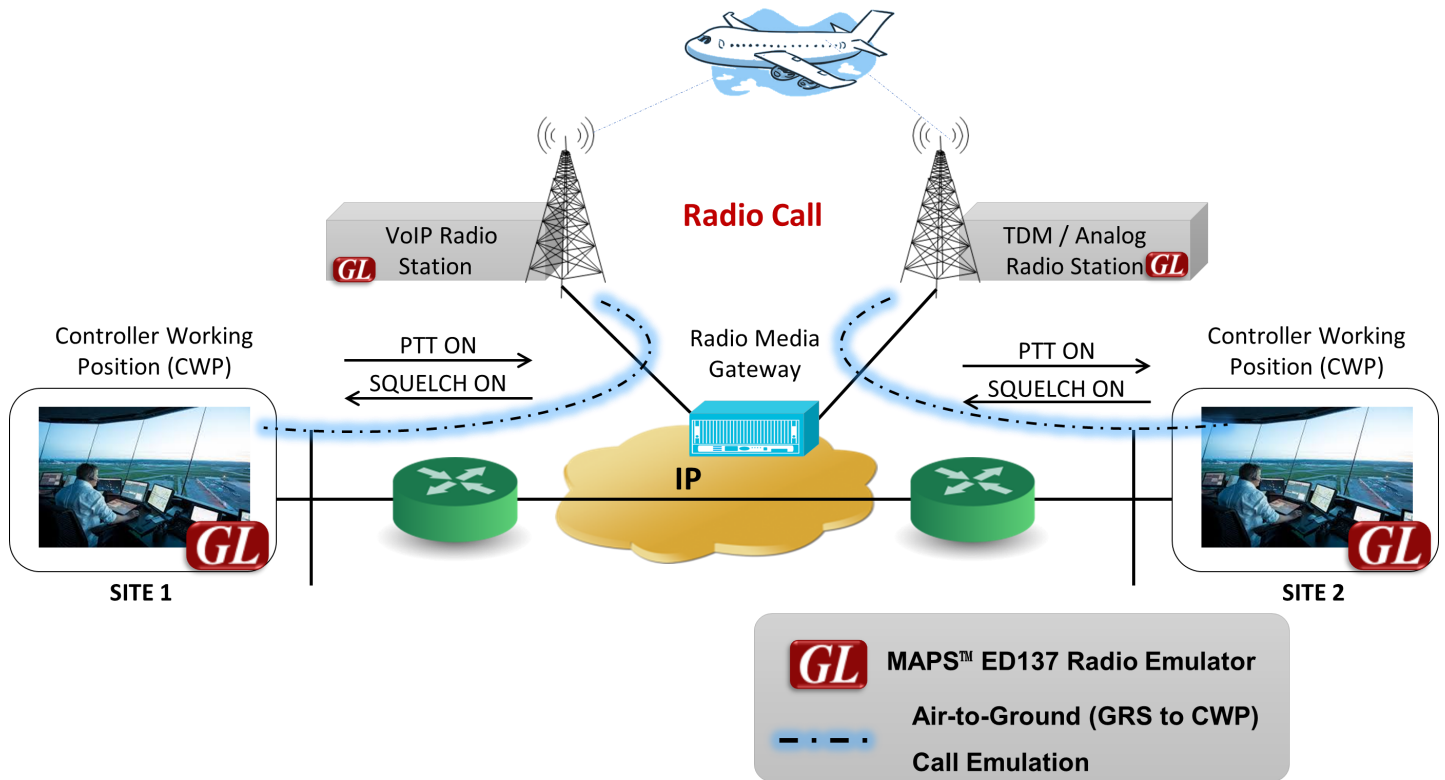


MAPS™ ED137 RADIO Emulator

(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 both Air-to-Ground calls and Ground-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) or Radio Media Gateway (RMG) entities and generates bulk calls (load testing) on the network.

MAPS™ ED137 Radio emulates Air-Ground calls supporting both ED137B Volume 1 Radio and ED137C 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 more than 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.

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Main Features

ED137 Signaling Emulation

- Fully integrated, complete test environment for ATM
- Supports testing CWP, Voice Communication Systems (VCS), GRS (or RMG), and VRS elements
- Supports both ED137B Volume 1 Radio and ED137C Volume 1 Radio versions of Air-Ground call simulation
- Support for Multiple Radio emulation (up to 200 Radios) within single instance of the software
- 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
- 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 scenarios
- Supports PTT Summation and Coupling PTT Summation modes
- ED137B 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
- ED137C 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'

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
- Provides aggregated voice quality statistics such as MOS/R-Factor, packet loss, duplicate and out of sequence packets
- 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
- 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

MAPS™ ED137 Radio Use Cases

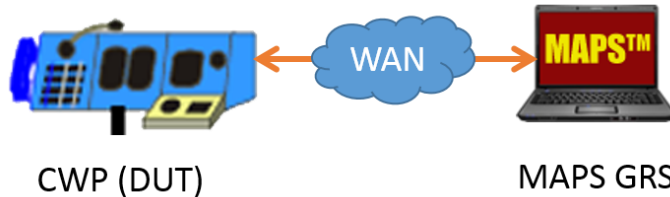
Scenario 1: MAPS™ acting as CWP and testing GRS

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 out going 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 image below.

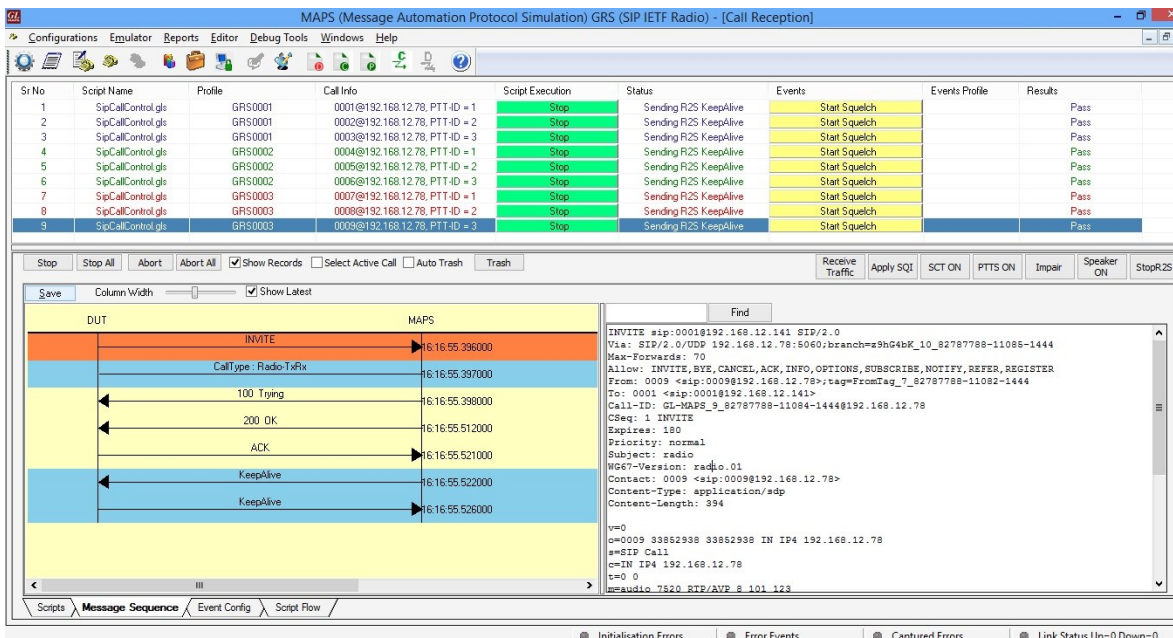


Figure: Call Reception at Radio (MAPS™ ED137 Radio)

CWP/GRS Radio Call Profiles

ED137B of EUROCAE standard option adds additional SDP Parameters and SIP headers while placing the calls which are required for generating R2S Keep Alive packets. These parameters for each profile (CWP/GRS) simulating a Radio call can be easily configured in the XML based configuration files. 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. The type of the call session can also be defined as Radio-Idle, Radio-Rxonly, Radio-TxRx or Radio, and Coupling types.

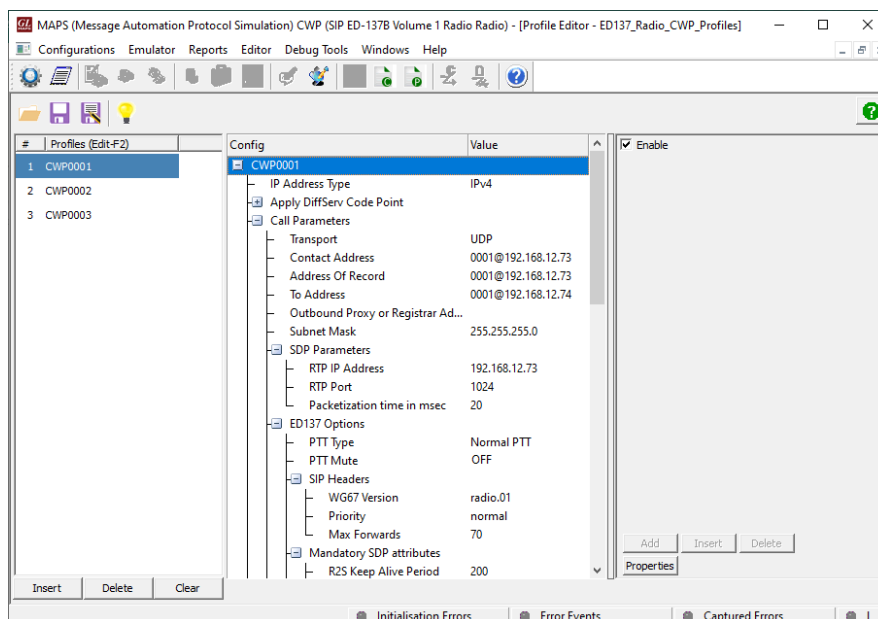


Figure: CWP Radio Call Profiles

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.

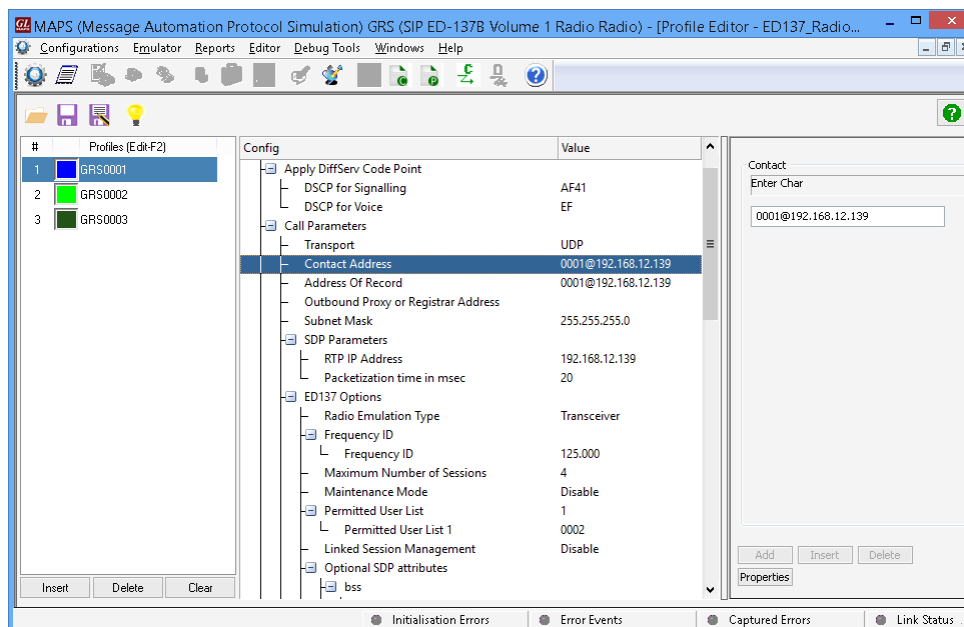


Figure: GRS Radio Call Profiles

Simulation Scenarios

Multiple Radio Emulation

MAPS™ ED137 Radio is enhanced to support multiple radios (GRS) emulation within single MAPS™ instance, receiving Radio calls from multiple CWP or VCS. It is possible to emulate multiple radios within single MAPS™ ED137 Radio instance using profile configuration.

ED137_Radio_GRS_Profiles is an XML configuration file that includes a set of multiple sub-profiles, which allows to configure multiple radios of similar capabilities with unique set of parameters such as Contact address, radio type, FID, permitted user list for each profile. Color coding feature is also included in the configuration which are used to sort and group all received calls related to individual radios.

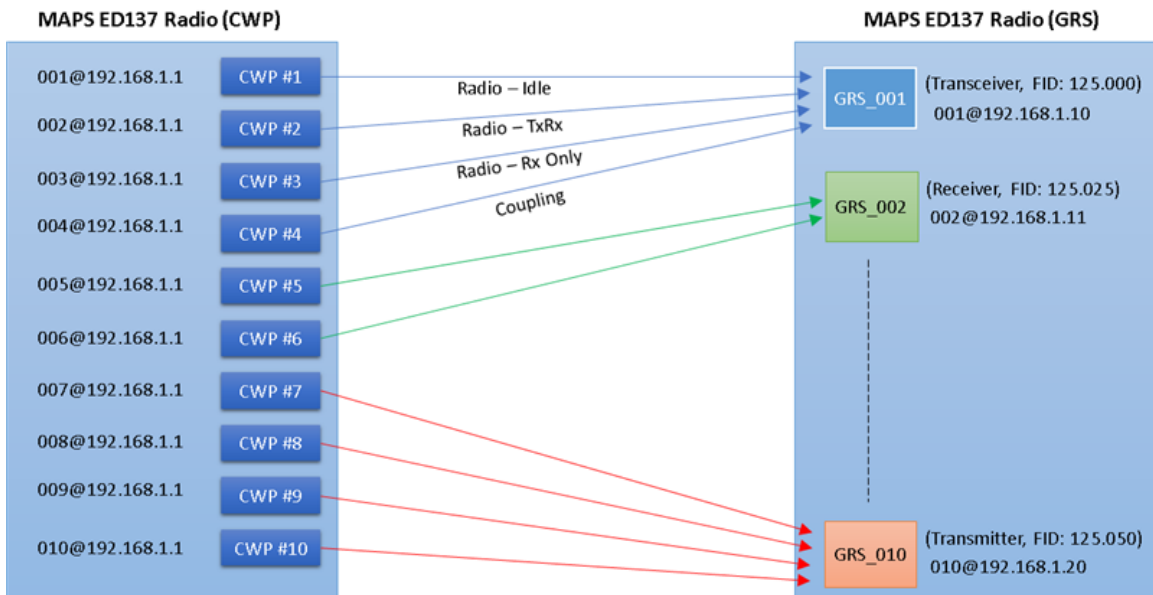


Figure: CWP Radio Call Profiles

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.

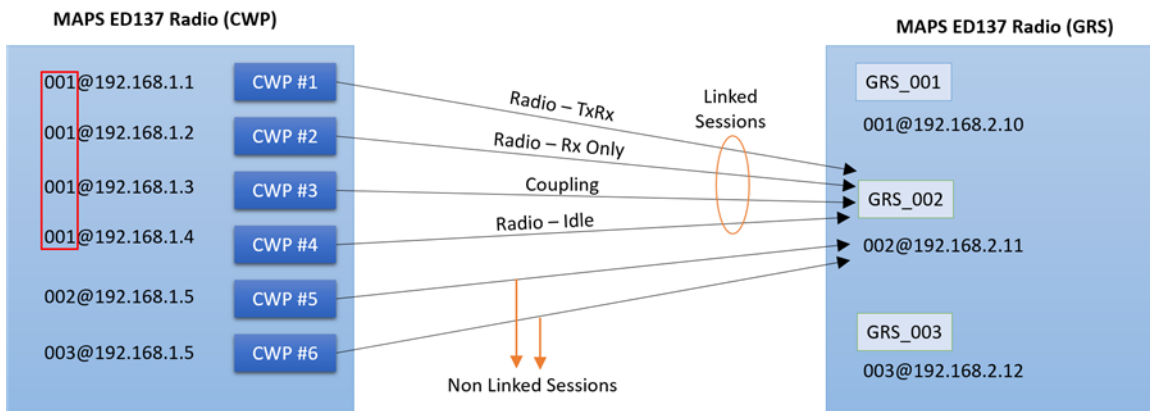


Figure: CWP Radio Call Profiles

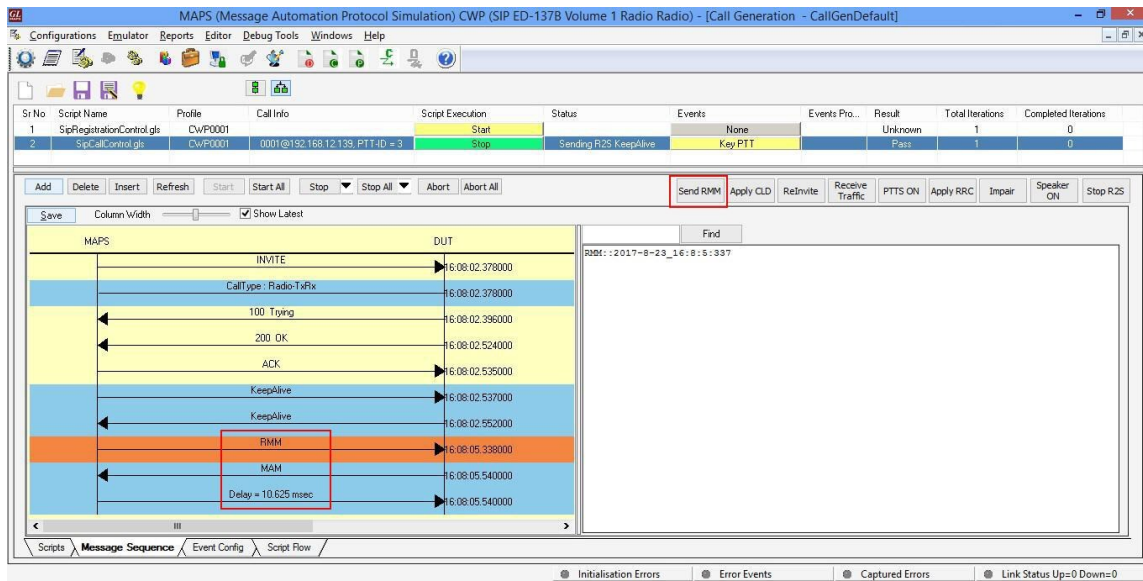
Emulation Scenarios

Emulation of Dynamic Delay Compensation Messages

Enhanced MAPS™ ED137B Radio adds Dynamic Delay Compensation as an additional feature in which specific RTP extension block used to introduce varying delay values at GRS in lab environment without actually making the real-time measurements.

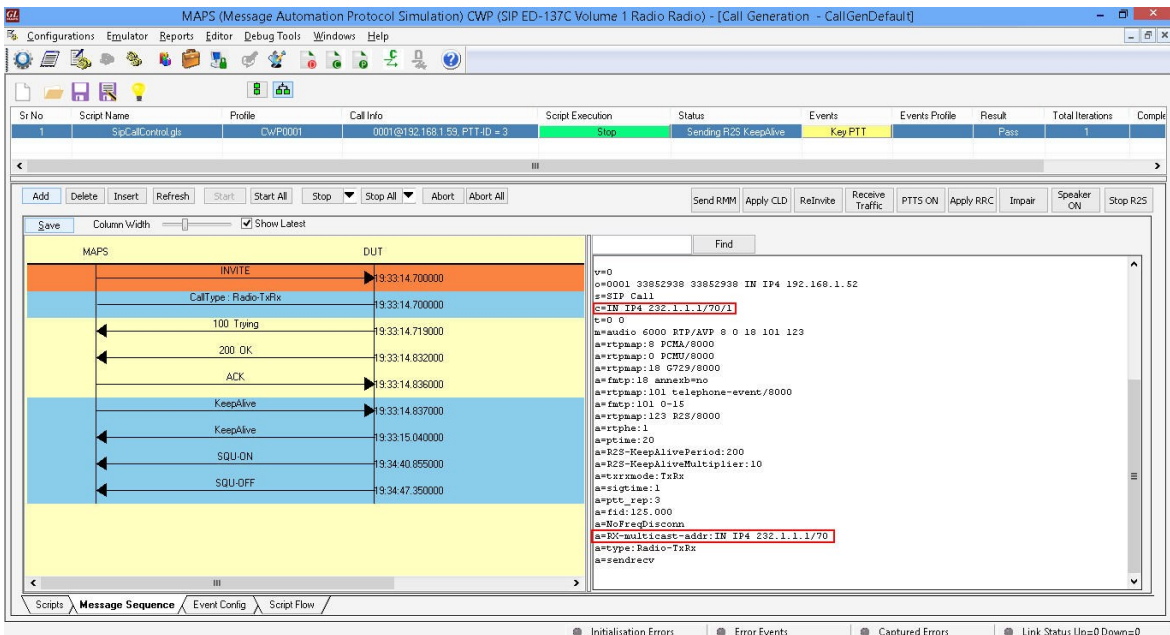
Once the Air-to-Ground call is established, CWP can send Dynamic Delay Compensation messages such as RMM to GRS. GRS replies with MAM. Message for each RMM received.

MAPS™ uses only relative time to calculate delay. After receiving the MAM message from GRS, delay is calculated and displayed in the message sequence graph along with RMM and MAM messages, as shown in the below image.



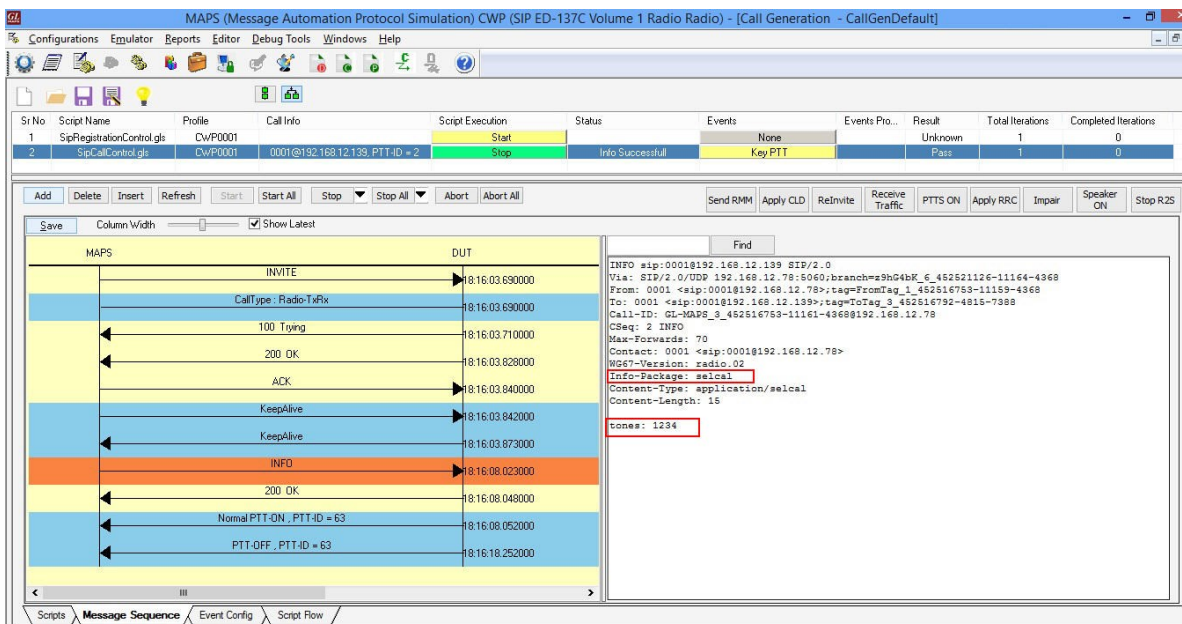
Radio Receiver Multicast Operation

In Multicast mode operation, multiple CWPs can send request to have multicast session with GRS to receive multicast RTP packets from GRS. CWP sends an INVITE request with SDP body containing multicast group address and TTL value to GRS. The GRS which supports multicast will extract the multicast address from SDP body in INVITE and starts sending the R2S/RTP packets to this multicast address. CWPs will send Internet Group Management Protocol (IGMP) join request to join the group and start receiving the multicast RTP packets. The router or switch with multicast feature will manage the subscription to multicast group and forwards the RTP packets received from GRS to all members of the group. The below image depicts the process.



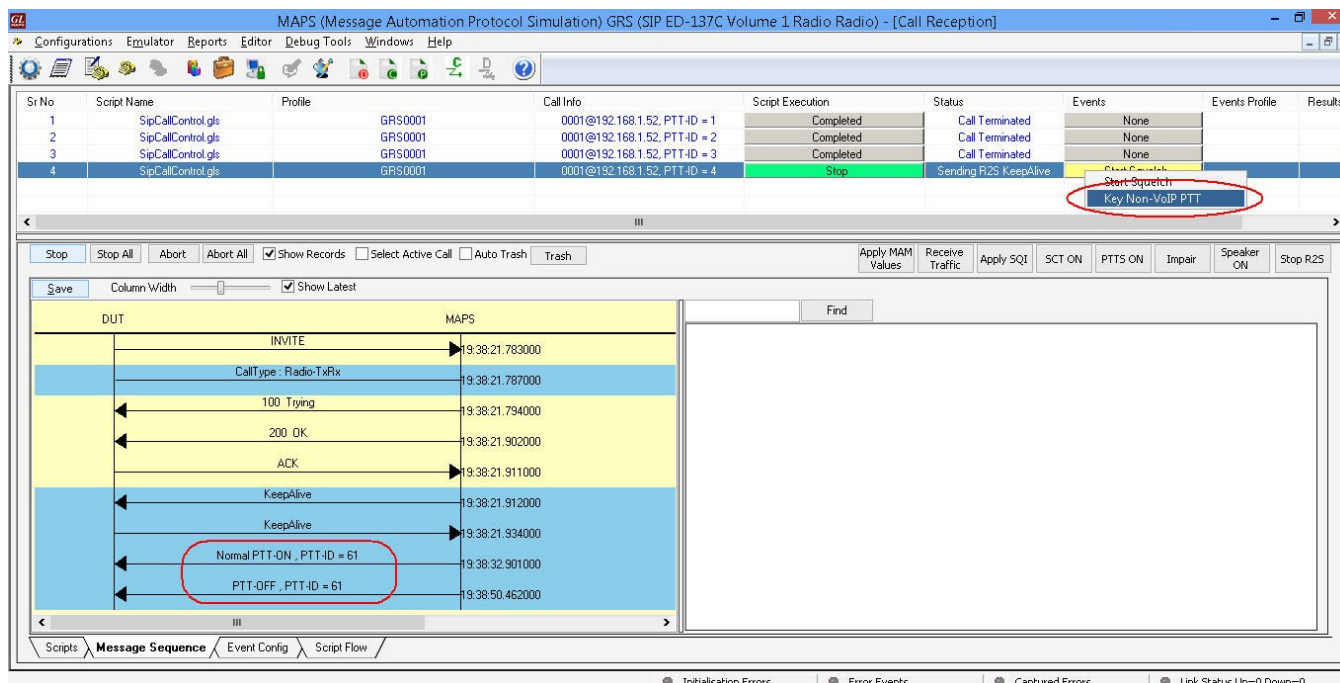
Selective Calling (SELCAL) Tone Transmission

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 image. 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 image 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. We 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 following figure 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

Time	Direction	Event
12:07:11.877000	MAPS → DUT	INVITE
12:07:11.882000	DUT → MAPS	Call type: Radio-TxRx, Priority: normal
12:07:11.890000	DUT → MAPS	100 Trying
12:07:12.009000	DUT → MAPS	200 OK
12:07:12.013000	DUT → MAPS	ACK
12:07:12.021000	DUT → MAPS	KeepAlive
12:07:12.030000	DUT → MAPS	KeepAlive
12:07:19.561000	DUT → MAPS	Normal PTT-ON, PTT-ID = 3
12:07:19.564000	DUT → MAPS	Normal PTT-ON, PTT-ID = 3

```

INVITE sip:0001@192.168.12.212 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.119:5060;branch=s9hG4bK-11-866008330-5221-6052
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, OPTIONS, SUBSCRIBE, NOTIFY, REFER, REGISTER
From: 0001 <sip:0001@192.168.12.119>;tag=FromTag-8-866008330-5218-6052
To: 0001 <sip:0001@192.168.12.212>
Contact: 0001 <sip:0001@192.168.12.119>
Call-ID: GL-MAPS-10-866008330-5220-6052@192.168.12.119
CSeq: 1 INVITE
Priority: normal
Subject: radio
WG67-Version: radio.02
Supported: 100rel
Content-Type: application/sdp
Content-Length: 478

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
    
```

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 NonVoIP PTT		Pass
2	SipCallControl.gls	GRS0002	0002@192.168.12.119, PTT-ID = 1	Stop	SQUELCH is ON	None		Pass
3	SipCallControl.gls	GRS0002	0003@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

Time	Direction	Event
15:21:20.444000	DUT → MAPS	INVITE
15:21:20.447000	MAPS → DUT	Call type: Radio-TxRx, Priority: normal
15:21:20.451000	MAPS → DUT	100 Trying
15:21:20.570000	MAPS → DUT	200 OK
15:21:20.592000	MAPS → DUT	ACK
15:21:20.596000	MAPS → DUT	KeepAlive
15:21:20.715000	MAPS → DUT	KeepAlive
15:21:30.888000	MAPS → DUT	Normal PTT-ON, PTT-ID = 1
15:21:30.889000	MAPS → DUT	Normal PTT-ON, PTT-ID = 1
15:21:30.890000	MAPS → DUT	Normal PTT-ON, PTT-ID = 1
15:21:30.891000	MAPS → DUT	SQU ON
15:21:30.892000	MAPS → DUT	SQU

```

INVITE sip:0001@192.168.12.141 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.119:5060;branch=s9hG4bK-9-97717278-1185-5080
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, OPTIONS, SUBSCRIBE, NOTIFY, REFER, REGISTER
From: 0001 <sip:0001@192.168.12.141>
To: 0001 <sip:0001@192.168.12.119>;tag=FromTag-6-97717278-1182-5080
Contact: 0001 <sip:0001@192.168.12.119>
Call-ID: GL-MAPS-8-97717278-1184-5080@192.168.12.119
CSeq: 1 INVITE
Priority: normal
Subject: radio
WG67-Version: radio.02
Supported: 100rel
Content-Type: application/sdp
Content-Length: 478

v=0
o=0002 33852938 33852938 IN IP4 192.168.12.119
s=SIP Call
c=IN IP4 192.168.12.119
t=0
m=audio 7864 RTP/AVP 0 8 18 101 123
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=rtcp:101 0-15
a=rtpmap:123 R2S/8000
a=rtpsh:1
a=rtpsh:10
a=R2S-KeepAlivePeriod:200
a=R2S-KeepAliveMultiplier:10
a=rtpsh:15
a=b3:R2S1
a=rtpsh:1
a=rtpsh:0
a=fd:126.000
a=rtype:Radio-TxRx
a=sendrecv
    
```


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 following figure shows the emulation of Combined GRS scenario.

The screenshot shows the MAPS interface for a call generation scenario. The top table lists the call details:

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

The main window displays a message sequence between MAPS and DUT:

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

The right pane shows the SIP INVITE message details:

```

INVITE sip:0001@192.168.12.212 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.119:5060;branch=z9hG4bK-4-1118866627-2174-5124
Max-Forwards: 70
Allow: INVITE,BYE,CANCEL,ACK,INFO,OPTIONS,SUBSCRIBE,NOTIFY,REFER,REGISTER
From: 0001 <sip:0001@192.168.12.119>;tag=FromTag-1-1118866627-2171-5124
To: 0001 <sip:0001@192.168.12.212>
Contact: 0001 <sip:0001@192.168.12.119>
Call-ID: GL-MAPS-3-1118866627-2173-5124@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 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=fatp:18 annex=no
a=rtpmap:101 telephone-event/8000
a=fmtn:101 0-15
    
```

The screenshot shows the MAPS interface for a call reception scenario. The top table lists the call details:

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

The main window displays a message sequence between DUT and MAPS:

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

The right pane shows the SIP INVITE message details:

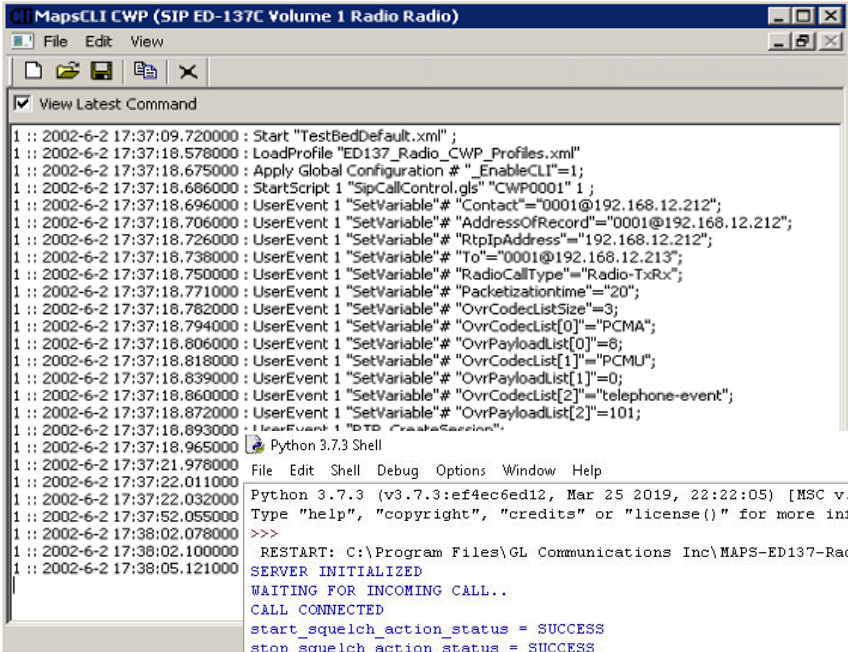
```

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 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=fatp:18 annex=no
a=rtpmap:101 telephone-event/8000
a=fmtn: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 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.



```

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 :: 2002-6-2 17:37:18.686000 : StartScript 1 "SipCallControl.gls" "CWP0001" 1 ;
1 :: 2002-6-2 17:37:18.696000 : UserEvent 1 "SetVariable"# "Contact"="0001@192.168.12.212";
1 :: 2002-6-2 17:37:18.706000 : UserEvent 1 "SetVariable"# "AddressOfRecord"="0001@192.168.12.212";
1 :: 2002-6-2 17:37:18.726000 : UserEvent 1 "SetVariable"# "RtpIpAddress"="192.168.12.212";
1 :: 2002-6-2 17:37:18.738000 : UserEvent 1 "SetVariable"# "To"="0001@192.168.12.213";
1 :: 2002-6-2 17:37:18.750000 : UserEvent 1 "SetVariable"# "RadioCallType"="Radio-TxRx";
1 :: 2002-6-2 17:37:18.771000 : UserEvent 1 "SetVariable"# "Packetizationtime"="20";
1 :: 2002-6-2 17:37:18.782000 : UserEvent 1 "SetVariable"# "OvrCodecListSize"=3;
1 :: 2002-6-2 17:37:18.794000 : UserEvent 1 "SetVariable"# "OvrCodecList[0]"="PCMA";
1 :: 2002-6-2 17:37:18.806000 : UserEvent 1 "SetVariable"# "OvrPayloadList[0]"=8;
1 :: 2002-6-2 17:37:18.818000 : UserEvent 1 "SetVariable"# "OvrCodecList[1]"="PCMU";
1 :: 2002-6-2 17:37:18.839000 : UserEvent 1 "SetVariable"# "OvrPayloadList[1]"=0;
1 :: 2002-6-2 17:37:18.860000 : UserEvent 1 "SetVariable"# "OvrCodecList[2]"="telephone-event";
1 :: 2002-6-2 17:37:18.872000 : UserEvent 1 "SetVariable"# "OvrPayloadList[2]"=101;
1 :: 2002-6-2 17:37:18.893000 : UserEvent 1 "DTMF CreateSession";
1 :: 2002-6-2 17:37:18.965000 : Python 3.7.3 Shell
1 :: 2002-6-2 17:37:21.978000 : File Edit Shell Debug Options Window Help
1 :: 2002-6-2 17:37:22.011000 : Python 3.7.3 (v3.7.3:ef4ec6ed12, Mar 25 2019, 22:22:05) [MSC v.1916 64 bit (AMD64)] on win32
1 :: 2002-6-2 17:37:22.032000 : Type "help", "copyright", "credits" or "license()" for more information.
1 :: 2002-6-2 17:37:52.055000 : >>>
1 :: 2002-6-2 17:38:02.078000 : RESTART: C:\Program Files\GL Communications Inc\MAPS-ED137-Radio - Copy\PythonClient\examples\ED137-Radio\ED137BasicAnsCall.py
1 :: 2002-6-2 17:38:05.121000 : SERVER INITIALIZED
WAITING FOR INCOMING CALL..
CALL CONNECTED
start_squelch_action_status = SUCCESS
stop_squelch_action_status = SUCCESS
terminate_call_status = SUCCESS
>>> |

```

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 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

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



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