Overview

Recent advances in Air Traffic Management over IP network has opened up both opportunities for providing better services, and challenges to ensure reliability, and performance. Among many other solutions for testing Air Traffic Control network, GL offers MAPS™ ED137 Telephone to simulate the functions of Controller Working Position (CWP) in Ground-to-Ground telephone calls it can simulate both Air-Ground calls as per ED137_1B: Radio and calls as per EUROCAE (European Organization for Civil Aviation Equipment) ED137 standards. The software not only provides complete control over call scenarios to be tested, but also the ability to customize the network parameters for signalling and VoIP traffic. It has the capability of generating more than 500 simultaneous calls on a core i7 systems.

MAPS™ also supports transmission and detection of various RTP audio traffic such as real-time audio, voice file, digits, single tone and dual tones.


Main Features

- Emulates Telephone interface at CWP endpoints as per ED-137/2B Compliance
- Emulates Ground-to-Ground Calls
- Supports all SIP Methods, Headers and Mandatory/Optional SDP attributes as per ED-137/1B
- Supported call types include –
  - Instantaneous Access (IA)
  - Priority Direct/Indirect Access (DA/IDA)
  - Routine Tactical Direct/Indirect Access
  - Routine Strategic Direct/Indirect Access
  - Routine General Purpose Direct/Indirect Access
  - Position Monitoring Call (Combined A/G and G/G, A/G only, and G/G only)
- Supports Apply Events on an ongoing call - Transfer Call, ReInvite, Receive Traffic, active call on Hold, Send Traffic, Impair traffic, play back call on Speaker
### Main Features

#### Signaling and Traffic Simulation
- Fully integrated, complete test environment for Air Traffic Management
- Supports testing CWP, VCS, GRS (or RMG), and VRS elements
- Supports hundreds of simultaneous calls and complete automation of bulk call generation with traffic
- IP Address Spoofing to automatically generate virtual IP address for a NIC
- Send and receive live speech, pre-recorded speech files, digits and tones.
- Supports all standard codecs, including G.711 (mu-Law and A-Law) and G.729
- Set impairments (Packet Loss, Packet Effects and Latency) in relevant profile in real time.
- Supports transport over IPv4, IPv6, UDP and TCP.
- Depicts easy to understand call flow graphs of SIP message exchanges and message contents (SIP headers and SDP attributes)
- Supports User authentication with Proxy, Registrar servers
- Handles strict routing & loose routing, when requests are routed through proxies
- Allows call rejection through use of SIP response codes (4xx, 5xx, 6xx)
- Supports multiple Profiles (Users/End points) from single node
- Allows to define DSCP (Differentiated Service Code Point) values for signalling and voice traffic
- Supports user-defined and automated traffic actions on the call
- Provides aggregated voice quality statistics such as MOS/R-Factor, packet loss, duplicate and out of sequence packets
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#### GUI Features
- Portable, easy to configure and use during in-the-field installation, system configuration/ test and commissioning
- Provides call statistics such as Active, Completed, Passed, Failed and Calls per second
- Provides Event logs, Captured errors and Error events.
- Automation, Remote access, and Schedulers to run tests 24/7
- Supported on Windows® 7, 8 or higher version operating systems
- Supports 64-bit version to enhance signalling performance

#### CLI
- Supports client-server functionality requires additional license; clients supported are TCL, Python, VBScript, Java, and .Net

#### Applications
- On field testing and troubleshooting by technicians
- In-the-field installation, system configuration and commissioning
- Functionality testing of nodes in next generation VoIP ATM
- Load testing and background traffic generation
- QoS monitoring - analyze calls for voice quality (MOS), packet loss, jitter, latency, etc.
- Centralized monitoring of Air Traffic ; real-time and/or historical data analysis with PacketScan™ and NetSurveyorWeb™
Configuration Scenarios

Scenario 1: MAPS™ acting as CWP1 to test another CWP
MAPS™ ED137 can be configured to generate/receive Telephone calls to another CWP.

Scenario 2: MAPS™ acting as CWP2 to test another CWP
MAPS™ ED137 can be configured to receive Telephone calls (DA/IDA) from another CWP.

Call Generation and Reception

In call generation, MAPS™ is configured for the outgoing 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 message flow between the configured entities are displayed in sequence. The message decodes for any particular selected message in the flow is also displayed, refer to the image below.

Once call is established between the two terminals messages are exchanged between the terminals.

The Event options on the window allows users to manually start/stop traffic, impair the traffic, transfer call, and playback the call using Speaker On option.

CWP Telephone Call Profiles

This feature allows loading profile to edit the values of the variables using GUI, replacing the original value of the variables in the message template. An XML file defines a set of multiple profiles with varying parameter values that allow users to configure call instances in call generation and to receive calls.

Unlimited number of user profiles can be created with call control and traffic parameters where each profile can simulate a CWP Telephone call.

Supported call types includes -
- Instantaneous Access, Priority Direct/Indirect Access,
- Routine Tactical Direct/Indirect Access,
- Routine Strategic Direct/Indirect Access,
- Routine General Purpose Direct/Indirect Access,
- Position Monitoring (Combined A/G and G/G, A/G only, and G/G only) Call.

IP spoofing feature allows to create multiple CWPs to be simulated using unique IP address from a single system.

Figure: MAPS™ CWP Simulating Telephone DA/IDA Call

Figure: CWP Telephone Call Profiles
**RTP Statistics Calculation**

MAPS™ ED137 Telephone provides global voice quality statistics on RTP, which includes metrics such as Listening MOS, Conversational MOS, Packet Loss, Discarded Packets, Out of Sequence Packets, Duplicate Packets, Delay and Jitter. These statistics are calculated and updated periodically on run time.

**Buyer’s Guide**

- **PKS119** - MAPS™ ED137 Telephone (includes PKS102)
- **PKS118** - MAPS™ ED137 Radio (includes PKS107, & PKS102)

**Related Software**

- **PKS102** - RTP Soft Core for RTP Traffic Generation
- **PKS107** - RTP EUROCAE ED137
- **PKS120** - MAPS™ SIP Emulator
- **PKS121** - MAPS™ SIP Conformance Test Suite (Test Scripts)
- **PKS126** - MAPS™ SIP I Emulator
- **PKS127** - MAPS™ SIP - IMS
- **PKS130** - MAPS™ SIGTRAN Emulator
- **PKS122** - MAPS™ MEGACO Emulator
- **PKS123** - MAPS™ MEGACO Conformance Test Suite (Test Scripts)
- **PKS124** - MAPS™ MGCP Protocol Emulation with Conformance Test Suite
- **PKV100** - PacketScan™ (Online and Offline)
- **XX170** - Network Surveillance Software with Centralized Database Engine and Client