If this is the first time use of MAPS™ SIP application, then it is recommended to follow all the steps explained in MAPS-SIP-Quick-Install-Guide to install MAPS™ SIP application before proceeding with the steps below.

Pre-requisites

The Quick check-out procedure explained in this document requires **a PC with 2 NIC cards** to perform loopback testing using a single MAPS™ SIP application.

If the PC has only one NIC card, then the MAPS™ SIP can be tested against any DUT in the network in a similar manner, with destination IP address and port set to that of the DUT's.

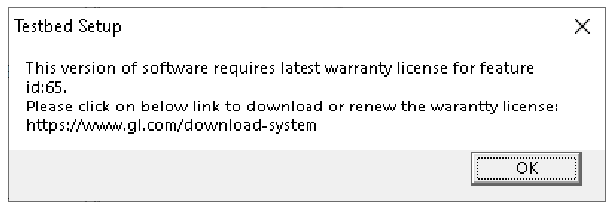
We assume that the following purchased licenses are installed on the test PC as explained in the **MAPS™ SIP Quick Install Guide**.

- ➢ PKS120 (MAPS for SIP)
- ➢ PKS102 (PacketGen RTP Soft Core) *

***Note:** Additional licenses may be required for optional applications. Please verify that all licenses purchased are installed.

Note:

- The "**Warranty Error**" as shown in the screenshot may be prompted, when the user tries to start the testbed, either the Warranty licenses are not installed, or the license is expired
- Ensure that the warranty license (**GLSupportWarrantyLicenseInstaller.exe**) is installed and also confirm that **PKS120** (**MAPS™ SIP**) is listed in Warranty Application List. Refer to **MAPS-SIP-Quick-Install-Guide**.



Quick Check Out Procedure

For **self-test** of MAPS™ SIP application, you may prepare **a single PC with 2 NIC cards**, one as source and other as destination. Ensure that both NIC cards are within the same subnet, assigned proper free IP addresses available in the subnet, and connected to a switch. If the system is connected to a LAN, contact your system administrator to avoid IP address conflicts before you perform the steps below. If the PC has only one NIC card, then the MAPS™ SIP can be tested against any DUT in the network in a similar manner, with destination IP address and port set to that of the DUT's.

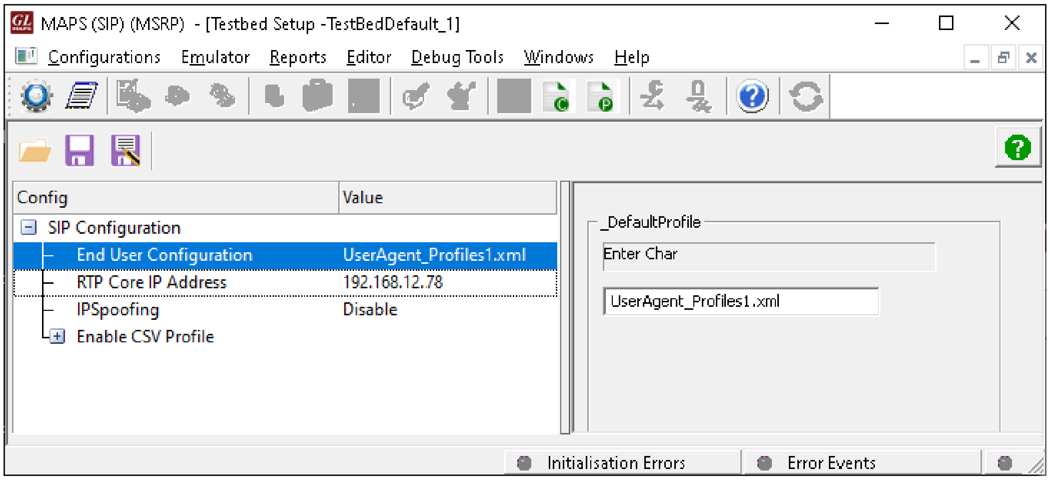
For illustration purposes, we assume the IP address for the NIC cards are configured as 192.xx.xx.78 (NIC #1) and 192.xx.xx.74 (NIC #2). Invoke two instances of **MAPS™ SIP** application.

The configurations below allow **first instance** of MAPS™ SIP to use **NIC 1** IP address as source and the **NIC 2** IP address as destination endpoint. Similarly, the **second instance** of MAPS™ SIP to use **NIC 2** IP address as source and the **NIC 1** IP address as destination endpoint to simulate SIP calls.

Configuring MAPS™ SIP instance as UAS

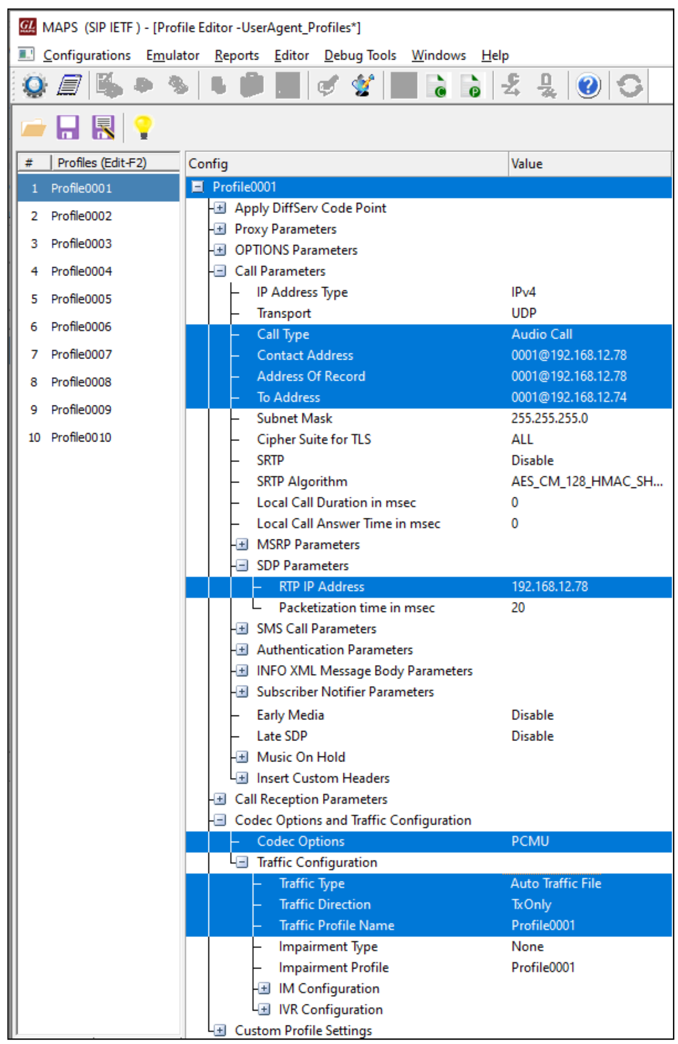
- Right-click on **MAPS-SIP** short-cut icon MAPS-SIP created on the desktop and select '**Run as Administrator**'. This instance of MAPS™ is configured for **Call Reception.**
- By default, **Testbed Setup** window loaded with **TestBedDefault** configuration is displayed. Verify the following settings.
  - ➢ Select **End User Configuration** parameter and change the profile name to **UserAgent_Profiles1.xml**
  - ➢ Set the **RTP Core IP** address to the **NIC #1 IP Address** of the system on which the RTP Core should be invoked
  - ➢ By default, **IPSpoofing** option is disabled

- ➢ Click on **Save As** icon  and save the testbed setup as **TestBedDefault_1.xml**



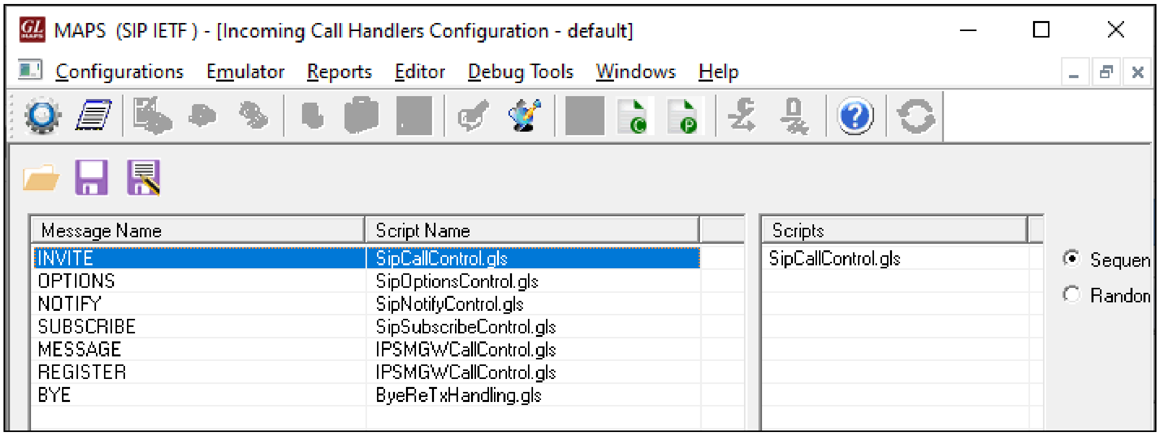
- From **MAPS-SIP** main window, select **Editor → Profile Editor** to invoke the **Profile Editor** window loaded with default **UserAgent_Profiles.** From the left pane, choose **Profile0001** profile. Verify the following settings**:**

  - ➢ Set **Call Type → Audio Call**
  - ➢ Edit **Contact Address →** 0001@192.168.12.78 (Enter the source **NIC 1 IP address** as SIP URI here)
  - ➢ Edit **Address of Record →** 0001@192.168.12.78 (Enter the source **NIC 1 IP address** as SIP URI here)
  - ➢ Edit **To Address →** 0001@192.168.12.74 (Enter the destination **NIC 2 IP address** as SIP URI here)
  - ➢ Edit **RTP IP Address → 192.168.12.78** (Enter the source **NIC 1 IP address** here)
  - ➢ Scroll down to **Codec Options and Traffic Configurations** and select **Codec** as **PCMU**
  - ➢ Set **Traffic Type** to **Auto Traffic File** type, and **Traffic Direction** to **TxOnly**
  - ➢ By default, **Traffic Profile Name** is set to **Profile0001**



- Click on **Save As** icon option and save the profile as **UserAgent_Profiles1** and close the **Profile Editor** window.

- On the same MAPS™ SIP instance, select **Configuration → Incoming Call Handler Configuration** to invoke the **Incoming Call Handlers Configuration** window. Verify that the **SipCallControl.gls** script is loaded against the **INVITE** message. Close the window.

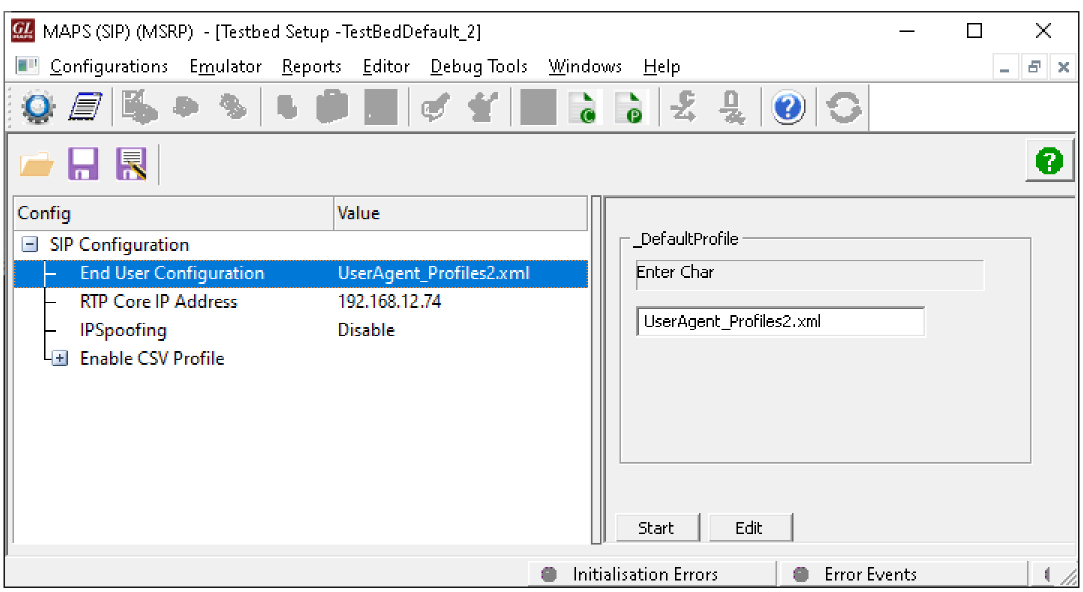


Configuring MAPS™ SIP Instance as UAC

- Right-click on **MAPS-SIP** short-cut icon  created on the desktop and select '**Run as Administrator**'. This instance of MAPS™ is configured for **Call Generation.**

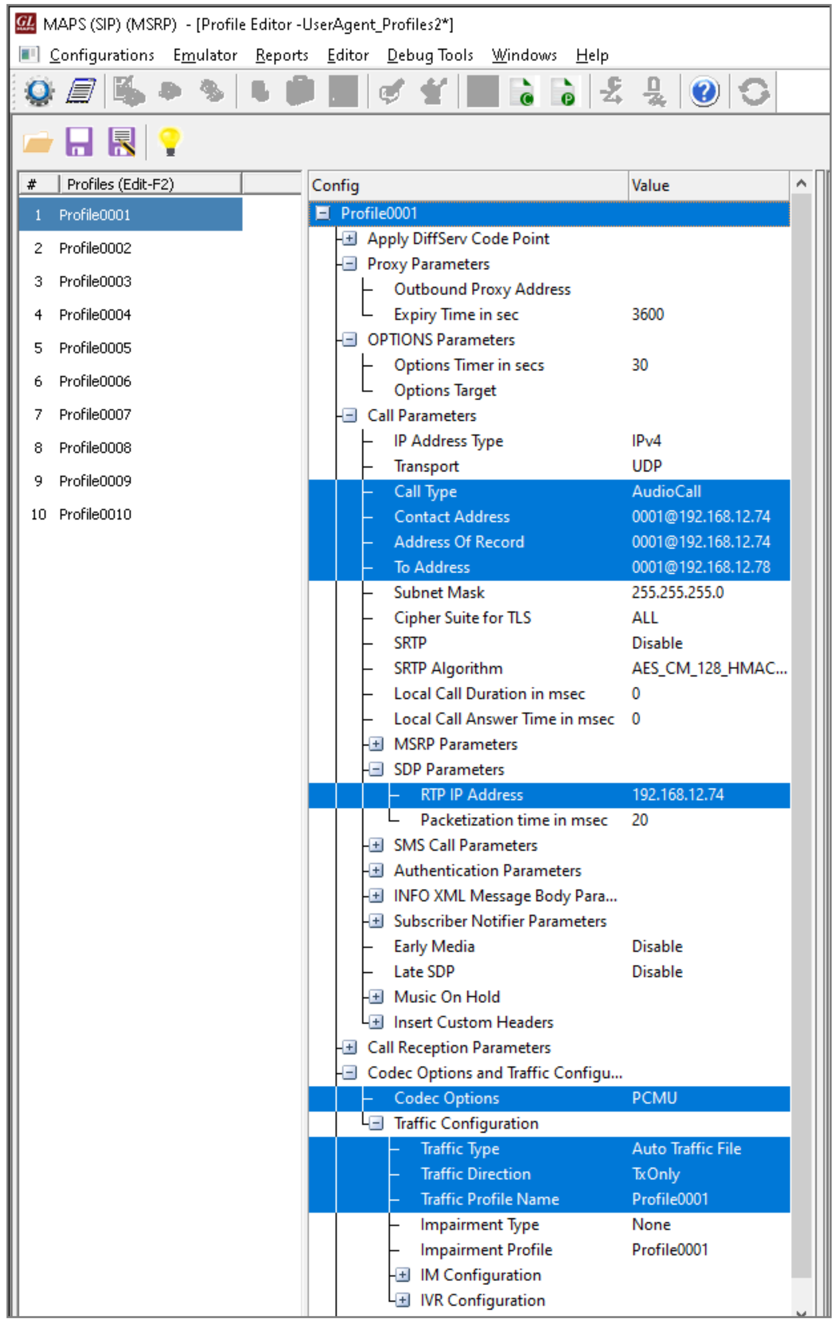
- By default, **Testbed Setup** window is loaded with **TestBedDefault** configuration is displayed. Verify the following settings:

  - Select **End User Configuration** parameter and change the profile name to **UserAgent_Profiles2.xml**
  - Set the **RTP Core IP** address to NIC #2 IP Address of the system on which the **RTP Core** should be invoked
  - By default, **IPSpoofing** option is disabled
  - Click on **Save As** icon  and save the testbed setup as **TestBedDefault_2.xml**



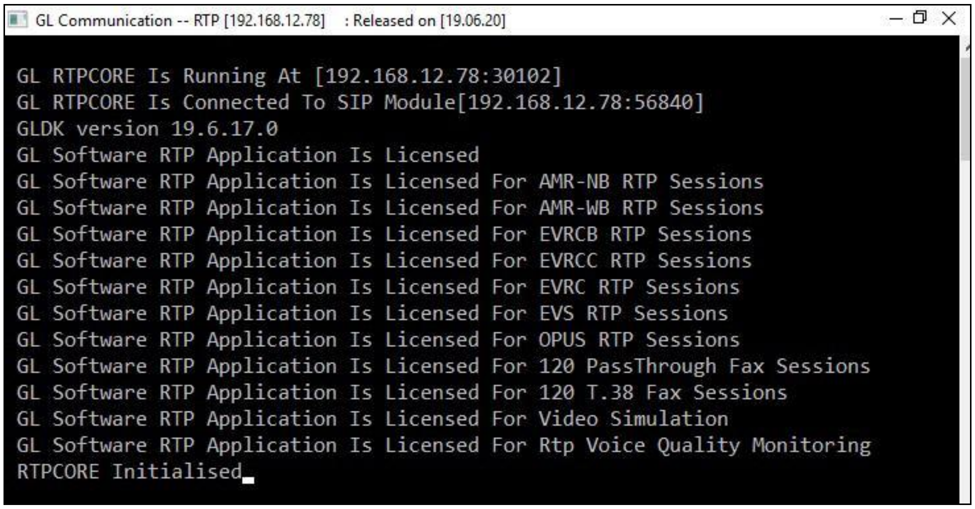
- From **MAPS-SIP** main window, select **Editor → Profile Editor** to invoke the Profile Editor window loaded with default **UserAgent_Profiles.** From the left pane, choose **Profile0001** profile. Verify the following settings:

  - Set **Call Type → Audio Call**
  - Edit **Contact Address → 0001@192.168.12.74** (Enter the source **NIC 2 IP address** as SIP URI here)
  - Edit **Address of Record → 0001@192.168.12.74** (Enter the source **NIC 2 IP address** as SIP URI here)
  - Edit **To Address → 0001@192.168.12.78** (Enter the destination **NIC 1 IP address** as SIP URI here)
  - Edit **RTP IP Address → 192.168.12.74** (Enter the source **NIC 2 IP address** IP Address here)
  - Scroll down to **Codec Options and Traffic Configurations** and select **Codec** as **PCMU**
  - Set **Traffic Type** to **Auto Traffic File** type, and **Traffic Direction** to **TxOnly**
  - By default, **Traffic Profile Name** is set to **Profile0001**



- Click **Save As** icon  and save the profile as **UserAgent_Profiles2** and close the Profile Editor window.

- Click on **Start** button in the testbed setup of both the MAPS™ instances and wait for the 2 RTP-Core console windows to appear in the taskbar. If the SIP/RTP Core console does not invoke with the MAPS™ Testbed start-up, refer to **Troubleshoot** section explained in *https://www.gl.com/Brochures/Brochures/Installation-Instructions-for-Dongle-Programs.pdf*.



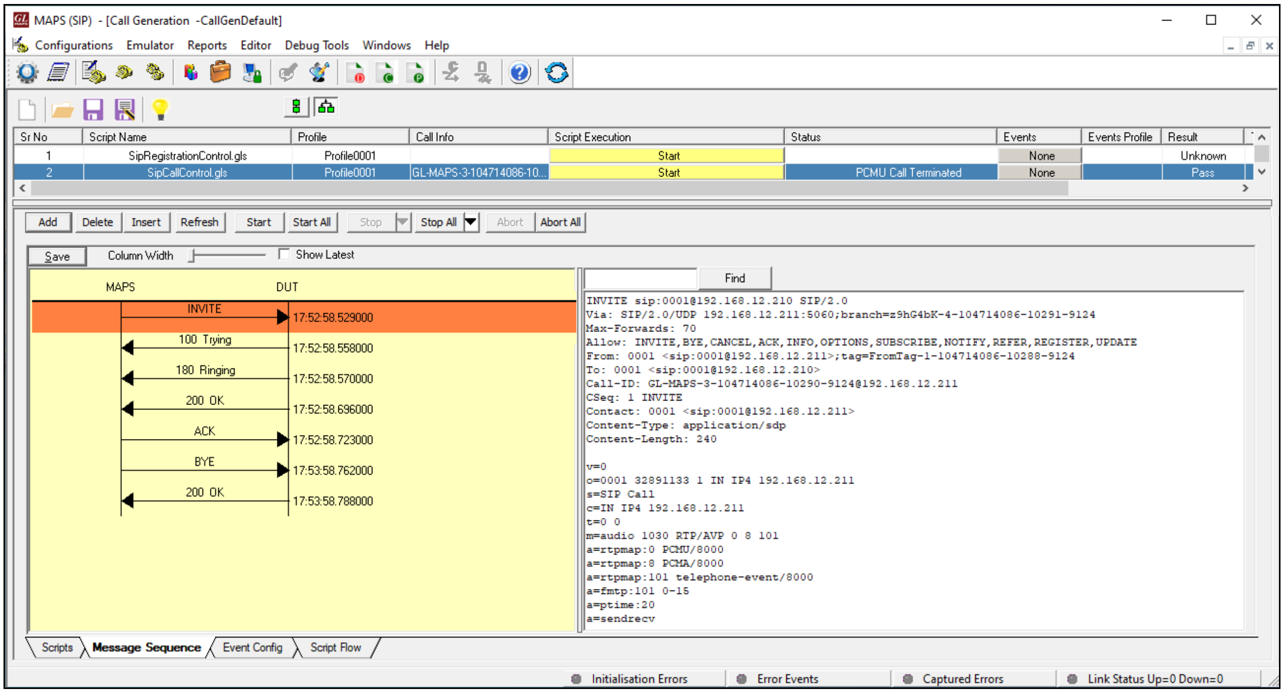
- From any of the MAPS™ SIP instance, click on **Call Generation** icon on to invoke the **Call Generation** window.
- By default, user will observe call instances loaded with **SipCallControl.gls** and **SipRegistrationControl.gls** scripts and **Profile0001** profile in the **Call Generation** window.
- Select the call instance loaded with **SipCallControl.gls** script and **Profile0001** profile and click on Start button to execute the script.

> **Note:**
> - User should double-click under **Profile** and select **Profile0001** against the script **SipCallControl.gls** for the first time.

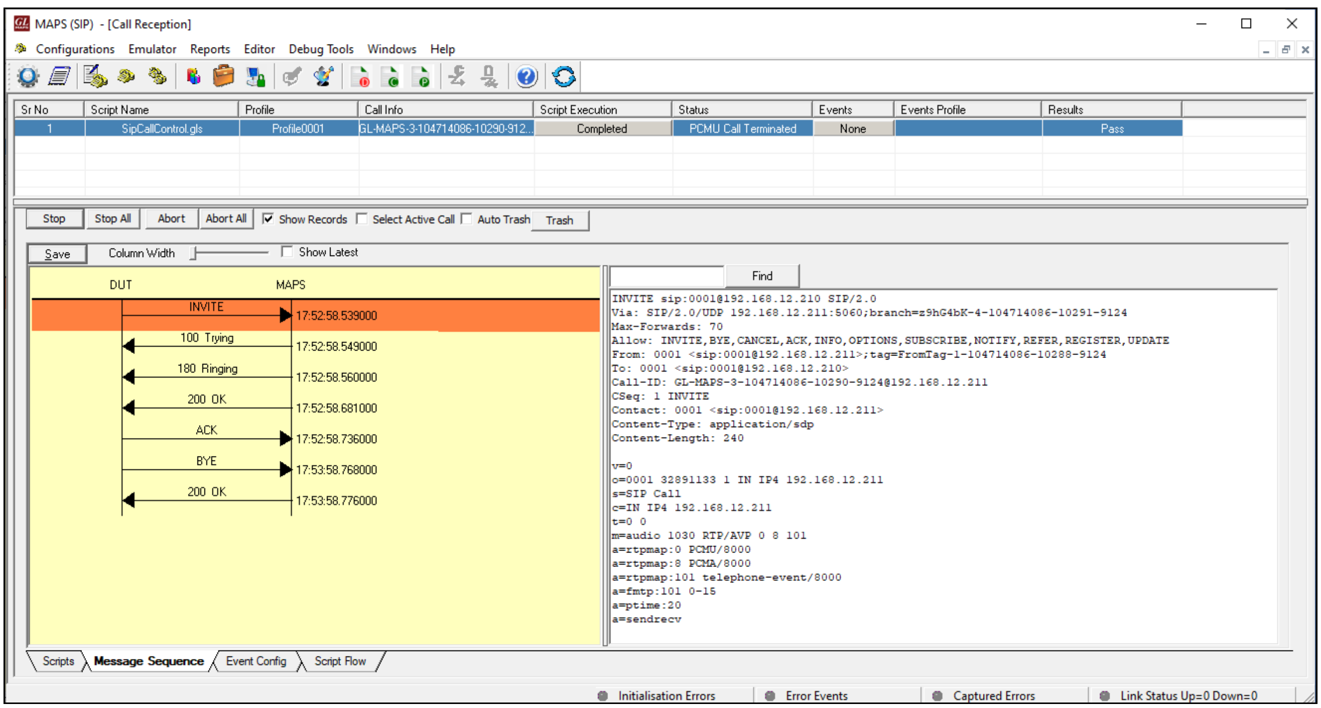
- Wait till call gets terminated, verify the **Message Sequence Flow** by selecting the call objects at both generation and reception end.

> **Note:**
> - Click on the **Message Sequence** tab available on the bottom of the GUI, to observe the ladder diagram for the established calls.

- Select any message in the ladder diagram and observe the respective decode message on the right pane for the respective message.



- On the second MAPS™ instance, click on **Call Reception** icon and observe the calls being received.



- This completes the functional verification of **MAPS™ SIP** application.

- For any technical queries, contact **GL Communications Inc**.