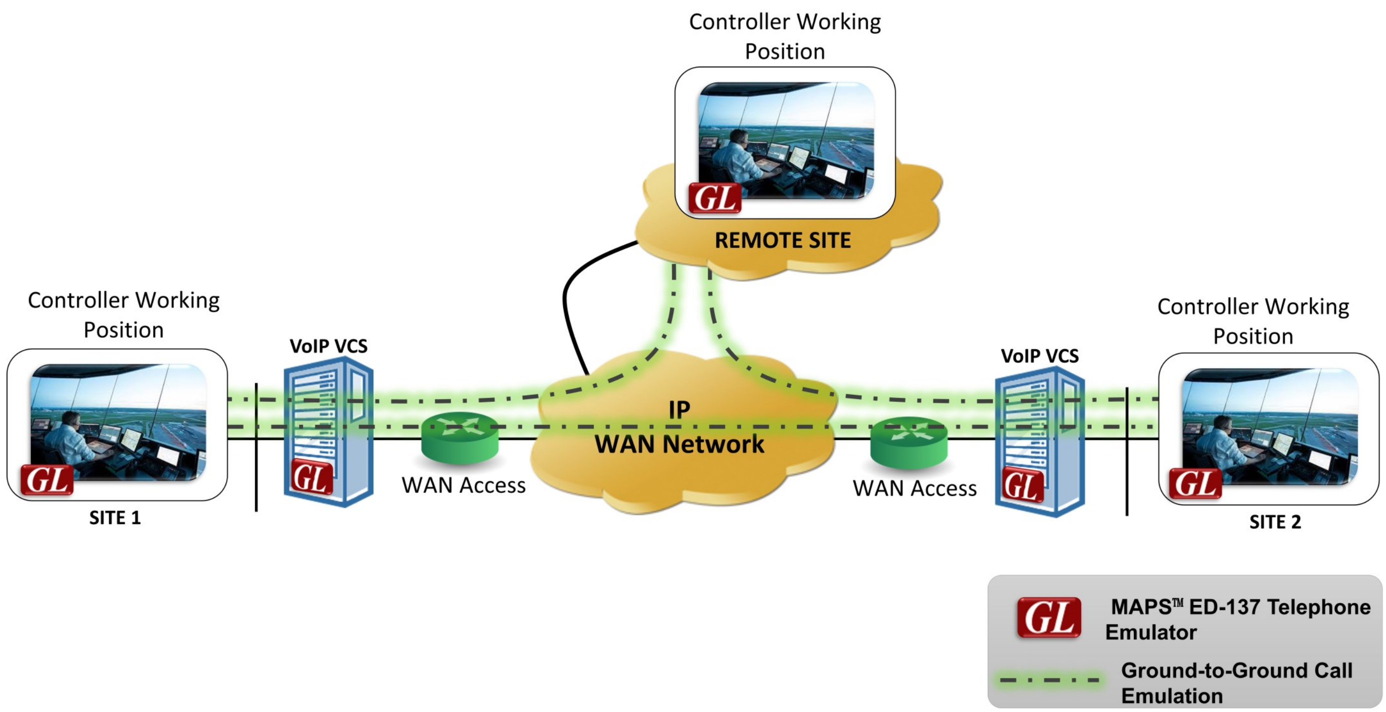
MAPS™ ED137 Telephone Emulator

Ground-to-Ground Calls Emulation



Overview

GL's **MAPS™ ED137 Telephone (PKS119)** can emulate Telephone interfaces for **Ground-to-Ground** calls at Controller Working Position (CWP) endpoints as per **ED-137/2B** and **ED-137/2C** versions, as defined under EUROCAE (European Organization for Civil Aviation Equipment) standards. The software not only provides complete control over call scenarios to be tested, but also the ability to customize the network parameters for signaling and VoIP traffic. It has the capability of generating more than 500 simultaneous calls on a core i7 systems.

The product supports transmission and detection of various RTP audio traffic such as real-time audio, voice file, digits, single tone and dual tones. The MAPS™ profile editor feature allows users to easily configure multiple CWPs profiles, allowing to emulate multiple CWP entities.

GL tools for signaling emulation and voice quality testing offer an end-to-end test solution for testing connections from the radio interfaces to the CWP and recording voice communications in the ATM network. GL's Air Traffic Management Solution also includes MAPS™ ED137 Recorder Emulators and MAPS™ ED137 Radio Emulators.

MAPS™ ED137 Telephone supports below addendums optionally.

- Addendum 2: FAA Legacy Telephone Interworking
- Addendum 4: Override Call
- Addendum 5: Voice Call

Application is also enhanced to support additional call scenarios such as Call Pick-up, Preset Conference call and Broadcast conference call.

For more information, refer to MAPS™ ED137 Telephone Emulator webpage.

GL Communications Inc.
818 West Diamond Avenue - Third Floor, Gaithersburg, MD 20878, U.S.A
(Web) www.gl.com - (V) +1-301-670-4784 (F) +1-301-670-9187 - (E-Mail) info@gl.com

Document Number: PKS119-01

Main Features

- Supports **ED-137 2B/ ED137 2C** and **Change 1** standards
- Validated against latest **VOTER** version
- Supports calls over **IPV4, IPv6, TCP** and **UDP**
- Supports all SIP Methods, Headers and Mandatory /Optional SDP attributes as per ED-137/2B and ED-137/2C
- Supported call types for ED-137/2B and ED-137/2C includes –
  - 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)
  - Meet Me
- Supports additional call scenarios such as Call Hold, Attended Call Transfer, Unattended Call Transfer, Call Pick-Up, Preset Conference, Broadcast Conference, Call Intrusion and more.

Signaling and Traffic Emulation

- Enhanced to support multiple CWP endpoints emulation
- Portable, easy to configure and use during in-the-field installation, system configuration/ test and commissioning
- Supported call types include Instantaneous Access, Priority Direct/Indirect Access, Routine Tactical Direct/Indirect Access, Routine Strategic Direct/Indirect Access, Routine General Purpose Direct/Indirect Access, and Position Monitoring (Combined A/G and G/G, A/G only, and G/G only) call
- Depicts easy to understand Call Flow Graphs of SIP message exchanges and Displays Message contents (SIP headers and SDP attributes)
- Allows call rejection through use of SIP response codes (4xx, 5xx, 6xx)
- Supports multiple Profiles (Users/End points) from single node
- Supports hundreds of simultaneous calls and load generation can be automated completely along with traffic
- Allows to define DSCP (Differentiated Service Code Point) values for signaling and voice traffic
- Supports complete customization of SDP and SIP headers, call flow, and messages
- Supports both UDP and TCP (IPv4 and IPv6)
- Handles Re-transmissions of messages with specific interval
- Supports User authentication with Proxy, Registrar servers
- Supports IP address spoofing for each endpoint to generate call using different IP address from a single system
- Supports OPTIONS PING feature used to check the connection status

Traffic

- Supports various traffic actions on the call such as Playback to Speaker, Send and Record audio file, Generate and detect inband digits, single tone and dual tone
- Supports ED-137 defined codecs - G.711 (mu-Law and A-Law) and G.729
- Supports User-defined and automated traffic actions on the call
- Applies impairments to the traffic such as Packet Loss, Latency, Duplicate and Out of sequence
- Provides aggregated voice quality statistics such as MOS/R-Factor, Packet Loss, Duplicate and out of sequence packets

Command Line Interface (CLI)

- Supports client-server functionality requires additional license; supports Python APIs

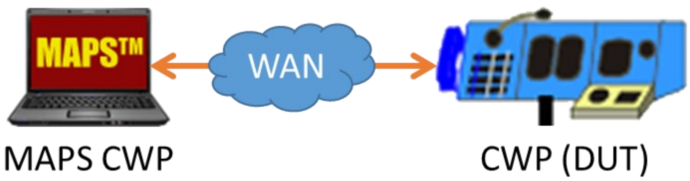
Applications

- 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

MAPS™ ED137 Telephone User Cases

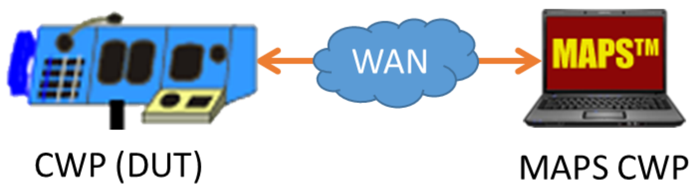
Scenario 1: MAPS™ acting as CWP1 to test another CWP

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.



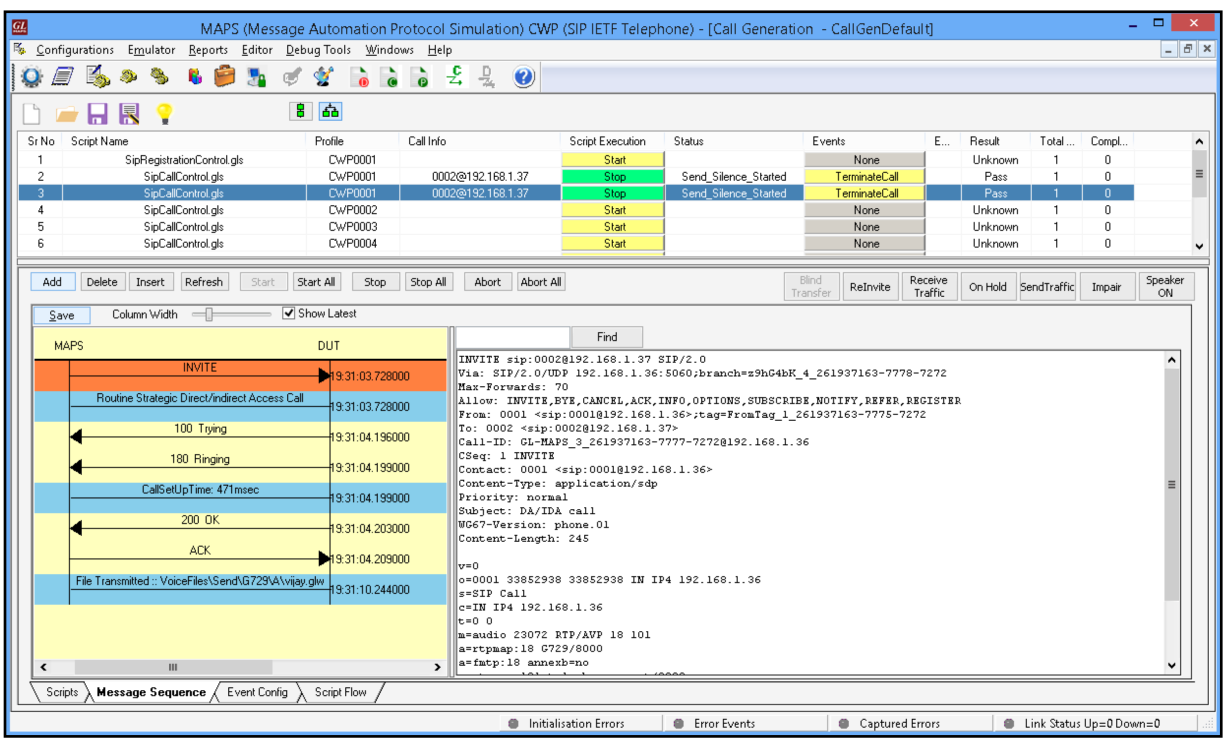
Call Generation and Reception

In call generation, MAPS™ is configured for the 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 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.



MAPS™ CWP Emulating Telephone DA/IDA Call

GL Communications Inc.

CWP Telephone Call Profiles

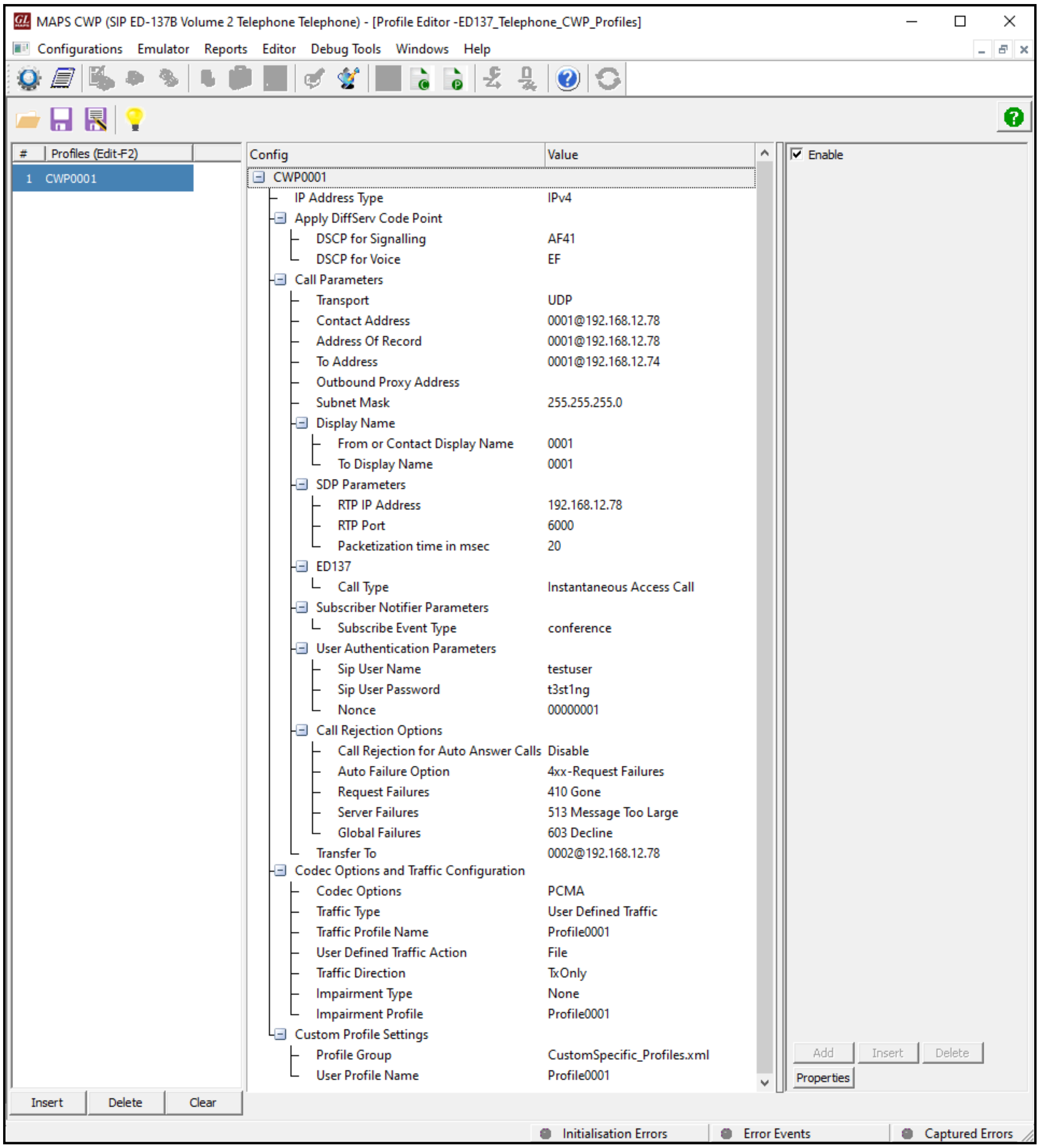
This feature allows loading profile to edit the values of the variables using GUI, replacing the original value of the variables in the message template. An XML file defines a set of multiple profiles with varying parameter values that allow users to configure call instances in call generation and to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

Unlimited number of user profiles can be created with call control and traffic parameters where each profile can emulate 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 emulated using unique IP address from a single system.



CWP Telephone Call Profiles

MAPS™ ED-137 Telephone Addendums

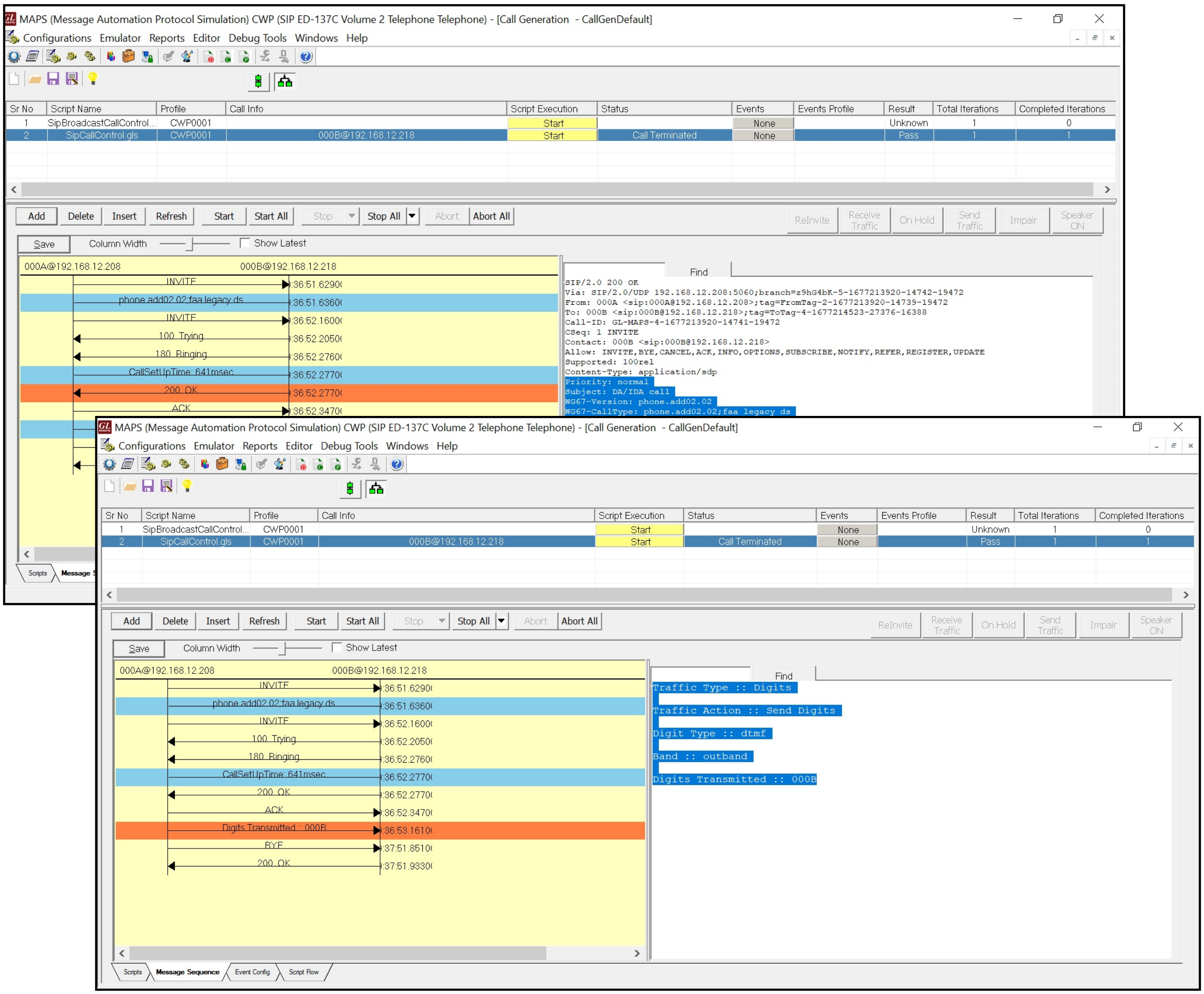
EUROCAE ED-137C Volume 2 specification describes basic requirements to establish, terminate, and modify speech media sessions of the Ground Telephone Service in an Air Traffic Services Ground Voice Network (AGVN). Along with the basic requirement, it also describes specific call functionalities through number of addendums. GL's MAPS ED-137 Telephone is now compliant with addendum 2, addendum 4 and addendum 5 specifications.Support for addendums are optional.

Addendum 2: FAA Legacy Telephone Interworking

Addendum 2 describes the provisions for the FAA Legacy Telephone Interworking, and translation between SIP endpoint(s) and an analog trunk interface to a legacy FAA VCS. The signaling translation between the SIP endpoints and the analog trunk is provided via a gateway. It details the interaction between the analog trunk and the SIP endpoint(s) for five FAA Legacy G/G call types. The supported call types are:

- Legacy Dial-Selective DA/IDA Call
- Legacy Selective Signaling DA/IDA Call
- Legacy Non-Selective DA/IDA Call
- Legacy Voice Call
- Legacy Override (OVR) Call

The below scenario shows emulation of FAA Legacy Dial-Selective DA/IDA call.



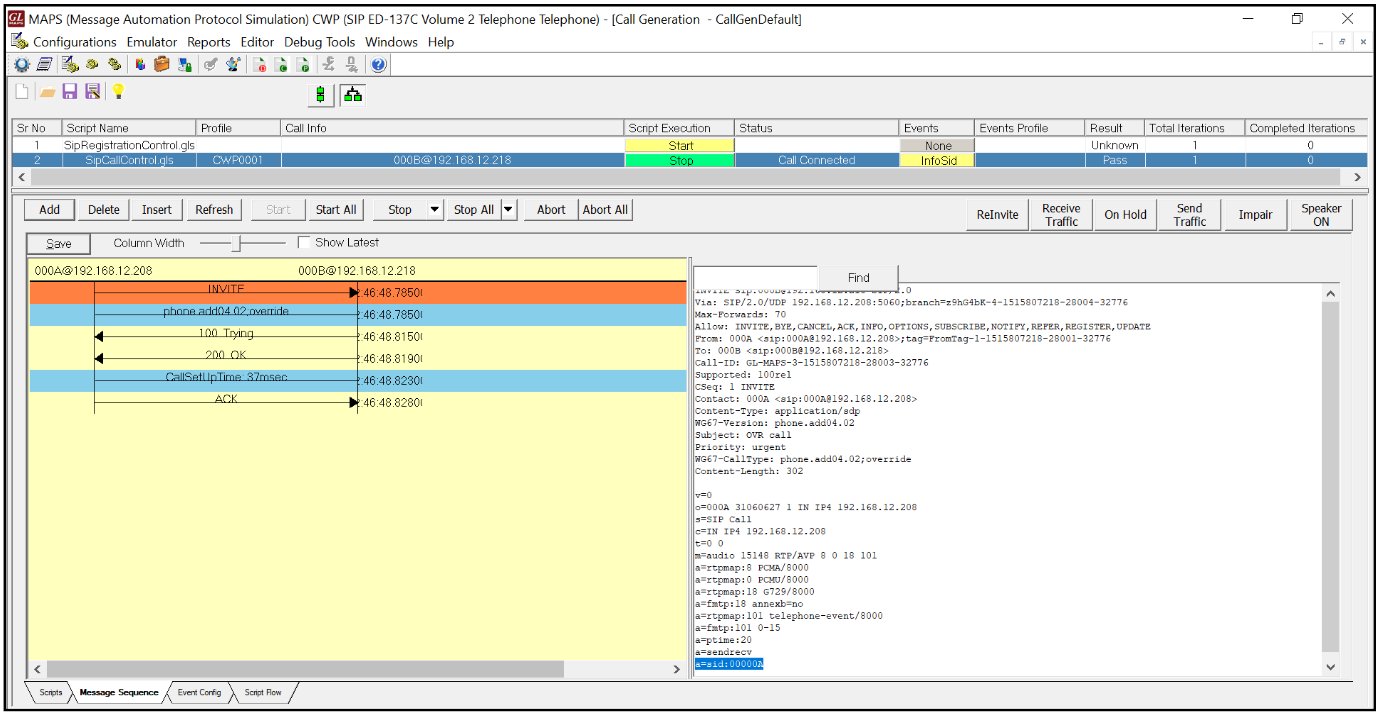
GL Communications Inc.

Addendum 4: Override Call

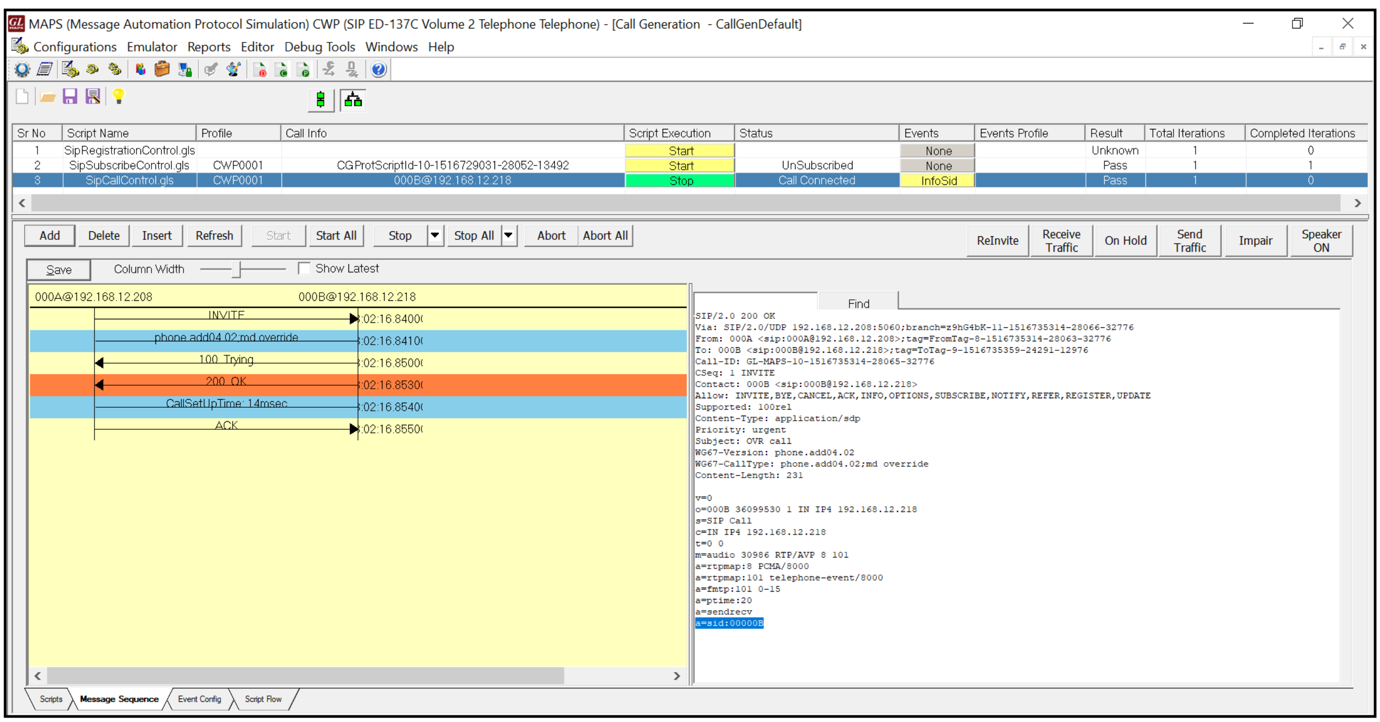
Addendum 4 describes the requirements for Override Calls. The Override Call Type is supplemental to the core ED-137C Volume 2 and meant to interact with other call types and features of the core volume; therefore, all applicable requirements established in the core ED-137C: Volume 2 apply to the Override Call Type. The Override Call Type will provide transmit and receive audio from various call types and as such, will create scenarios where Loop Closure and Echo are to be detected and mitigated. The supported call types are:

- Override Call
- MD Override Call

The below scenario shows call emulation of Override call.



The below scenario shows call emulation of MD Override call.



 GL Communications Inc.

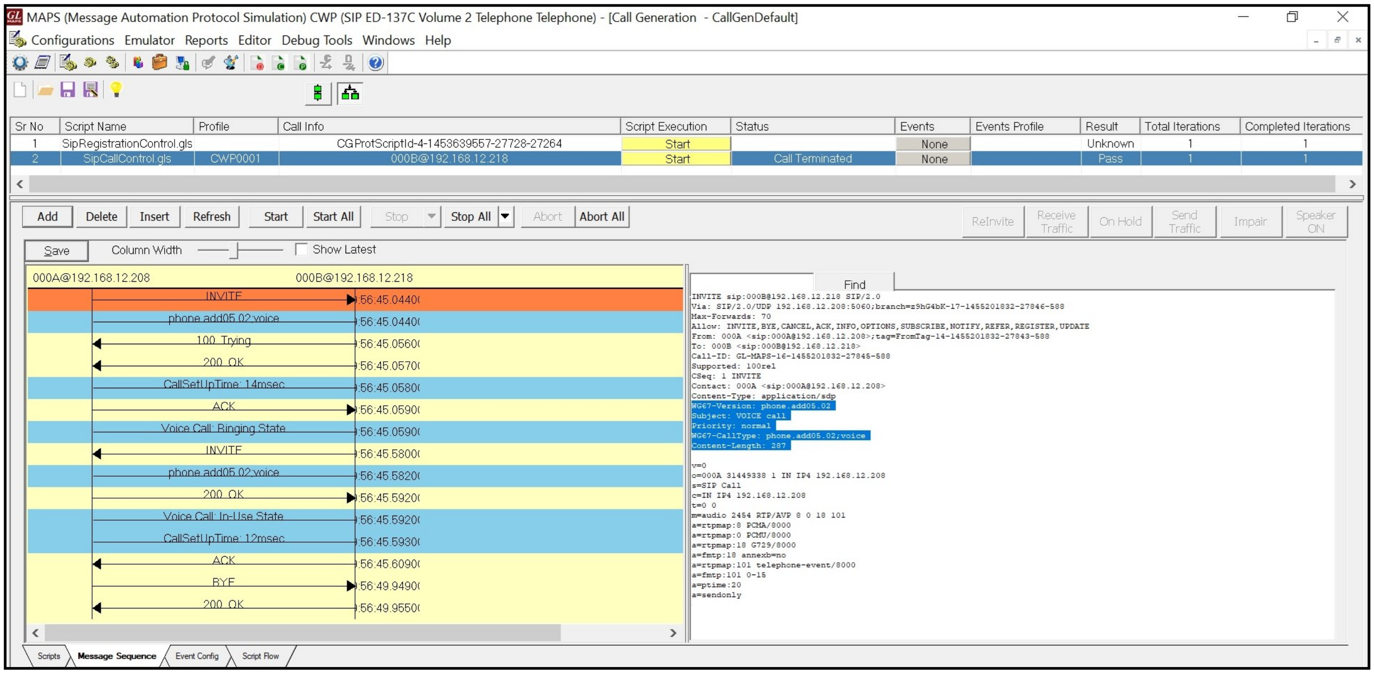
Addendum 5: Voice Call

The Voice call type is supplemental to the core ED-137C Volume 2; therefore, all requirements established in the core ED-137C: Volume 2 apply to the Voice call type unless otherwise specified or superseded. The Voice call type is intended to act as a modified DA/IDA call that forms a new call type to satisfy implementation specific requirements. The Voice call type is initiated as a DA/IDA call. However, upon initiation the Voice call provides an immediate Voice page from the calling to the called user, replacing ringing tones. The called party of a Voice call answers in the same way as a DA/IDA call and once active, the Voice call is maintained similarly as well.

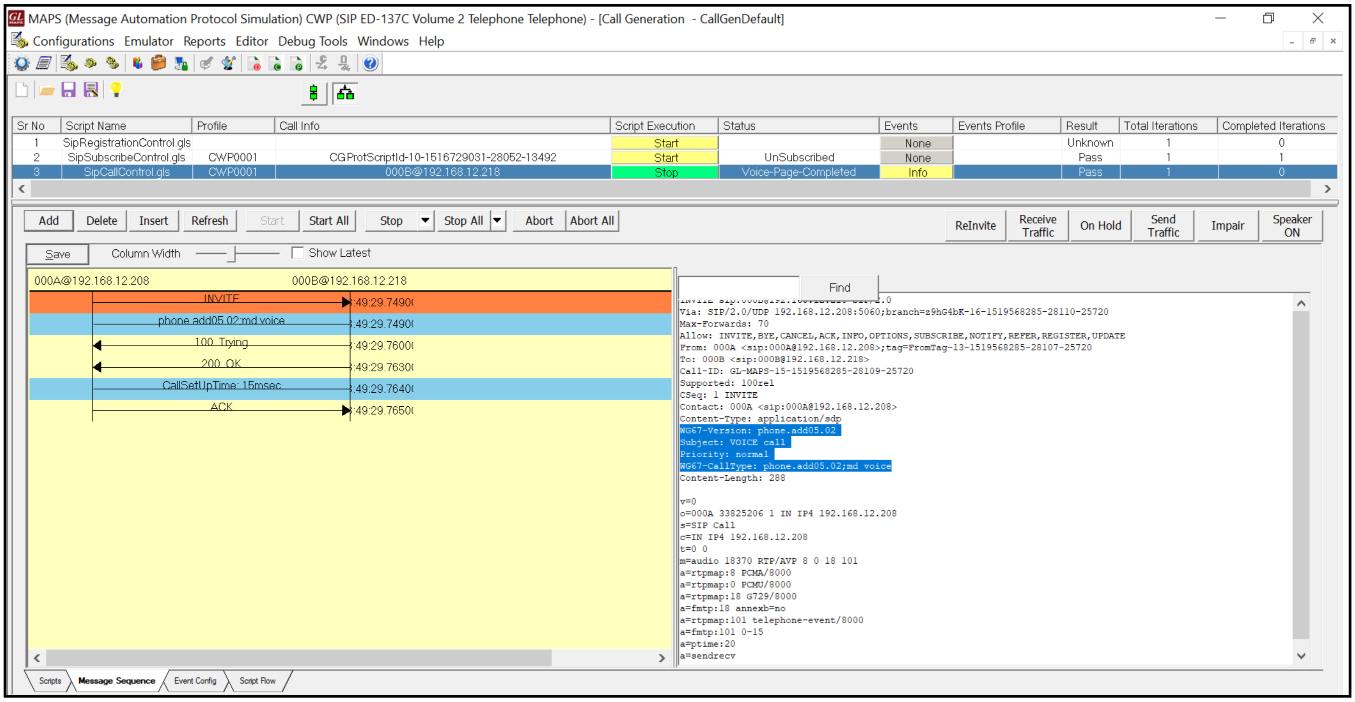
The supported call types are:

- SD Voice Call
- MD Voice Call

The below scenario shows call emulation of SD Voice call.



The below scenario shows call emulation of MD Voice call.



GL Communications Inc.

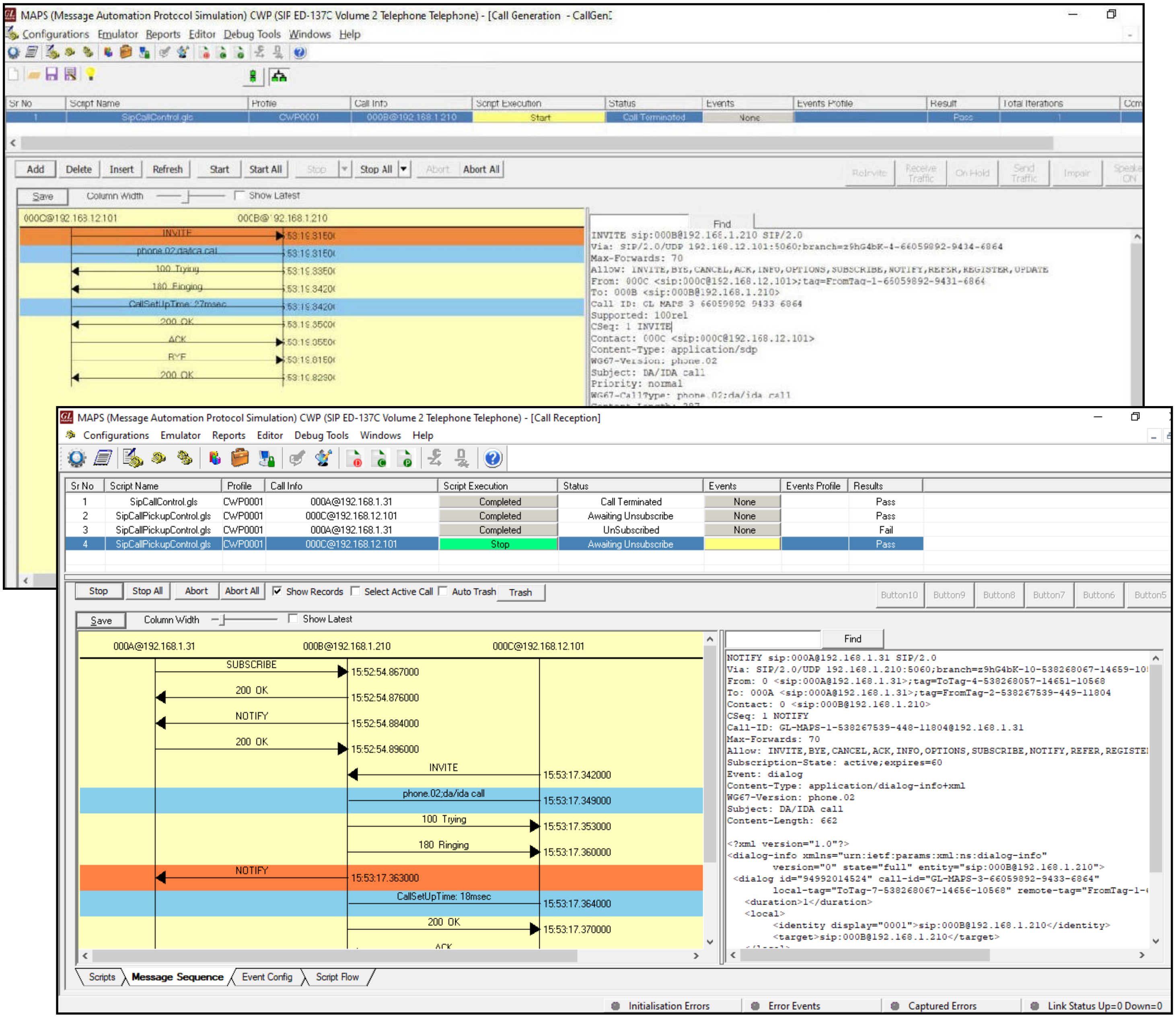
Additional Call Features

MAPS™ ED137 Telephone supports below listed additional call features:

- Call Pickup
- Preset Conference
- Broadcast Conference
- Call Intrusion

Call Pickup

The Call Pickup services enables a user not involved in an early dialog to answer calls on behalf of other parties and may apply to Routine Direct/Indirect Access Calls. The party wishing to pick up the call sends a SUBSCRIBE to the remote entity to retrieve the dialog information. In addition, the party wishing to pick up an early dialog sends an INVITE with a Replaces Header to the calling party.

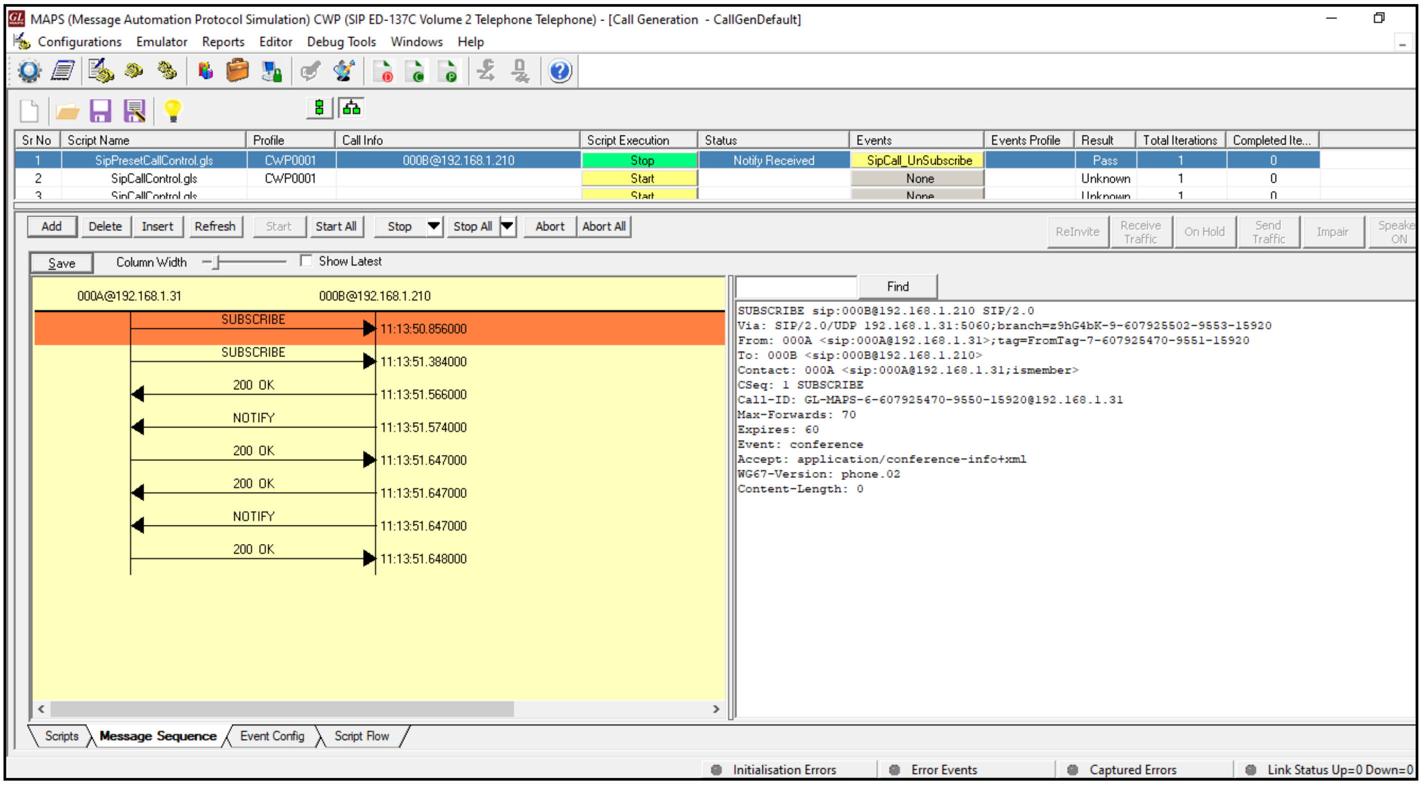


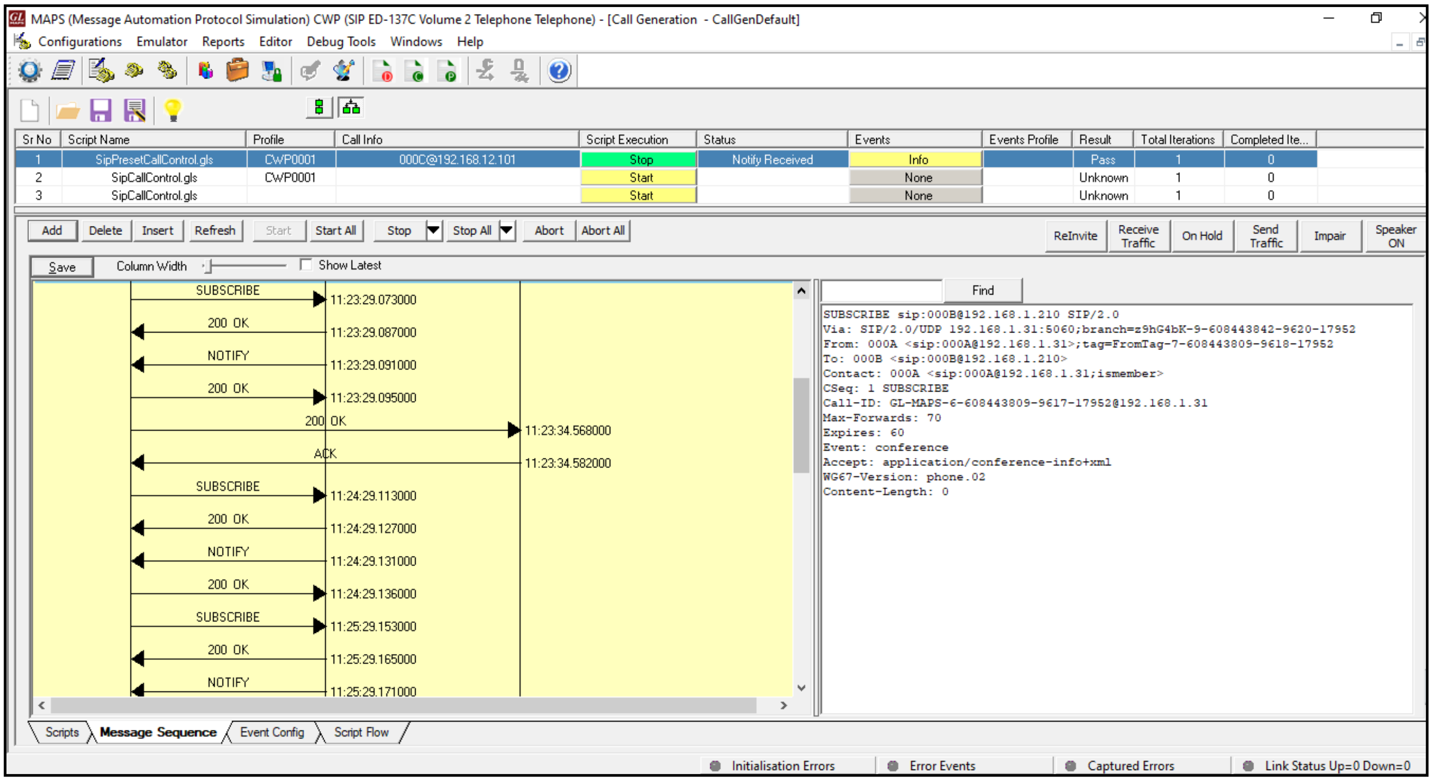
GL Communications Inc.

Additional Call Features

Preset Conference

A Preset Conference call is a Conference call type used to contact all other members assigned to a specific call. The initiating member of a Preset Conference call establishes contact with the call Focus with the intent of contacting the remaining members, via communication with the Focus. The dedicated script i.e. SipPresetCallControl.gls is used to emulate Preset Conference call.



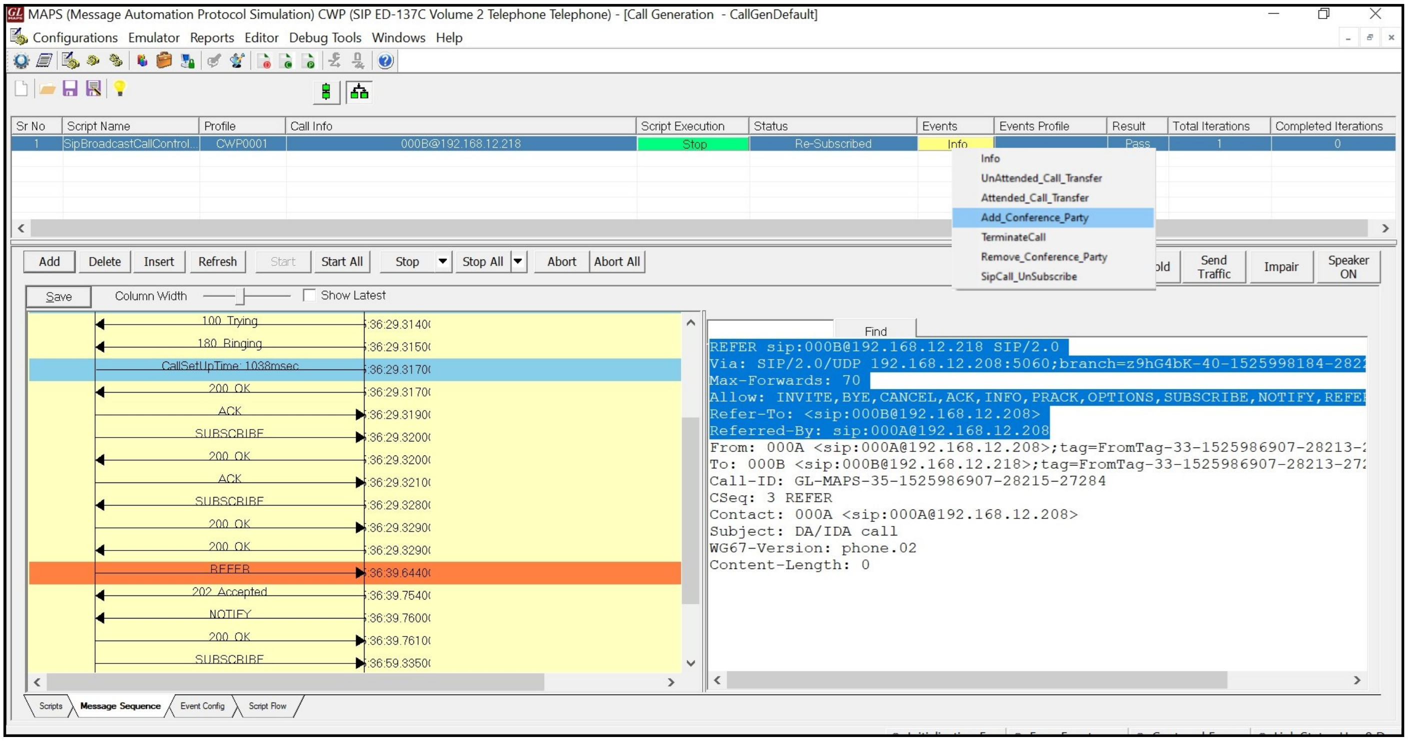


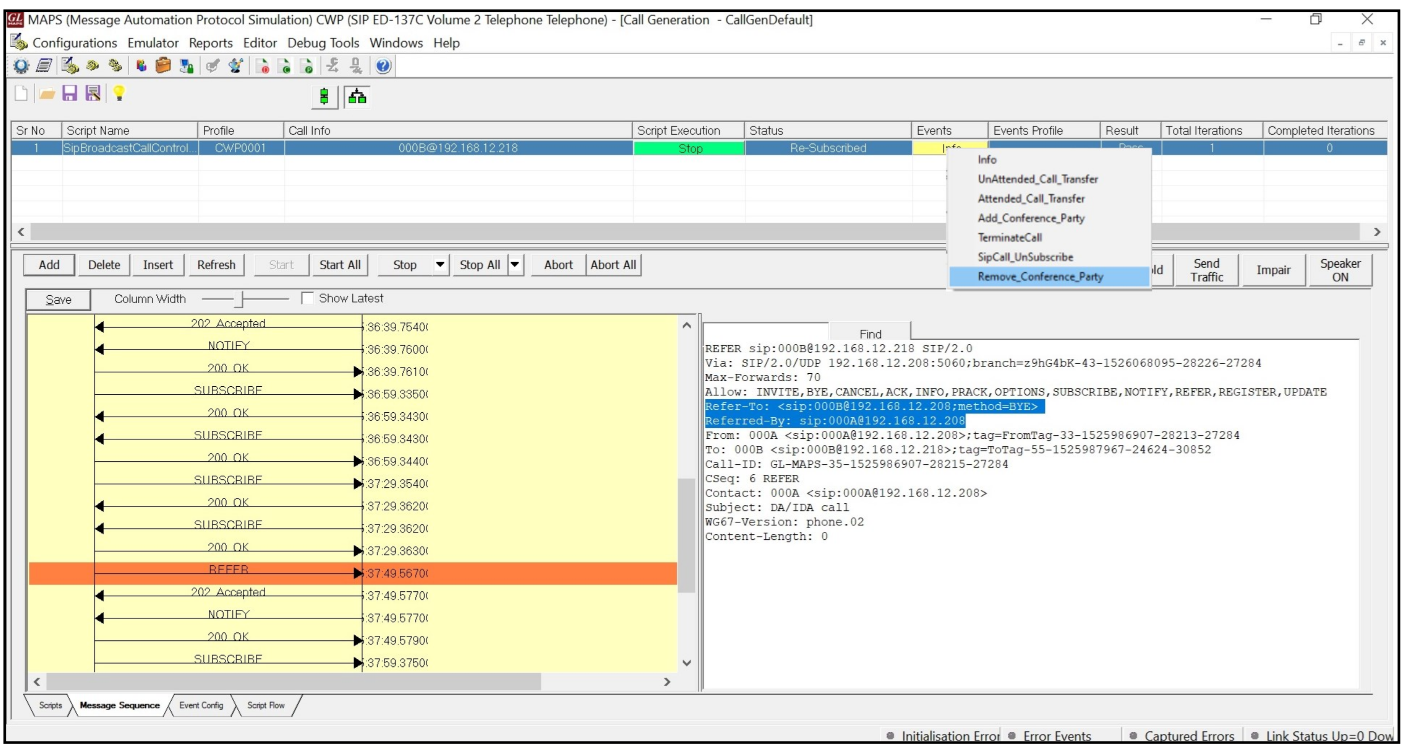
Additional Call Features

Broadcast Conference

A Broadcast Conference call is an ad-hoc Conference call where a calling party can dynamically create a conference and add or remove other user agents. The execution of a conference is based on RFC 3891 (“Replaces Header”)

Basically, MAPS™ act as a SIP end-point and don’t have FOCUS capability. As a SIP user agent, we can establish a SIP session with FOCUS and SUBSCRIBE for a Conference event over the same dialog.User Event option is provided within the established dialog to indicate the FOCUS to Add and Remove conference parties using REFER method. On termination of the call we indicate the Un-Subscription of Conference event to FOCUS by setting Expires value as 0.



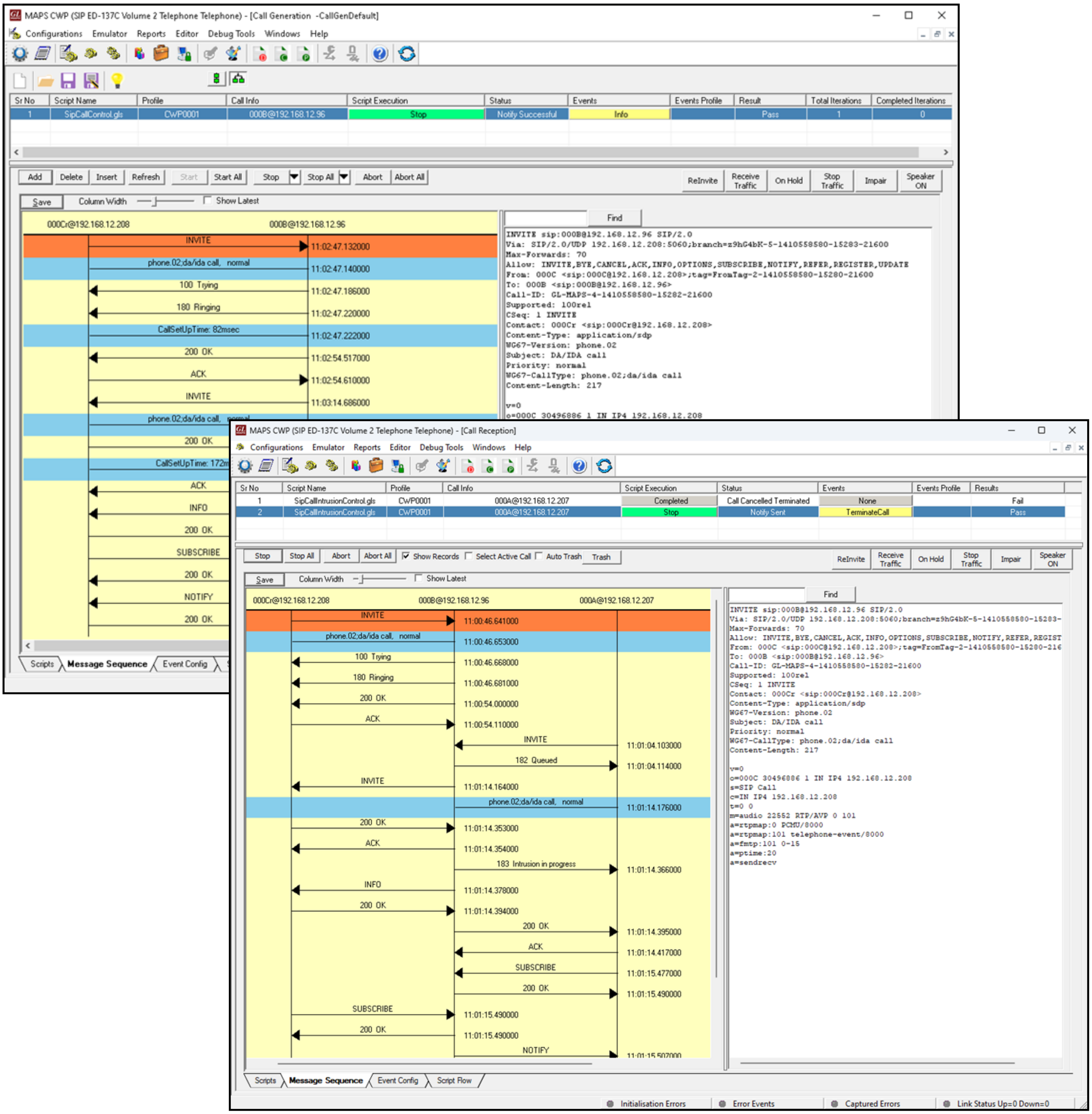


GL Communications Inc.

Additional Call Features

Call Intrusion

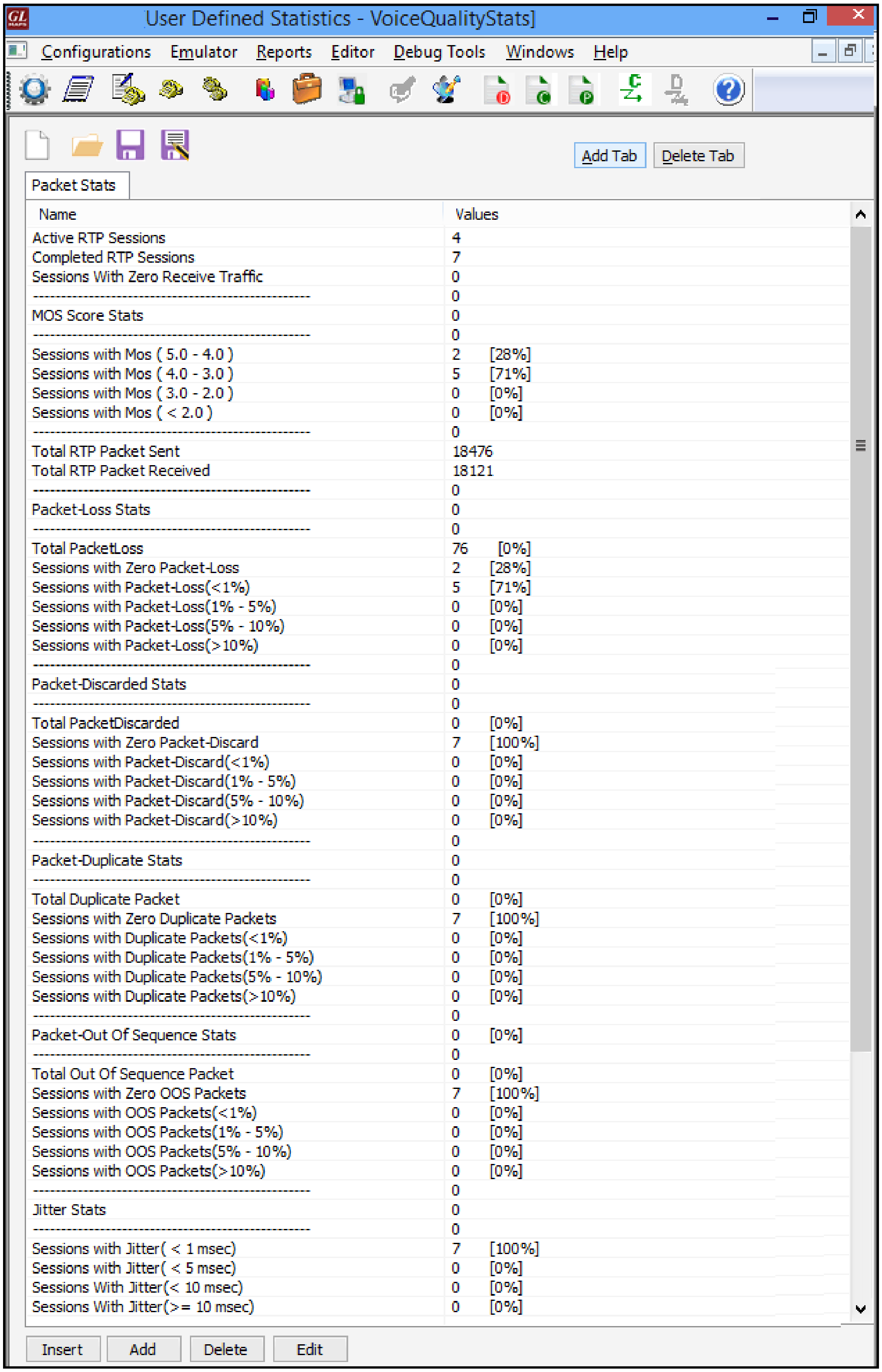
Call Intrusion ensures prompt addressing of priority calls by automatically joining all parties in a conference if the intended recipient is busy. When encountering a Busy Called UA, priority calls are intruded upon, facilitating immediate communication. Users receive visual and/or audible notifications before intrusion, maintaining awareness. While users can protect themselves from intrusions, call protection settings do not prevent priority call intrusions. This feature enhances communication efficiency by prioritizing critical calls, even in busy scenarios, and provides users with control over their call environment. Basically, MAPS act as a SIP end-point and don’t have FOCUS capability. As a SIP user agent, we can establish SIP sessions with FOCUS.



GL Communications Inc.

RTP Statistics Calculation

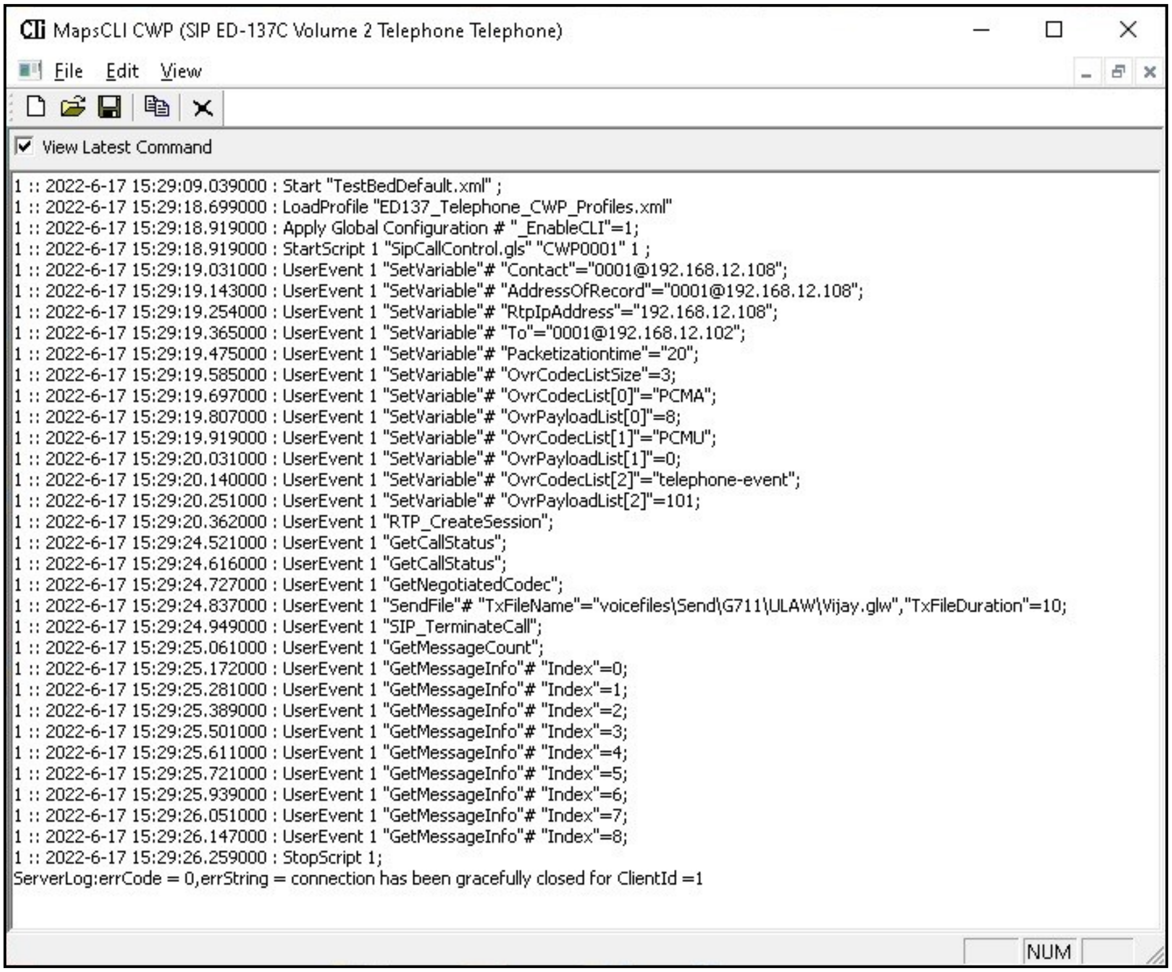
Optionally MAPS™ ED137 Telephone provides global voice quality statistics on RTP, which includes metrics such as Listening MOS, Conversational MOS, PacketLoss, Discarded Packets, Out of Sequence Packets, Duplicate Packets, Delay and Jitter. These statistics are calculated and updated periodically on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

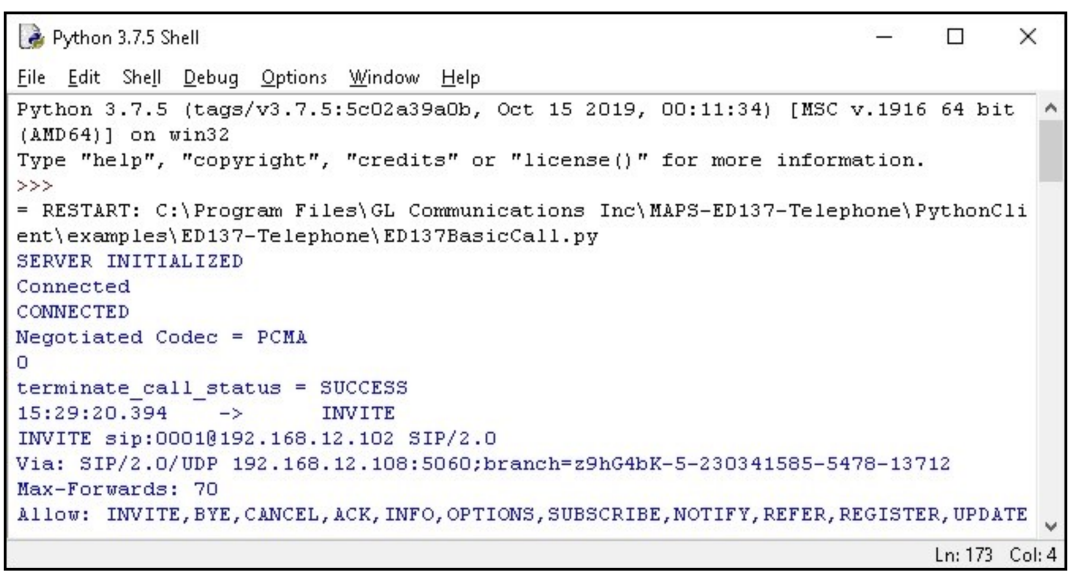


RTP statistics calculation

Command Line Interface (CLI)

MAPS™ can be configured as server-side application, to enable remote controlling of the application through multiple command-line based clients. Supported clients include TCL, Python, VBScript, Java, and .Net. Client provides a simple scripting language, with programming facilities Clients can remotely perform all functions such as start testbed setup, load scripts, and profiles, apply user events such as send digits/file/tones, detect digits/file/tones, dial, originate call, terminate call, start and stop traffic and so on. User can also generate and receive calls through commands. The figure below depicts MAPS™ Python Client Interface used to place call and handle traffic between the end terminals. Also, observe the executed commands in the MAPS™ CLI Server window after completing the call.





Buyer's Guide

Item No	Product Description
PKS119	MAPS™ ED137 Telephone (includes PKS102)
PK1192	Addendum 2: FAA Legacy Telephone Interworking (requires additional license)
PK1194	Addendum 4: Override Call
PK1195	Addendum 5: Voice Call

Item No	Related Software
PKS118	MAPS™ ED137 Radio (includes PKS107 and PKS102)
PKS117	MAPS™ ED137 Recorder (includes PKS102)
PKS102	RTP Soft Core for RTP Traffic Generation
PKS107	RTP EUROCAE ED-137
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
PKV100	PacketScan™ (Online and Offline)
PKV170	Network Surveillance Software with Centralized Database Engine and Client

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

For more information, refer to MAPS™ ED-137 Telephone Emulator webpage.