GL Communications Inc.

Telecommunications • Software • Consulting

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April 2024

Release Notes for MAPS™ ED137 Radio - version 24.4.18

<table>
<tr><th>Reason for Release / Description of Enhancement</th><th>Version</th></tr>
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<u>Enhancements:</u>

Application is enhanced to support all the features as per ED137C_Volume_1_Radio Change 2 recommendations.
Application is enhanced with optional feature to send packets with RRC responses from GRS at the defined intervals.
Application is enhanced with optional feature to enable/disable active voice call preemption at GRS node.

<u>Bug Fixes:</u>

Fix at RTP Core:

Corrected PTT-ID in case of separated GRS scenario.
PTT-ID on the Receive session was not reflecting the PTT-ID of Transmit session.
Corrected RRC response from GRS during active PTT.
For RRC request, all the bits in RRC response header were set to zero during active PTT. Now GRS will send the current status in the RRC response.

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<u>Enhancements:</u>

Enhancement at RTP Core: Provided INI option to configure ED137A and ED137B version header for emulation and processing packets with either 0x0067 or 0x0167.
Enhancements to handle Invalid test scenarios for VOTER.

"To" header validation at GRS upon reception of ACK message.
"Request URI" and "From" header validation in case of In-Dialog and Out-Of-Dialog OPTIONS request.
Provided a "None" option in the Radio version configuration under GRS User profiles.
Optional "Expires in sec" configuration option is provided at CWP User profiles.
When configured a non-zero value, "Expires" header included in the INVITE request.
Script enhancements to handle request expiry at both CWP and GRS based on the value in Expires header.
Provided an option "All" in BSS method at CWP User profiles. When selected "All", all the BSS methods are sent in the INVITE request.

Script enhancements to add or delete virtual IPV6 addresses. [MAPSInit.gls & MAPSShutDown.gls]
Script enhancements to "Stop squelch” if PTT is keyed for transceiver radio emulation at GRS.
MAPS™ enhancement to Get System IP address utility function to return IPV4/IPV6 based request.
Script fixes at CWP, For ACK to 3xx (When 1xx is not received).
Script enhancement to "Apply RRC" if enabled in the user profiles, when "Key PTT" action is performed.
Script fixes to enable or disable PTT/Squelch user events.
Script fixes to handle script exit during few CLI test scenarios.
Script fixes to handle first call exit during multicast call scenarios.

<u>Bug Fixes:</u>

Fix at RTP Core to handle loopback failures : Start r2s when ptt stop is detected on combined GRS

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Enhancements: • Emulator is upgraded with below changes as per ED-137/1C Change 1 specification: ➢ Provides option to include 'Recv-Info' header and negotiate for **Selcal** feature. ➢ Handled negotiation of WG67-Version header. In the case the UAS does not support any version, it SHALL respond with 501 Not Implemented or 405 Not Allowed. ➢ Changed the format to the reason header value for Invalid Multicast values and added a new reason header value for not supporting any of the requested WG67 versions (cause=2016). • Provide option to define Non-VoIP keying source in GRS profile and include Non-VoIP keying source details in NOTIFY to WG67-KeyIn event. • Provide option to define normal and emergency permitted users list in GRS profile. • Option to generate provisional 180/183 responses at GRS. • Added cause codes 2011, 2012, 2013, 2014, and 2015 to release cause configuration under GRS profile. • Enhanced GRS to reject call with "503 Service Unavailable" error when RTP session creation fails. • Handled PTT priority of both Non-VoIP and VoIP sources. • Include RRC request with first RTP packet when PTT is keyed. **Bug Fixes:** • R2S Keepalive packets did not include Radio signaling info as per negotiated "sigtime" value. • Fixed absence of R2S Keepalive packets when RTP packet generation stops after reaching End Of File or transmission duration. • Included Cause and Reason header in 5xx responses. • Included missing "IN IP4" string in multicast SDP attribute. • Removed 'Contact' header from BYE, CANCEL, 200 OK to BYE. • Fixed RTPCore crash on receiving invalid PTT type. • Fixes to handle SUBSCRIBE and NOTIFY transaction using SIP port other than 5060 • Corrected the VIA header to use new Branch Id values for every new NOTIFY transaction. • Corrected the "Subscription-State" header value to "terminated" in NOTIFY message indicating the end of subscription. • Corrected the NOTIFY message templates to remove extra crlf in message body. • Fixed call/transaction rejection with "404 Not Found" when request URI includes SIP port.	**23.4.24**

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<u>Enhancements:</u>

Package built on VS2019.
Includes MAPS™ GUI and framework updates.
Validated against VOTER Version 4.1.30.1 BASELINE SPECIFICATION Volume 1 - GRS Radio ED-137/1C compliance.
Validated against VOTER Version 4.1.30.2 BASELINE SPECIFICATION Volume 2 - VCS Radio ED137/1C compliance.

<u>Bug Fixes:</u>

Includes MAPS™ core fixes to manage a virtual IP address list of more than 72.
Includes parser fixes for SIP headers.
Includes fixes for IncrGlobalVariable utility function data validation
Includes Fix to avoid GetCallStatus timeout in CLI.
Includes script updates with proper impairment label names to sync with CLI APIs.
Includes script updates to fix get_message python API at GRS.
Includes script updates to assign record route port value to default SIP port when it is exclusively not mentioned.
Includes script updates to address combined GRS simulation issues.
Includes fixes to include only necessary SIP headers in Request and Responses as per ED137/1C specification

BYE Transaction request and response omit 'Contact' header.
Includes fixes to include Subject and Priority headers only in INVITE requests and omit in all other SIP methods and responses.

Includes new custom call control script to automate key ptt and squelch cycle sequentially.

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<u>Enhancements:</u>

Included below Python APIs to get ED-137 header values received on the air-to-ground session.

get_rrc_values() // to get RRC header values
get_sqi_values() // to get SQI header values
get_cld_values() // to get CLD header values
get_base_headers() // to get ED-137 base header values
get_mam_values() // to get MAM header values
get_rmm_values() // to get RMM header values

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Enhancements: - MAPS source code upgraded to Visual Studio 2019. - Includes license upgrade to check the warranty. **Bug Fixes:** - Handled sigtime negotiation during call establishment and adding WG67 RTP extension headers to RTP packets as per negotiated sigtime.	**20.5.26**
Enhancements: - Provided option to configure WG67 Version in the profile. - Supports Call duration randomization. - Provided option to include or exclude Receiver Multicast Group and Multicast SDP Attribute. - Provided option to include or exclude Recv Info Header. - Enhanced to display RMM, MAM and RRC values in call graph. - Supports sending RMM continuously based on time interval provided at global configuration at CWP. - Enhanced to display R2S-Local Hold time in Message Sequence in case if the call gets terminated due to keep alive time out error. - Supports setting SCT bit ON, when both PTT and Squelch is ON. - Enhanced to stop speaker, when PTTM is applied. - Provided option to record GLW file Continuously. **Bug Fixes:** - Fix to handle load local profile on the call reception dialog, if the request URI of incoming call does not match exactly when it has SIP port added. - CSeq fix for NOTIFY method. Earlier we used to send CSeq what we received in SUBSCRIBE. - Parse error fix in SIP URI with numeric and dot in between. - GUI changes for not to expand all branches of a node, if that node is disabled. - Fix for not to display "Key Non-VoIP PTT" for Radio-Idle sessions. "Contact" header value import fix for "200 OK" to SUBSCRIBE response. - Fix to handle Multicast call without multicast group value in its address. - Fix to handle multicast calls without multicast SDP attribute. - Fix related to call Authentication, retaining digest parameters for transactions after digest negotiation and incrementing nonce for new transactions within a dialog. "GetRxFileName" fixes for DateTimeFormat, now we are appending sequence number with milliseconds timestamp. - Fix to apply PTTS, PTTM and SCT on all the session of a radio if it is applied on single session also. - Fix to continue selcal transmission once it is overridden by higher priority PTT.	**19.12.26**

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Enhancements: • Supports CLI mode with Python API's • CLI support for MAPS requires purchase of PKS170 license • Provides Option to define multiple Impairment profile with different values **Bug Fixes:** • RTP Core fix to support different sampling rates for telephone-event codec • Other minor bug fixes	**8.9.19**
Enhancements: • Provides option to select cause for BYE and 603 messages • Supports sending out of dialog NOTIFY message with active connections at GRS • Supports software license feature	**8.3.19**
Enhancements: • Provides counters for Pilot-Pilot induced SCT and Controller-Pilot induced SCT (synchronous transmission) • Supports sending out of dialog SUBSCRIBE and NOTIFY messages. • Supports writing Call Statistics to file. • Provides option to include customer specific profiles and scripts. **Bug Fixes** • Delay in incoming audio play back is fixed. • Added missing 'Recv-Info' header in initial INVITE message for calls supporting SELCAL feature. • Now change in radio call type after receiving Re-INVITEs are updated to call graph. • Provided option to stop sending RTP extended header CLD (Climax Time Delay).	**8.1.9**
• Radio version number is updated to 'radio.02' in WG67-Version SIP header in all requests and responses. • RRCE shared and non-shared variables are reflected properly to VCS. • The event package includes the frequency id (fid) parameter in order to indicate the current frequency used. If fid is not available, it will be set to '000.000'. Whenever GRS changes its fid, SIP NOTIFY message with changed fid will be sent to all subscribed CWPs. • PTT-id values 60, 61 and 62 are reserved for non-VoIP keying sources and SHALL not be assigned to VoIP keying sources; PTT-id value 63 is reserved for the optional SELCAL feature. • Optional SDP attribute <disconnect mode> with value 'NoFreqDisconn' can be added in INVITE to request GRS not to disconnect session in case of frequency id change. • SELCAL feature: Supports sending SELCAL tones using SIP INFO message. • Supports Radio Receiver Multicast Operation. • Supports simulating Test PTT. • Simulates non-VoIP source PTT keying. • Measurement Answer Message (MAM) includes the GRS receiver delay Ts2.	**7.8.22**

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Enhancements: • Simulate Dynamic Delay Compensation messages RMM and MAM. • Supports sending simultaneous squelch on all connected sessions to a Radio. • Supports sending simultaneous squelch on all connected sessions of selected multiple Radios. • Sample script provided to perform automated periodic PTT on AG calls. • Sample script provided to perform automated periodic SQU on AG calls. • Option to define multiple traffic profiles. • Support sending audio using microphone and playing audio to speaker on multiple sessions.	**7.7.14**
Enhancements: • Supports Multiple Radio Simulation: ➢ MAPS ED137 Radio has been enhanced to simulate multiple radios within a single instance. ➢ IP addresses specified in the Radio profile will be created as virtual IP addresses on the system NIC interface. Profile Editor also includes Color coding option to create different profiles. ➢ Sorting and color codes are used to group all calls associated to individual radios in Call Reception window. • Enhanced to handle Linked Session Management: ➢ The Linked Session functionality provides to the GRS endpoint the opportunity to detect SIP sessions which are coming from the same user but from different equipment (i.e. different IP Address) to guarantee higher service availability. ➢ So, 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. It will treat the linked sessions and treat them 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. ➢ User has the option to enable or disable linked session management for each radio. • Button options in Call Generation and Reception window to apply Events on an ongoing call ➢ "Receive Traffic" user button to provide an option to receive traffic (i.e. record to file, detect digits/tones) for every active call. ➢ ReInvite user button is provided to send a Re-Invite from CWP to update an active SIP session. ➢ Enhanced to update Signal Quality Information (SQI) in run-time for an active call. ➢ Radio handles the incoming PTT and prioritize accordingly	**7.2.3**