlaceCall.gls and APS_AnswerCall.gls. The table has columns for Sr No, Script Name, Profile, Call Info, Script Execution, Status, Events, Events Profile, Result, Total Iterations, and Completed Iteration. The APS_PlaceCall.gls script is in the 'Stop' state, and the APS_AnswerCall.gls script is in the 'Ready for Voiceband Measurement' state.

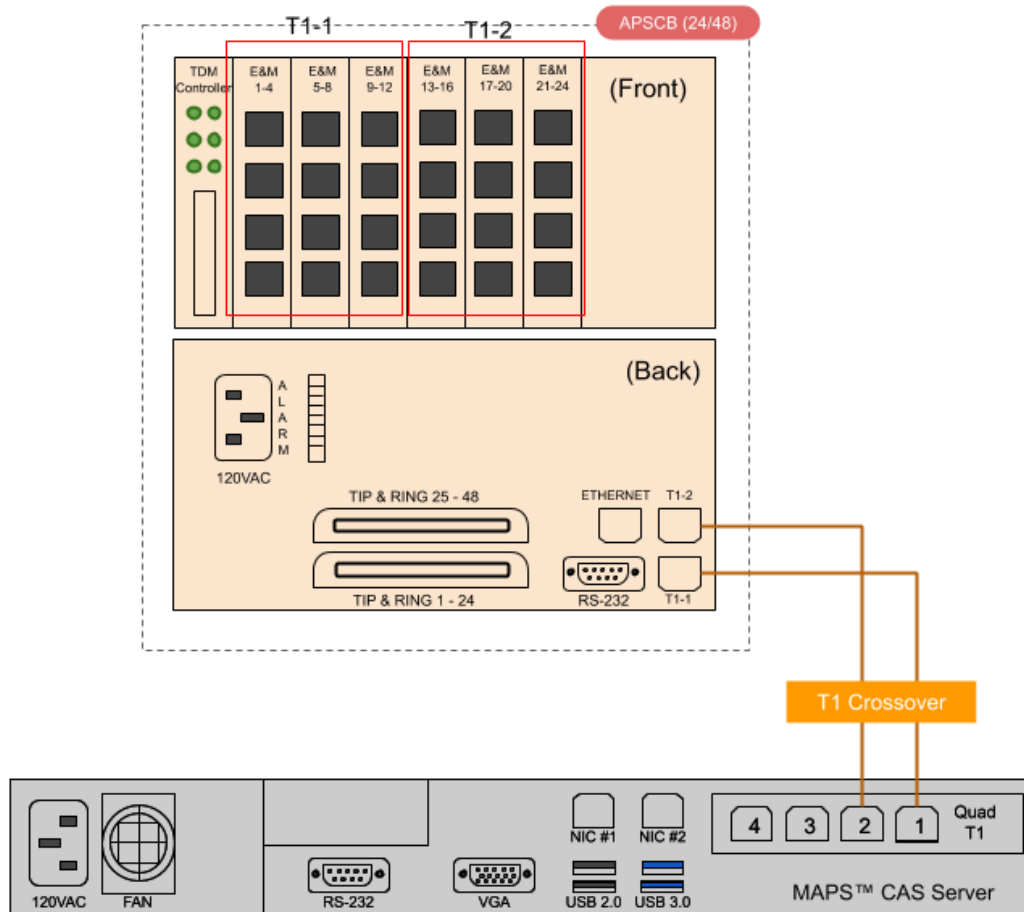
Below the table is a section for adding, deleting, inserting, refreshing, starting, and stopping the call. The 'Start' button is highlighted. Below this is a section for saving, column width, and showing the latest. The main area displays a sequence of events for the call, including Onhook (RING ungrounded), TIP ungrounded, RING grounded, TIP grounded, Offhook, Tone Detected (Dial Tone), Dialing (126), Tone Detected (Ringback Tone), Path Verification, and Path Verification Ack. The events are listed with timestamps and a 'Find' button is available.

On the right side, there is a 'Find' button and a text area showing the state of the call: 'State :: Call Connected', 'Signaling Bits :: 1, 1, 1, 1', and 'Receiving On Card :: 1 Timeslot :: 0'.

At the bottom, there is a navigation bar with tabs for Message Sequence, Event Config, Script Flow, 1004 Hz Net Loss Report, Attenuation Distortion Test Report, 3-Tone Slope Test Report, Signal/C-Notch Noise Test Report, and Intermodulation Distortion Test Report.

MAPS™ APS E&M

A channel bank is required to provide analog 4-wire E&M interfaces. Without a channel bank, MAPS APS will perform the signaling required to emulate E&M signaling. The connection diagram below shows an Aditi 600 channel bank with E&M service cards.



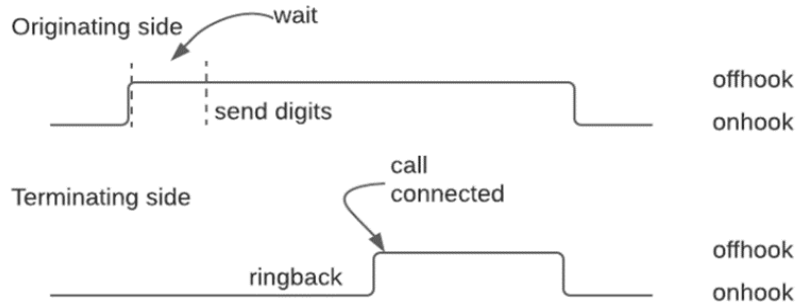
Each E&M service card provides four female RJ-45 connectors. Each T1 supports 3 E&M service cards for a total of 12 E&M interfaces.

Normal Mode (Terminating/Channel Equipment)		Tandem Mode (Originating/Switching Equipment)	
Pin	Name	Pin	Name
1	SG (Signal Ground)	1	SG (Signal Battery)
2	E	2	M
3	R1	3	R1
4	R	4	R
5	T	5	T
6	T1	6	T1
7	M	7	E
8	SB (Signal Battery)	8	SG (Signal Battery)

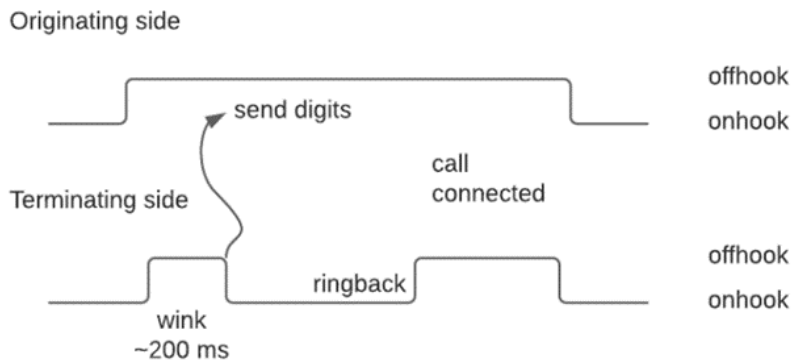
MAPS™ APS E&M (Contd.)

E&M Signaling Types I, II, III, IV, and V are supported by the E&M service card. Dial Supervision Signaling: Three types of start dial supervision signaling are supported and can be selected via the E&M Profiles in MAPS APS.

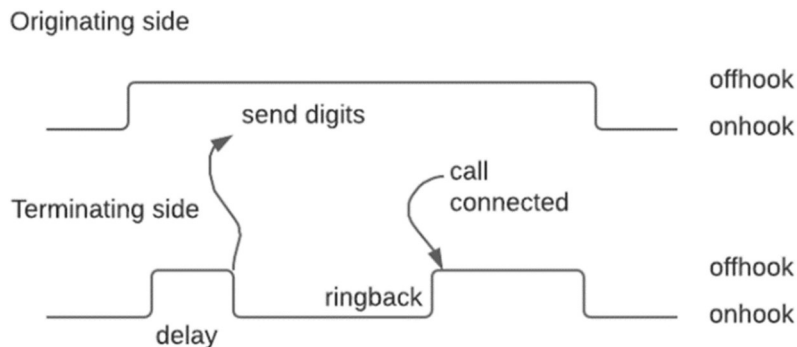
Immediate Start: The originating side goes offhook, waits for a finite period then sends the dial digits to the far end



Wink Start: The originating side goes offhook, waits for a temporary offhook pulse from the other end, then sends the dial digits



Delay Dial: The originating side goes offhook and waits for about 200 ms, then checks to see if the far end is onhook. If the far end is onhook then it sends the dial digits. If the far end is offhook, it waits until it goes onhook then sends the dial digits.



MAPS™ APS E&M Call Emulation

In call generation, MAPS™ is configured for the out going messages, while in call receive mode, it is configured to respond to incoming messages. Tests can be configured to run once, multiple iterations and continuously. Also, allows users to create multiple entries using quick configuration feature.

The editor allows to run the added scripts sequentially (order in which the scripts are added in the window) or randomly (any script from the list of added script as per the call flow requirements). The test scripts may be started manually or they can be automatically triggered by incoming messages. The below image shows the emulation of E&M Originating call.

The screenshot displays the 'Call Generation - Default' window. At the top, there is a table showing the status of two test scripts:

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile	Result	Total Iteratio...	Completed Iterations
1	E&M_Originating.gls	Line001	Line001,1,1,0	Stop	Call Connected	OutboundRelease...		Pass	1	0
2	E&M_Terminating.gls	Line013	Line013,25,2,0	Stop	Call Connected	InboundReleaseCall		Pass	1	0

Below the table is a toolbar with buttons: Add, Delete, Insert, Refresh, Start, Start All, Stop, Stop All, Abort, Abort All. To the right are checkboxes for various test parameters: Impulse Noise, IMD, 1004 Hz Tone, CNN, S/N, VQT, Group Delay, Phase Jitter, 3-Tone Slope.

The main area shows a 'Message Sequence' diagram. It is divided into two columns: 'MAPS' and 'DUT'. The sequence of events is as follows:

- Onhook :: 0, 0, 0, 0 (from MAPS to DUT) at 16:22:28.983000
- Onhook :: 0, 0, 0, 0 (from DUT to MAPS) at 16:22:29.063000
- Offhook :: 1, 1, 1, 1 (from MAPS to DUT) at 16:22:29.343000
- ReceiveDigits :: 102 (from DUT to MAPS) at 16:22:34.024000
- Ringback Tone (from MAPS to DUT) at 16:22:34.024000
- Offhook :: 1, 1, 1, 1 (from MAPS to DUT) at 16:22:47.064000

At the bottom, there is a navigation bar with tabs: Message Sequence (selected), Event Config, Script Flow, 1004 Hz Net Loss Report, Attenuation Distortion Test Report, 3-Tone Slope Test Report, and Signal/C-Notch Noise Test Report.

Buyer's Guide

Item No	Product Description
XX651	MAPS™ APS Emulation (requires Channel Bank for Bulk Calls)
XXFT0	Fax Emulation for OctalExpress T1 Analyzer
XXFT4	30 Port T1 Fax Licenses for OctalExpress T1 Analyzer
XX610	Transmit and Receive File Capability
XX620	Transmit/Detect digits
PKS170	CLI Support for MAPS™
VQT002	Voice Quality Testing - PESQ only

Item No	Related Hardware
DP005B	Rackmount PC, 19", Xeon - Standard
DP005B	Rackmount PC, 19", Xeon - Standard – VQT Analysis (Optional)
APSCB-48	48 Port Analog Phone Simulator (APS), 115V
APSCB-24	24 Port Analog Phone Simulator (APS)
2W APS– FXO or FXS or Combination	
APSCB-24-2W	24-Port 2W Analog Phone Simulator (APS), or Central Office Switch Simulator
APSCB-48-2W	48-Port 2W Analog Phone Simulator (APS), or Central Office Switch Simulator
APSCB-72-2W	72-Port 2W Analog Phone Simulator (APS), or Central Office Switch Simulator
APSCB-96-2W	96-Port 2W Analog Phone Simulator (APS), or Central Office Switch Simulator
APSCB-120-2W	120-Port 2W Analog Phone Simulator (APS), or Central Office Switch Simulator
FTE001	QuadXpress T1E1 Main Board (Quad Port– requires additional licenses)
4W APS– FXO or FXS or Combination	
APSCB-24-4W	24-Port 4W Analog Phone Simulator (APS), or Central Office Switch Simulator
APSCB-48-4W	48-Port 4W Analog Phone Simulator (APS), or Central Office Switch Simulator
APSCB-72-4W	72-Port 4W Analog Phone Simulator (APS), or Central Office Switch Simulator
APSCB-96-4W	96-Port 4W Analog Phone Simulator (APS), or Central Office Switch Simulator
ETE001	OctalXpress T1E1 Main Board plus Daughter Board (Octal Port– requires additional licenses)
ETA008	Eight Port License for T1
PTE001	tProbe™ T1 E1 Base Unit
XTE001	Dual T1 E1 Express (PCle) Boards (requires additional licenses)

Note: PCs which include GL hardware/software require Intel or AMD processors for compliance.

For more information, refer to [2Wire Analog and 4Wire E and M Bulk Call Generator](#) webpage.



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