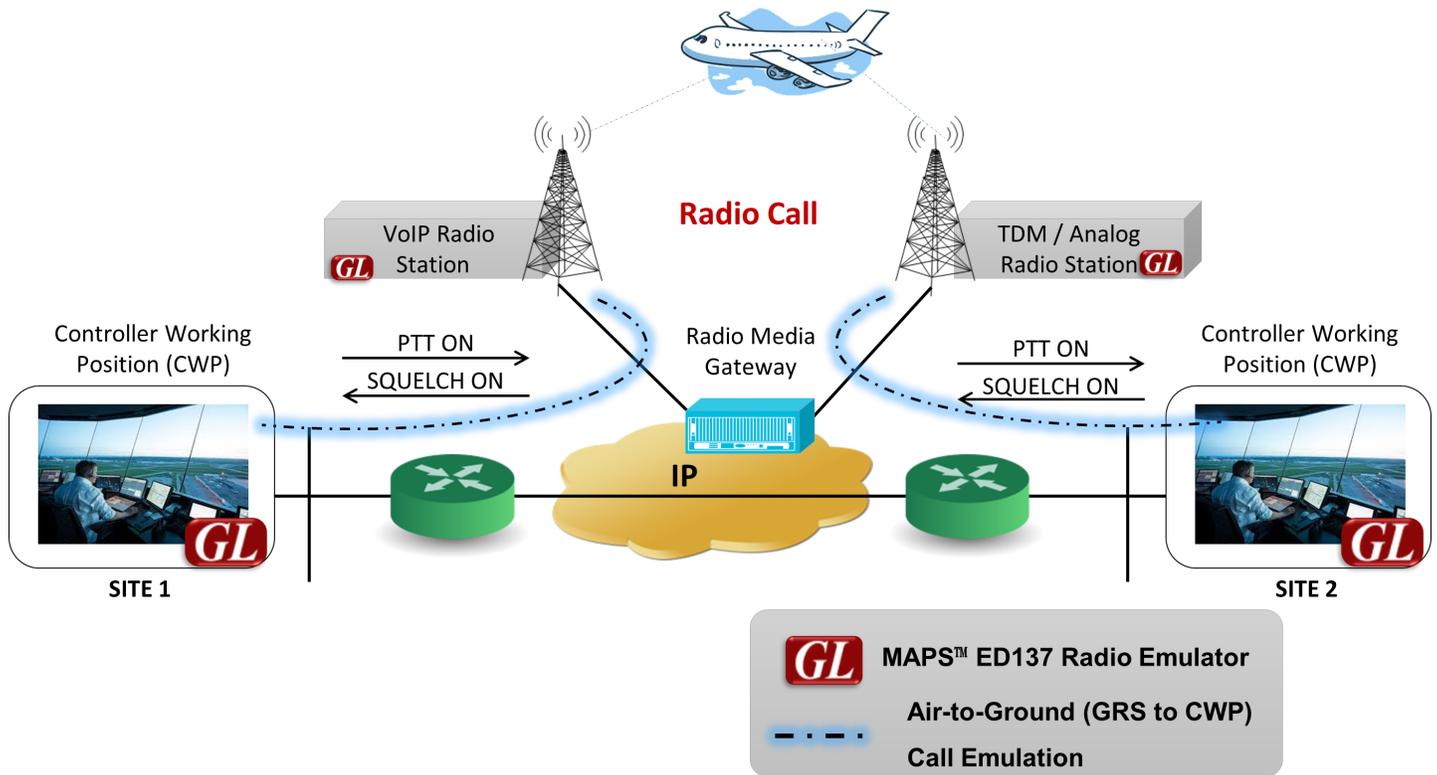


MAPS™ ED137 RADIO Emulator

(Air-to-Ground Calls Emulation)



Overview

Recent advances in Air Traffic Management (ATM) 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 [Message Automation & Protocol Simulation \(MAPS™\)](#) ED137 Radio which can emulate Air-to-Ground calls as per EUROCAE (European Organization for Civil Aviation Equipment) standards.

GL's MAPS™ ED137 Radio emulates the functions of Controller Working Position (CWP) and Ground Radio Station (GRS) and generates bulk calls (load testing) on the network supporting both ED-137B Volume 1 Radio and ED-137C Volume 1 Radio versions. 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 up to 500 simultaneous calls on a core i7 system.

MAPS™ ED137 Radio can be used to set up voice sessions over a network, then send and record test voice signals for assessing voice quality and performance. It can support important features such as transmission and detection of various RTP audio traffic such as, real time audio, voice file, digits, single tone, and dual tones, (with additional licenses) required to maintain reliable communication over air traffic network.

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. GL's ATM solution also includes [MAPS™ ED137 Recorder](#) and [MAPS™ ED137 Telephone](#) emulators.

For more details, refer to [MAPS™ ED137 Radio Emulator](#) webpage.

 **GL Communications Inc.**

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Main Features

ED-137 Signaling Emulation

- Fully integrated, complete test environment for ATM
- Supports testing CWP, Voice Communication Systems (VCS), GRS and Radio Media Gateway (RMG)
- Supports ED-137/1B, ED-137/1C, Change 1 and Change 2 standards
- Validated against latest VOTER version
- Support for Multiple Radio emulation (up to 500 Radios) within single instance of the software
- Supports hundreds of simultaneous Air to Ground calls and complete automation of bulk call generation with traffic
- IP Address Spoofing to automatically generate virtual IP address for a NIC
- Linked Session Management to group and identify all calls belonging to particular Radio
- 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 multiple profiles (users/end points) from single node and allows to define DSCP (Differentiated Service Code Point) values for signaling and voice traffic
- Supports Separated GRS (Tx and Rx) and Combined GRS simulation
- Provides option to define non-VoIP keying source parameters in GRS profile and include the details of the Non-VoIP keying source in the NOTIFY to the WG67-KeyIn event
- Enhanced to support the PTT priority for both Non-VoIP and VoIP sources
- Supports PTT Summation and Coupling PTT Summation modes
- **ED-137C Volume 1 Radio**
 - CWP and GRS nodes support Radio Receiver Multicast operation
 - Supports selective calling with SELCAL tone transmission and emulates non-VoIP source PTT keying
 - WG67 KEY-IN event package includes Frequency ID (FID) parameter to inform the User Agent about the new FID
 - No disconnect of active sessions when GRS frequency id changes
 - Includes PTT type - Test PTT and all SIP requests and responses will have WG67-Version header updated to 'radio.02'
 - Provides an option to send packets with RRC responses from GRS at defined intervals
 - Supports enable/disable active voice call preemption at GRS node
- **ED-137B Volume 1 Radio**
 - Support for user-events that can be applied dynamically on an established call (Re-Invite, PTT Priority, Signal Quality Information (SQI), CLIMAX Time Delay, Receive Traffic, Delay Compensation messages (RMM and MAM), Radio Remote Control, Impairments, Play to Speaker, RTP Audio/R2S-Keepalives)
 - Supports emulation of Dynamic Delay Compensation messages - Request for Measurement Message (RMM) and Measurement Answer Message (MAM)
 - Supports sending simultaneous squelch on selected multiple Radios
 - Sample script provided to perform automated periodic Push-To-Talk (PTT) and Squelch (SQU) operations on Air-to-Ground (A-G) calls
 - Option to define multiple traffic profiles for simultaneous simulation of various traffic actions
 - Supports sending audio using microphone and playing audio to speaker on multiple sessions

Traffic Emulation

- Send and receive live speech, pre-recorded speech files, digits and tones
- Set impairments (Packet Loss, Packet Effects and Latency) in relevant profile in real time
- Supports user-defined and automated traffic actions on the call
- Supports all standard codecs, including G.711 (μ -Law and A-Law) and G.729

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 Radio Use Cases

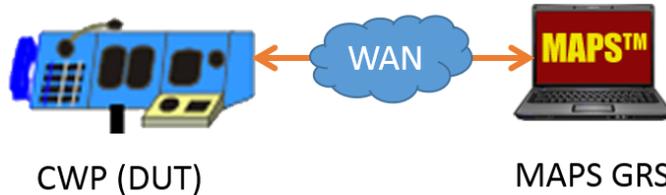
Scenario 1: MAPS™ acting as CWP and testing GRS

MAPS™ ED137 acting as CWP generates Radio calls to GRS.



Scenario 2: MAPS™ acting as GRS and testing CWP

MAPS™ ED137 can be configured to act as GRS to receive Radio calls from CWP or VCS.



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.

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

Many other Events can be applied at CWP/GRS over the established call such as User-defined Signal Quality Information (SQI), Play back the call on Speaker, Start/Stop sending RTP Audio/R2S-Keepalives, and Start/Stop Impairments.

Color coding feature included in the configuration is used to sort and group all received calls related to individual radios as shown in the below screenshot.

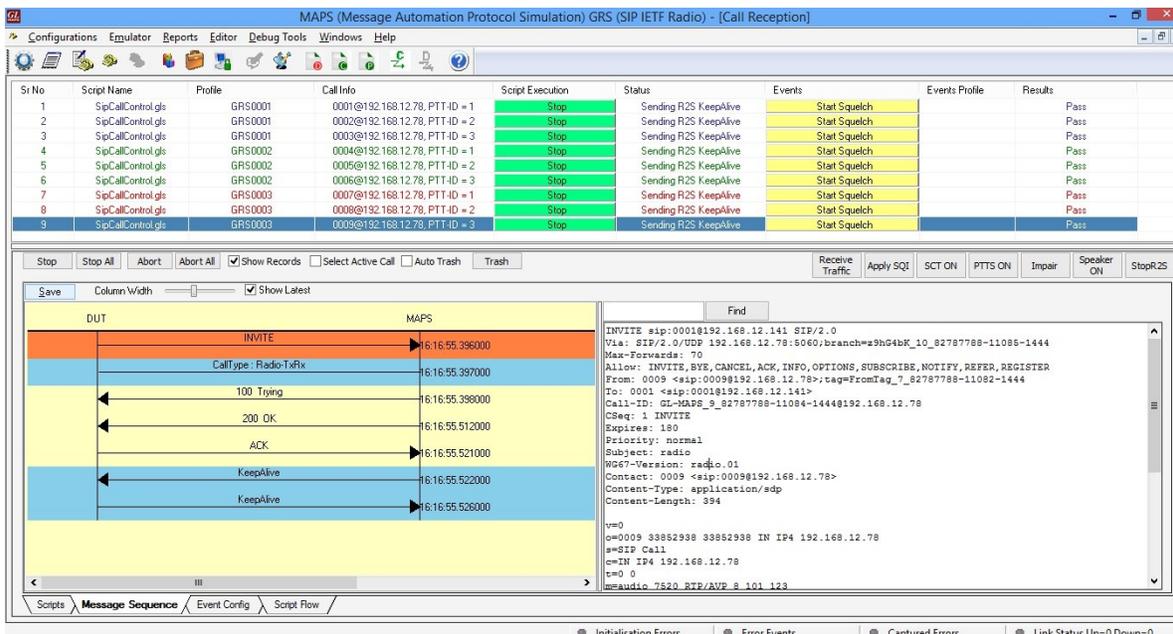


Figure: Call Reception at Radio (MAPS™ ED137 Radio)

CWP/GRS Radio Call Profiles

ED-137 Volume 1 Radio Interface defines additional SDP Parameters and SIP headers needs to be configured to simulate Air-to-Ground calls. These parameters for each profile (CWP/GRS) emulating a Radio call can be easily configured in the XML based configuration files (Profiles). Similar to signaling, traffic configuration files allow users to customize the traffic parameters.

At MAPS™ CWP, PTT types such as Normal, Priority, Emergency, and Coupling PTT ON are supported activating transmission over the air. Radio call type can be defined as Radio-Idle, Radio-RxOnly, Radio-TxRx or Radio, and Coupling types.

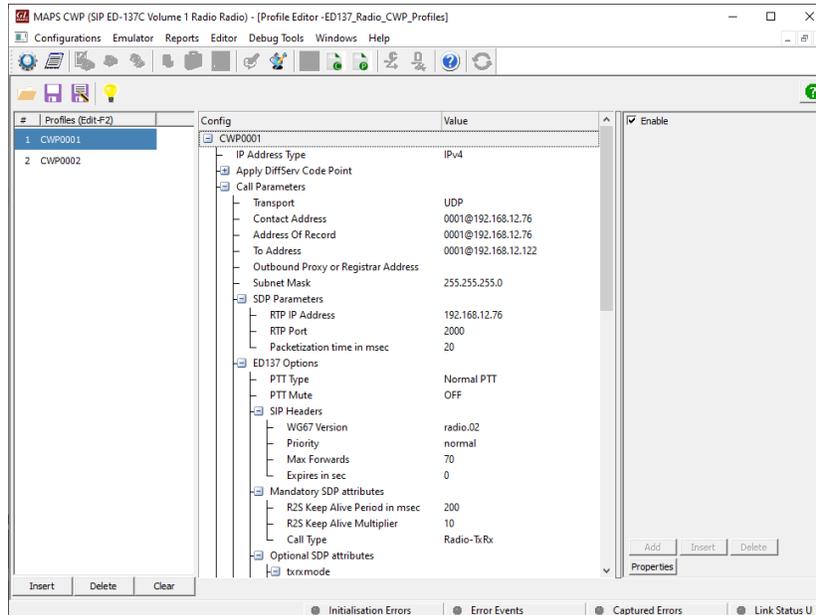


Figure: CWP Radio Call Profiles

MAPS™ ED137 GRS node profile configuration allows to emulate multiple radios within a single instance. Each emulated radio with similar capabilities can be configured with a set of parameters like Contact Address, Radio Type, Frequency-ID, Permitted User list. Key parameter settings such as Radio Emulation Type is configured for the terminal to act as Transceiver/Transmitter/Receiver and the Frequency ID of the GRS based on which the incoming calls are accepted/rejected. IP spoofing feature allows multiple CWPs/Radios to emulate using unique IP address from a single system. Also, includes color coding configuration which is used to sort and group all received calls related to individual radio.

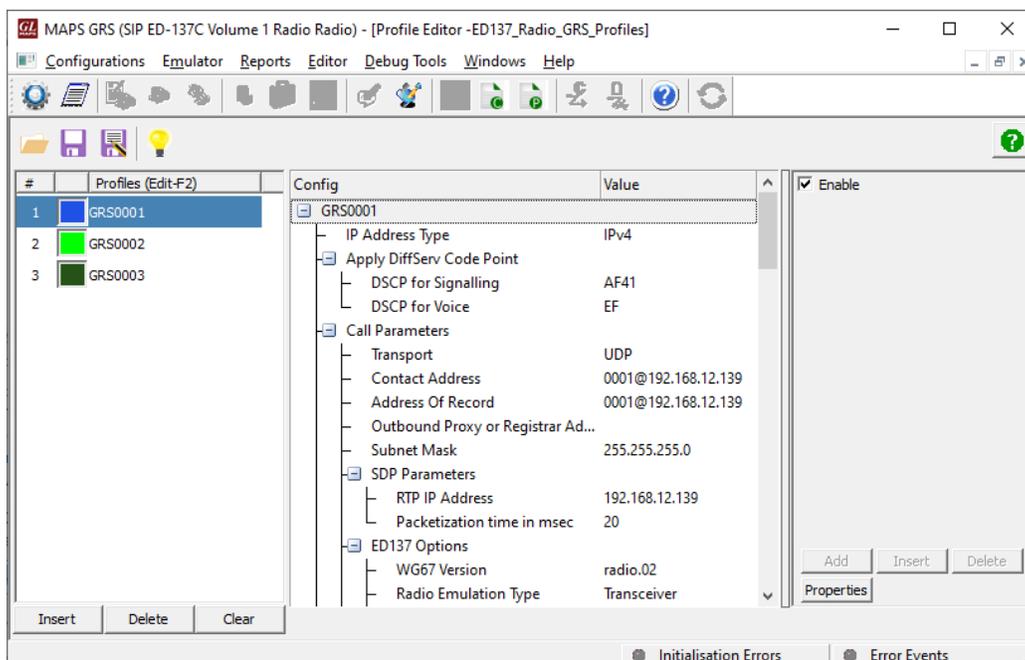


Figure: GRS Radio Call Profiles

Simulation Scenarios

Multiple CWPs/Radio Emulation

MAPS™ ED137 Radio can simulate multiple CWPs or Radios on within single MAPS™ instance. User can choose to simulate either CWP or Radios when the application is invoked.

ED137_Radio_GRS_Profiles or ED137_Radio_CWP_Profiles is an XML configuration file that includes a set of multiple sub-profiles. These sub profiles represent a CWP or Radio node. Profiles allows to configure multiple CWPs or Radios of similar capabilities with unique set of parameters such as Contact address, radio type, FID, permitted user list for each profile.

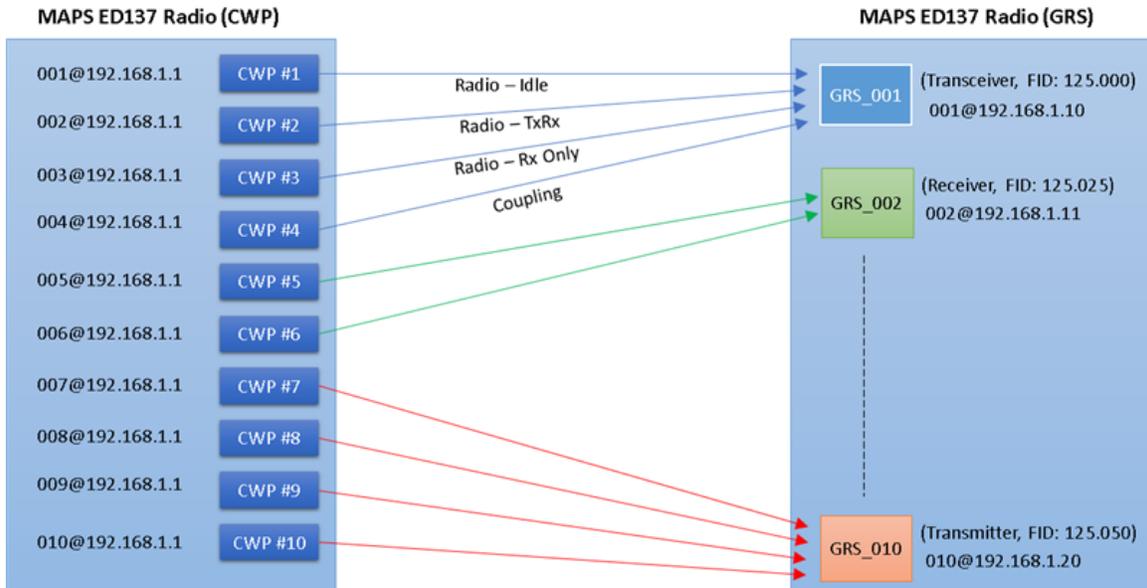


Figure: CWP Radio Call Profiles

Linked Session Management

The Linked Session functionality provides to the GRS endpoint the opportunity to detect all SIP sessions which are coming from the same user but from different equipment (i.e. different IP Address) to guarantee higher service availability.

GRS can identify the calls coming with same User part in From Address but with different IP/host address and with 'ls-pl' SDP parameter included within the SIP headers. It will treat the linked sessions as one single logical session to radio.

The linked session functionality enables the GRS endpoint to support handling of redundant connections between VCS endpoint and GRS endpoint for all types of connections.

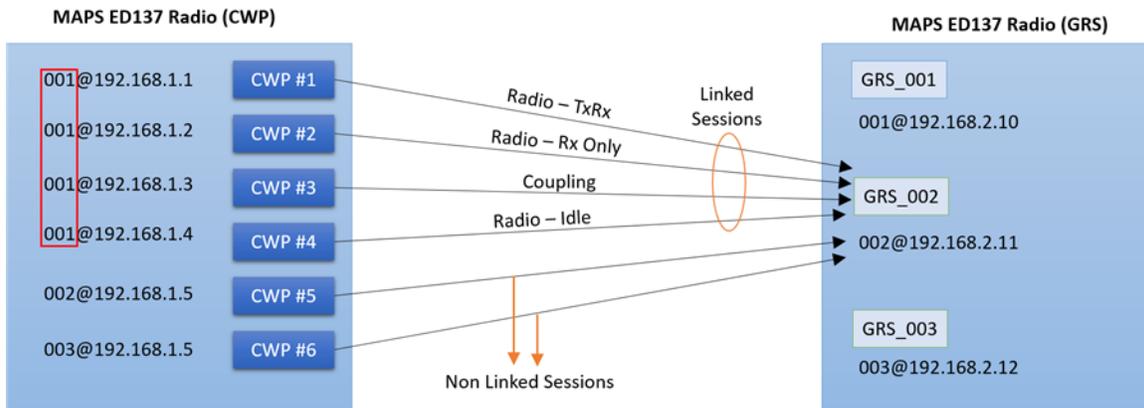
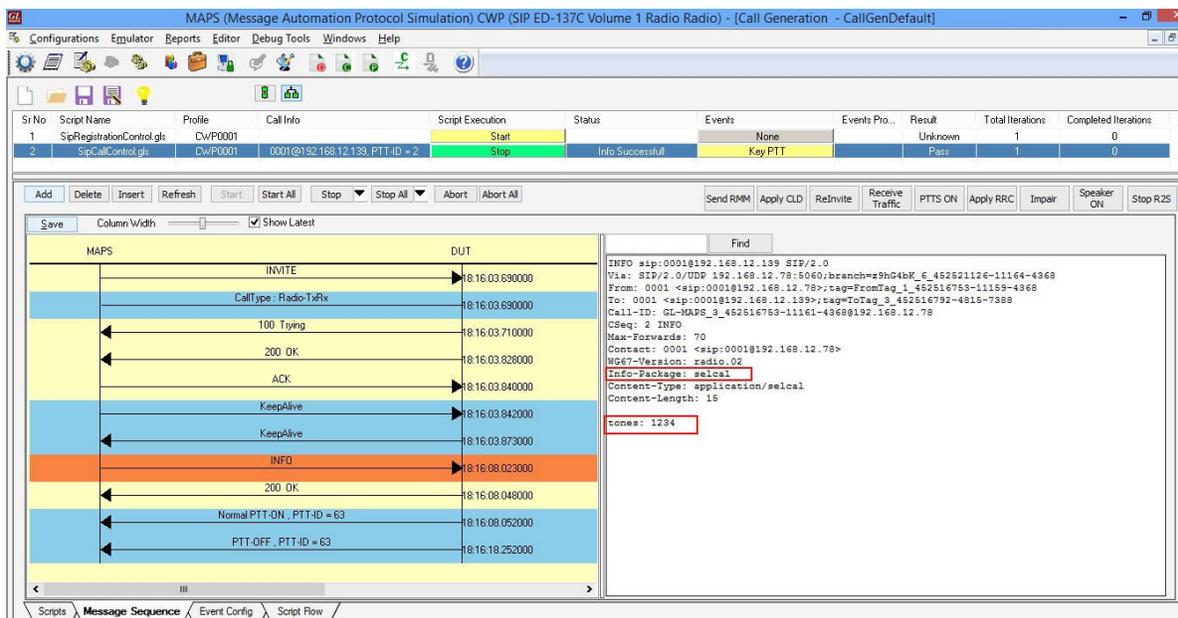


Figure: CWP Radio Call Profiles

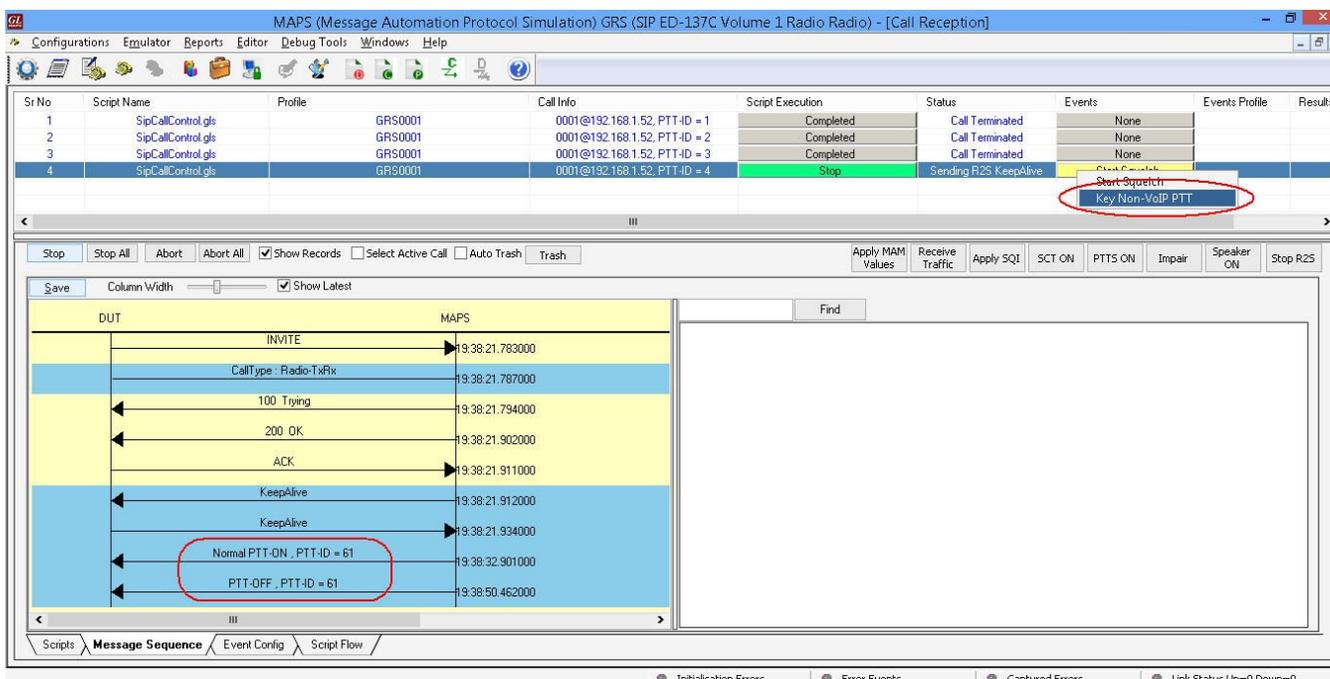
Selective Calling (SELCAL) Tone Transmission

SELCAL is a signaling method used to alert the aircraft crew members selectively to an incoming message from a ground station. The CWP endpoints emulated by MAPS™ ED137 Radio supports sending SELCAL tones to GRS using the SIP INFO method. SELCAL tone defined in CWP profile will be sent in the body of INFO message as shown in below screenshot. GRS replies with 200 OK message to INFO request and sends Normal PTT_ON confirmation with PTT-ID=63 in RTP downstream header to CWP. PTT-ID=63 is reserved for SELCAL tone transmission at GRS. PTT priorities are handled at GRS as per ED137C for this transmission.



Emulates Non-VoIP Source PTT Keying

GRS endpoints emulated by MAPS™ ED137 Radio supports simulation of non-VoIP source PTT Keying. User can simply apply “Key non-VoIP PTT” event on the selected Radio call at GRS. This will trigger GRS to send Normal PTT_ON confirmation with configured PTT-Id (60, 61 or 62) in RTP downstream header to all CWPs indicating that PTT from a non-VoIP source is being transmitted at GRS. PTT-Ids 60, 61 and 62 are reserved for non-VoIP sources. The below screenshot shows the non-VoIP source PTT ON/OFF confirmations on the call graph.



Separated GRS Scenario

A Separated GRS group is formed by one Transmitter and one or more Receivers, all configured with same frequency F1. Users can have one CWP connected to Transmitter (F1) and one or more CWPs connecting to a single Receiver (F1) or to multiple Receivers with same frequency F1. The below screenshot shows the emulation of separated GRS scenario.

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile	Result	Total Iterations
1	SipCallControl.gls	CWP0001	0001@192.168.12.212, PTT-ID = 3	Stop	Send_File-Started	Unkey PTT		Pass	1
2	SipCallControl.gls	CWP0002	0001@192.168.12.141, PTT-ID = 5	Stop	SQUELCH is ON	None		Pass	1
3	SipCallControl.gls	CWP0003	0001@192.168.12.141, PTT-ID = 6	Stop	SQUELCH is ON	None		Pass	1
4	SipCallControl.gls	CWP0004	0001@192.168.12.141, PTT-ID = 7	Stop	SQUELCH is ON	None		Pass	1
5	SipCallControl.gls	CWP0005	0001@192.168.12.141, PTT-ID = 8	Stop	SQUELCH is ON	None		Pass	1
6	SipCallControl.gls	CWP0006	0001@192.168.12.141, PTT-ID = 9	Stop	SQUELCH is ON	None		Pass	1

MAPS	DUT	Time	Event
		12:07:11.877000	INVITE sip:0001@192.168.12.212 SIP/2.0
		12:07:11.882000	Via: SIP/2.0/UDP 192.168.12.119:5060;branch=s9hG4bK-11-866008330-5221-6052
		12:07:11.890000	100 Trying
		12:07:12.009000	200 OK
		12:07:12.013000	ACK
		12:07:12.021000	KeepAlive
		12:07:12.030000	KeepAlive
		12:07:19.561000	Normal PTT-ON, PTT-ID = 3
		12:07:19.564000	Normal PTT-ON, PTT-ID = 3

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile	Results
1	SipCallControl.gls	GRS0001	0001@192.168.12.119, PTT-ID = 1	Stop	Sending R2S KeepAlive	Key Non-VoIP PTT		Pass
2	SipCallControl.gls	GRS0002	0002@192.168.12.119, PTT-ID = 2	Stop	SQUELCH is ON	None		Pass
3	SipCallControl.gls	GRS0002	0002@192.168.12.119, PTT-ID = 2	Stop	SQUELCH is ON	None		Pass
4	SipCallControl.gls	GRS0002	0004@192.168.12.119, PTT-ID = 3	Stop	SQUELCH is ON	None		Pass
5	SipCallControl.gls	GRS0002	0005@192.168.12.119, PTT-ID = 4	Stop	SQUELCH is ON	None		Pass
6	SipCallControl.gls	GRS0002	0006@192.168.12.119, PTT-ID = 5	Stop	SQUELCH is ON	None		Pass

DUT	MAPS	Time	Event
		15:21:20.444000	INVITE sip:0001@192.168.12.141 SIP/2.0
		15:21:20.447000	Via: SIP/2.0/UDP 192.168.12.119:5060;branch=s9hG4bK-9-97717278-1185-5080
		15:21:20.451000	100 Trying
		15:21:20.570000	200 OK
		15:21:20.592000	ACK
		15:21:20.596000	KeepAlive
		15:21:20.719000	KeepAlive
		15:21:30.889000	Normal PTT-ON, PTT-ID = 1
		15:21:30.889000	Normal PTT-ON, PTT-ID = 1
		15:21:30.890000	Normal PTT-ON, PTT-ID = 1
		15:21:30.891000	SQU-ON
		15:21:30.892000	SQU-ON

Combined GRS Scenario

In case of Combined GRS (Tx + Rx), CWP establishes a RadioTxRx or Coupling session to a Transceiver. When PTT is keyed on this session Transceiver will loop back the received audio on the same session with PTT Type, PTT-Id set as received and Squelch bit set ON. The below screenshot shows the emulation of Combined GRS scenario.

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile	Result	Total Iterations
1	SipCallControl.gls	CWP0001	0001@192.168.12.212, PTT-ID = 2	Stop	SQUELCH is ON	Unkey PTT		Pass	1

MAPS	DUT	Time	Event
		10:21:30.185000	INVITE
		10:21:30.195000	Call Type : Radio-TxRx , Priority : normal
		10:21:30.225000	100 Trying
		10:21:30.332000	200 OK
		10:21:30.343000	ACK
		10:21:30.354000	KeepAlive
		10:21:30.456000	KeepAlive
		10:21:32.547000	Normal PTT-ON , PTT-ID = 2
		10:21:32.551000	Normal PTT-ON , PTT-ID = 2
		10:21:32.555000	SQI
		10:21:32.556000	SQU-ON


```

INVITE sip:0001@192.168.12.212 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.119:5060;branch=z9hG4bK-4-111886627-2174-5124
Max-Forwards: 70
Allow: INVITE,BYE,CANCEL,ACK,INFO,OPTIONS,SUBSCRIBE,NOTIFY,REFER,REGISTER
From: 0001 <sip:0001@192.168.12.119>;tag=FromTag-1-111886627-2171-5124
To: 0001 <sip:0001@192.168.12.212>
Contact: 0001 <sip:0001@192.168.12.119>
Call-ID: GL-MAPS-3-111886627-2173-5124@192.168.12.119
CSeq: 1 INVITE
Priority: normal
Subject: radio
WG67-Version: radio.02
Supported: 100rel
Content-Type: application/sdp
Content-Length: 480

v=0
o=0001 33852938 33852938 IN IP4 192.168.12.119
s=SIP Call
c=IN IP4 192.168.12.119
t=0
m=audio 2000 RTP/AVP 0 8 18 101 123
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annex=no
a=rtpmap:101 telephone-event/8000
a=fmtn:101 0-15
    
```

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile	Results
1	SipCallControl.gls	GRS0001	0001@192.168.12.119, PTT-ID = 1	Completed	Call Terminated	None		Pass
2	SipCallControl.gls	GRS0001	0001@192.168.12.119, PTT-ID = 2	Stop	SQUELCH is ON	Start Squelch		Pass

DUT	MAPS	Time	Event
		10:21:09.609000	INVITE
		10:21:09.639000	Call Type : Radio-TxRx , Priority : normal
		10:21:09.670000	100 Trying
		10:21:09.793000	200 OK
		10:21:09.847000	ACK
		10:21:09.851000	KeepAlive
		10:21:09.985000	PTTS-OFF
		10:21:09.986000	KeepAlive
		10:21:13.095000	Normal PTT-ON , PTT-ID = 1
		10:21:13.096000	Normal PTT-ON , PTT-ID = 1
		10:21:13.098000	Normal PTT-ON , PTT-ID = 1
		10:21:13.099000	SQU-ON
		10:21:13.100000	SQI

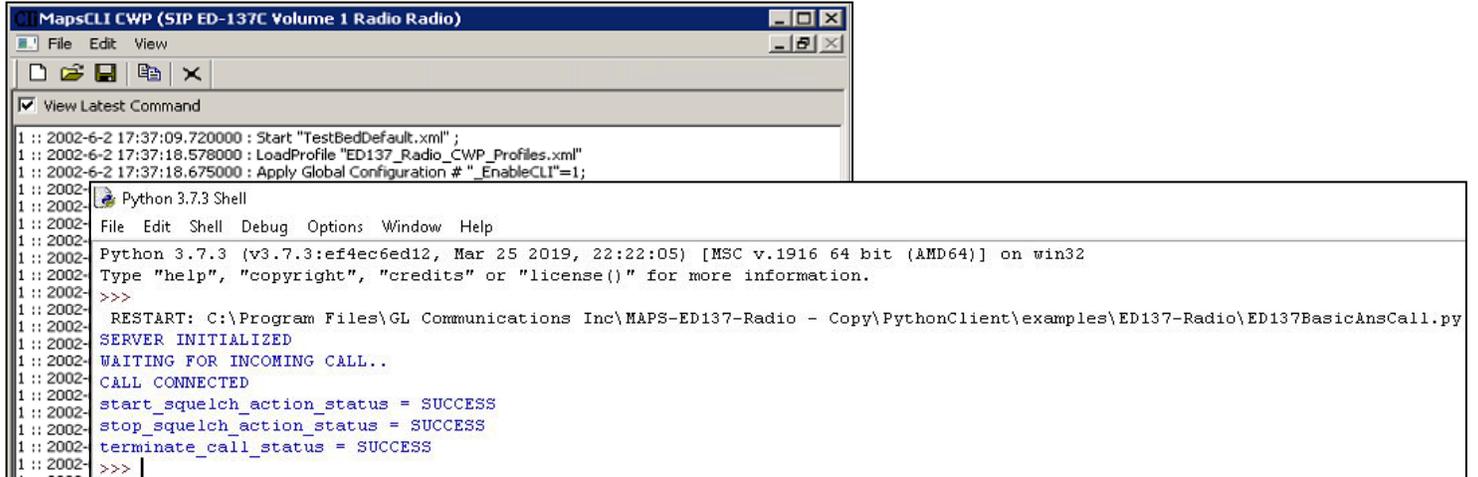

```

INVITE sip:0001@192.168.12.212 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.119:5060;branch=z9hG4bK-5-1118846033-2162-1504
Max-Forwards: 70
Allow: INVITE,BYE,CANCEL,ACK,INFO,OPTIONS,SUBSCRIBE,NOTIFY,REFER,REGISTER
From: 0001 <sip:0001@192.168.12.119>;tag=FromTag-2-1118846033-2159-1504
To: 0001 <sip:0001@192.168.12.212>
Contact: 0001 <sip:0001@192.168.12.119>
Call-ID: GL-MAPS-4-1118846033-2161-1504@192.168.12.119
CSeq: 1 INVITE
Priority: normal
Subject: radio
WG67-Version: radio.02
Supported: 100rel
Content-Type: application/sdp
Content-Length: 480

v=0
o=0001 33852938 33852938 IN IP4 192.168.12.119
s=SIP Call
c=IN IP4 192.168.12.119
t=0
m=audio 2000 RTP/AVP 0 8 18 101 123
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annex=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:123 R2S/8000
a=rtphe:1
    
```

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 Python. The 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. Users can also generate and receive calls through commands. The below screenshot depicts MAPS™ Python client interface used to place call and handle traffic between the end terminals.



```
MapsCLI CWP (SIP ED-137C Volume 1 Radio Radio)
File Edit View
View Latest Command
1 :: 2002-6-2 17:37:09.720000 : Start "TestBedDefault.xml";
1 :: 2002-6-2 17:37:18.578000 : LoadProfile "ED137_Radio_CWP_Profiles.xml"
1 :: 2002-6-2 17:37:18.675000 : Apply Global Configuration # "_EnableCLI"=1;
1 :: 2002-
1 :: 2002- Python 3.7.3 Shell
1 :: 2002- File Edit Shell Debug Options Window Help
1 :: 2002- Python 3.7.3 (v3.7.3:ef4ec6ed12, Mar 25 2019, 22:22:05) [MSC v.1916 64 bit (AMD64)] on win32
1 :: 2002- Type "help", "copyright", "credits" or "license()" for more information.
1 :: 2002- >>>
1 :: 2002- RESTART: C:\Program Files\GL Communications Inc\MAPS-ED137-Radio - Copy\PythonClient\examples\ED137-Radio\ED137BasicAnsCall.py
1 :: 2002- SERVER INITIALIZED
1 :: 2002- WAITING FOR INCOMING CALL..
1 :: 2002- CALL CONNECTED
1 :: 2002- start_squelch_action_status = SUCCESS
1 :: 2002- stop_squelch_action_status = SUCCESS
1 :: 2002- terminate_call_status = SUCCESS
1 :: 2002- >>>
```

Buyer's Guide

Item No	Product Description
PKS118	MAPS™ ED137 Radio (includes PKS107 and PKS102)
PKS119	MAPS™ ED137 Telephone (includes PKS102)
PKS117	MAPS™ ED137 Recorder (includes PKS102)

Item No	Related Software
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

For more details, refer to [MAPS™ ED137 Radio Emulator](#) webpage.



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