

If this is your First-Time-Use of MAPS™ SIP application, then we recommend you 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 (fax machine) 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 following the procedure as explained in the MAPS™ SIP Quick Install Guide.

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

Licenses for T.38 Fax Simulation

- PKS211 (T.38 Fax Simulation) *

Licenses for T.30 Pass Through Fax Simulation

- PKS200 (RTP Pass Through Fax Emulation) * Additionally requires one of the following licenses as per the number of sessions requirement
 - PKS202 - 2 Fax Ports
 - PKS203 - 8 Fax Ports
 - PKS204 - 30 Fax Ports
 - PKS205 - 60 Fax Ports
 - PKS206 - 120 Fax Ports

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

Quick Check Out Procedure

Simulation of T.30 pass through mode fax and T.38 fax (UDPTL) can be verified by configuring 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 (fax machine) 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 that the IP address for the NIC cards are configured as 192.xx.xx.213 (NIC #1) and 192.xx.xx.212 (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 T.38/ T.30 pass through fax over successfully established SIP calls.

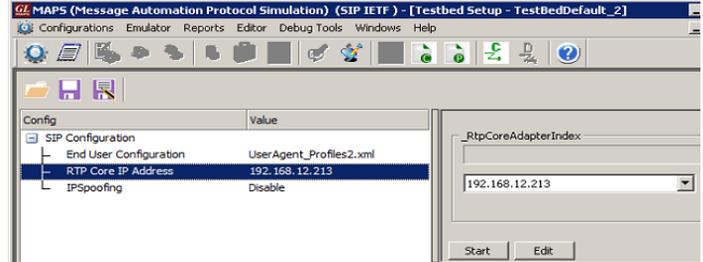
First MAPS™ SIP instance as UAS



- 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 Reception**.

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

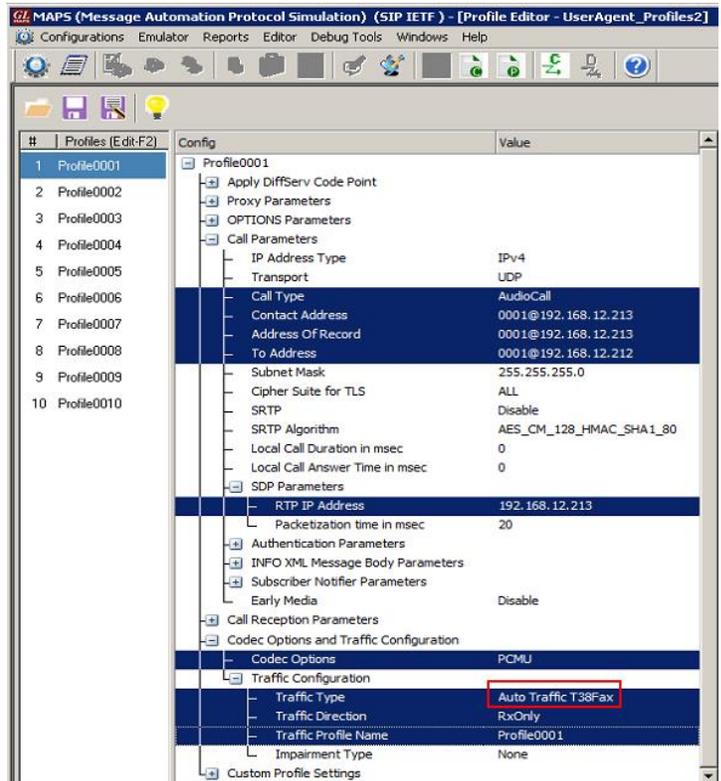
- Select **End User Configuration** parameter and change the profile name to **UserAgent_Profiles2.xml**
- Set the RTP Core IP address to the **NIC #1 IP Address** of the system on which the RTP Core is invoked.
- By default, **IPspoofing** option is disabled
- Click  **Save As** option and save the testbed setup as **TestBedDefault_2** configuration file.



- From **MAPS-SIP** main window, select **Editor** → **Profile Editor**. Profile Editor window is invoked. Profile Editor window is invoked loaded with default **UserAgent_Profiles**. From the left pane, choose **Profile0001** profile, and verify the settings.

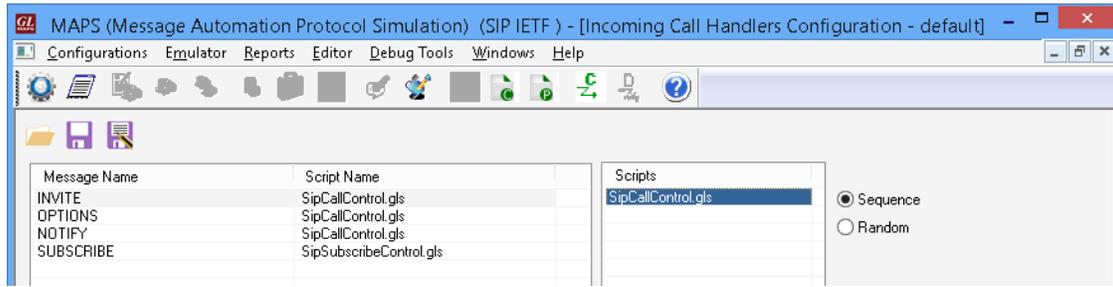
- Verify the settings in **Profile0001** profile:

- Set **Call Type** → **Audio Call**
- Edit **Contact Address** → 0001@192.168.12.213 (Enter the source **NIC 1 IP address** as SIP URI here)
- Edit **Address of Record** → 0001@192.168.12.213 (Enter the source **NIC 1 IP address** as SIP URI here)
- Edit **To Address** → 0001@192.168.12.212 (Enter the destination **NIC 2 IP address** as SIP URI here)
- Edit **RTP IP Address** → 192.168.12.213 (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 T38 Fax** type [**Note:** Set to **Auto Traffic Fax** type for T.30 pass through Fax Simulation]
- Set **Traffic Direction** to **RxOnly**.
- By default, **Traffic Profile Name** is set to **Profile0001**.



- Click  **Save As** save the profile as **UserAgent_Profiles2** file.

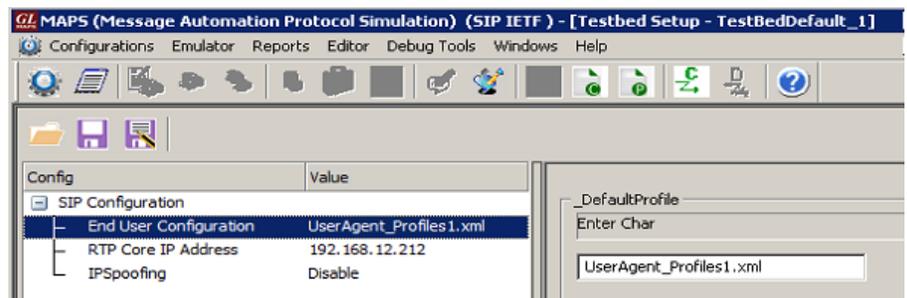
- On the same MAPS™ SIP instance, from **Configuration** → invoke **Incoming Call Handler Configuration** window. Verify that the **SipCallControl.gls** script is loaded against the **INVITE** message. Close the window.



Second 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 displayed loaded with **TestBedDefault** configuration file. Verify the following settings:

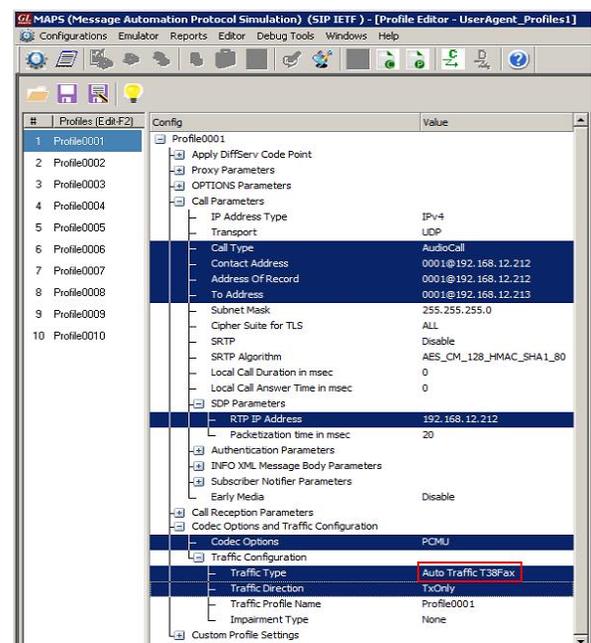
- Select **End User Configuration** parameter and change the profile name to **UserAgent_Profiles1.xml**.
- Set the RTP Core IP address to NIC #2 IP Address of the system on which the RTP Core should be invoked.
- By default, **IPspoofing** option is disabled.



- Click  **Save As** option and save the testbed setup as **TestBedDefault_1.xml** file.

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

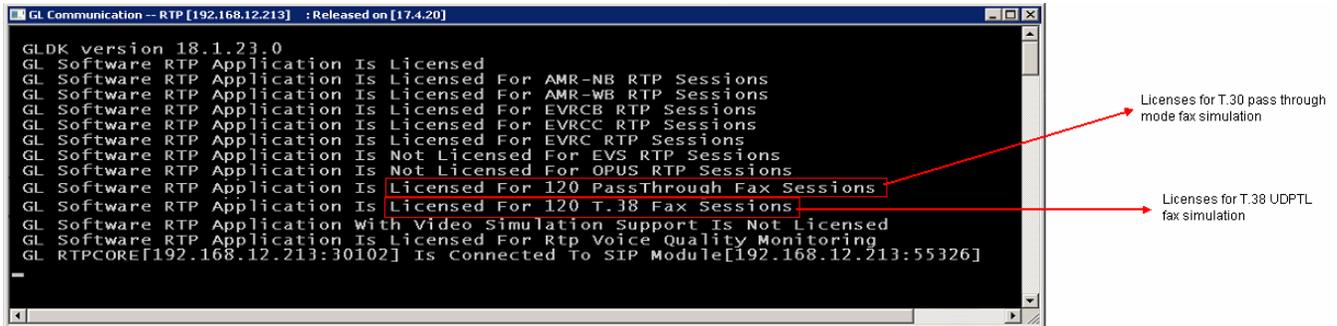
- Set **Call Type** → **Audio Call**
- Edit **Contact Address** → **0001@192.168.12.212** (Enter the source **NIC 2 IP address** as SIP URI here)
- Edit **Address of Record** → **0001@192.168.12.212** (Enter the source **NIC 2 IP address** as SIP URI here)
- Edit **To Address** → **0001@192.168.12.213** (Enter the destination **NIC 1 IP address** as SIP URI here)
- Edit **RTP IP Address** → **192.168.12.212** (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 T38 Fax** type [**Note:** Set to



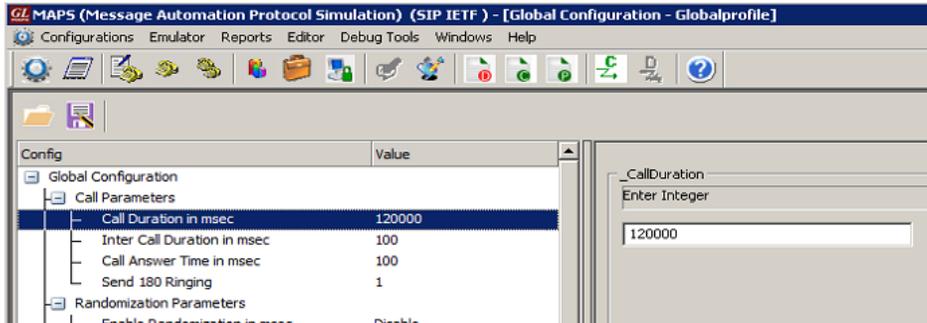
Auto Traffic Fax type for T.30 pass through Fax Simulation]

- Set *Traffic Direction* to *TxOnly*.
- By default, *Traffic Profile Name* is set to *Profile0001*.

- Click  **Save As** option and save the profile as **UserAgent_Profiles1** file. Exit from Profile Editor window.
- Click **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>.



- On the second MAPS™ Call generation instance, select **Configurations** menu → **Global Configuration**. Set the **Call Duration in msec** to **120000**, which is required for the test.

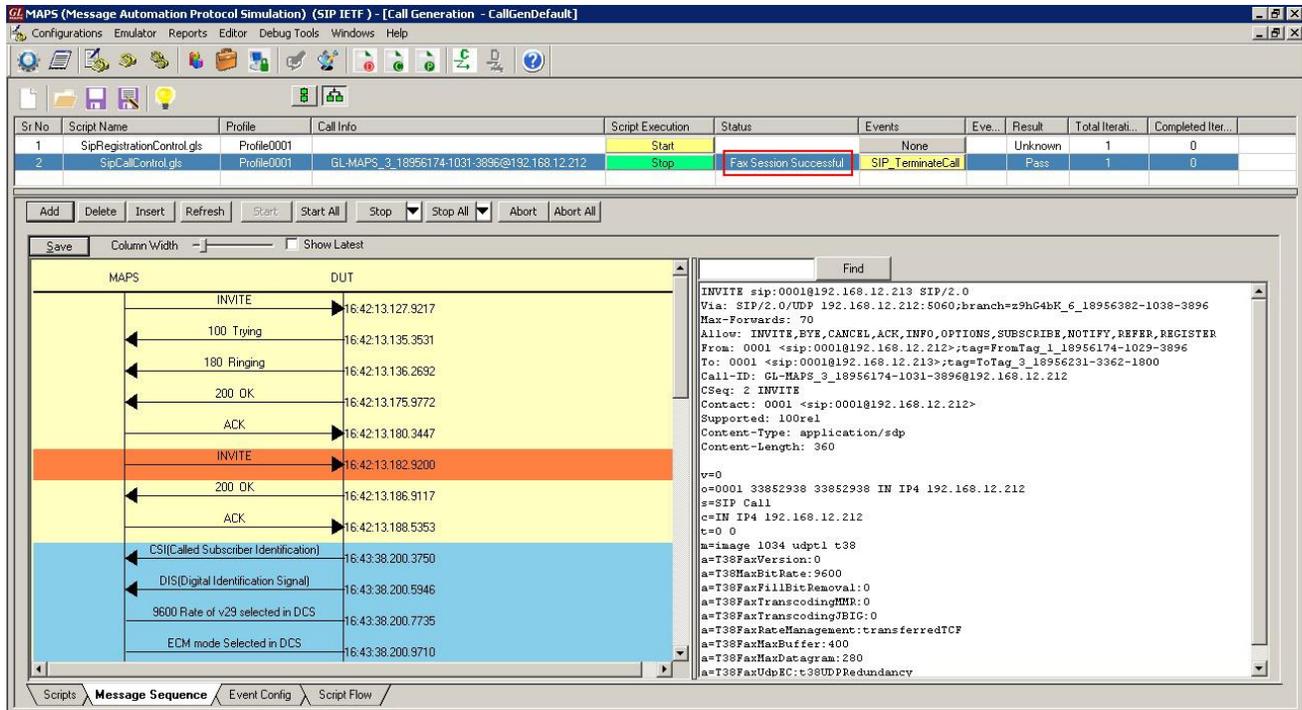


- From the MAPS™ SIP instance configured for call generation, click on **Call Generation**  icon on the main window to invoke the **Call Generation** window.

- By default, you 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  **Start** button to execute the script.
- Note:** If the profile is not loaded by default, then the user should double-click under **Profile** column and select **Profile0001** from the drop-down and set it against the script **SipCallControl.gls**.

- Wait till call gets terminated, verify the **Message Sequence Flow** by selecting the call objects at both generation and reception end.
- Select any message in the ladder diagram and observe the respective decode message on the right pane for the respective message.

Note: Uncheck the 'Show Latest' box available in the middle toolbar, to scroll and view the complete message sequence flow.



Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Event	Result	Total Iterati...	Completed Iter...
1	SipRegistrationControl.gls	Profile0001		Start		None		Unknown	1	0
2	SipCallControl.gls	Profile0001	GL-MAPS_3_18956174-1031-3896@192.168.12.212	Stop	Fax Session Successful	SIP_TerminateCall		Pass	1	0

Message Sequence (MAPS to DUT):

- INVITE → 16:42:13.127.9217
- 100 Trying ← 16:42:13.135.3531
- 180 Ringing ← 16:42:13.136.2692
- 200 OK ← 16:42:13.175.9772
- ACK → 16:42:13.180.3447
- INVITE → 16:42:13.182.9200
- 200 OK ← 16:42:13.186.9117
- ACK → 16:42:13.188.5353
- CSI(Called Subscriber Identification) ← 16:43:38.200.3750
- DIS(Digital Identification Signal) ← 16:43:38.200.5946
- 9600 Rate of v29 selected in DCS ← 16:43:38.200.7735
- ECM mode Selected in DCS ← 16:43:38.200.9710

SIP Log:

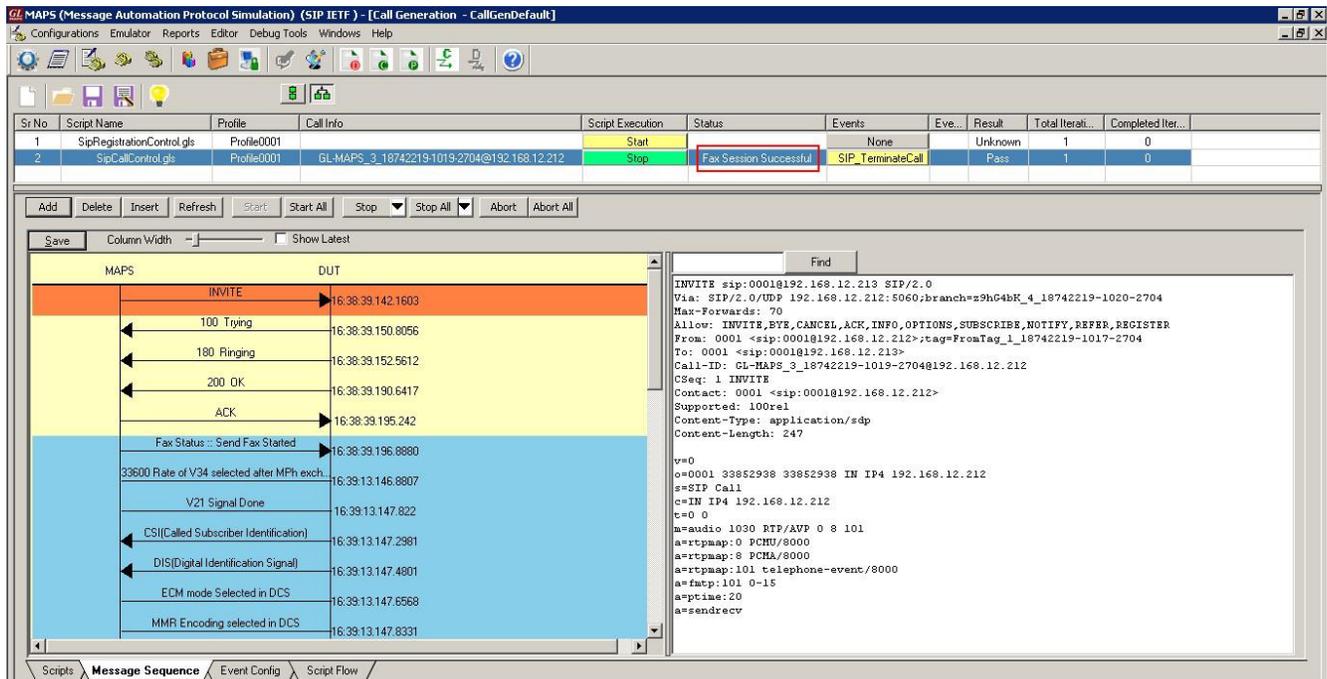
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INVITE sip:0001@192.168.12.213 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.212:5060;branch=z9hG4bK_6_18956382-1038-3896
Max-Forwards: 70
Allow: INVITE,BYE,CANCEL,ACK,INFO,OPTIONS,SUBSCRIBE,NOTIFY,REFER,REGISTER
From: 0001 <sip:0001@192.168.12.212>;tag=FromTag_1_18956174-1029-3896
To: 0001 <sip:0001@192.168.12.213>;tag=ToTag_3_18956231-3362-1800
Call-ID: GL-MAPS_3_18956174-1031-3896@192.168.12.212
CSeq: 2 INVITE
Contact: 0001 <sip:0001@192.168.12.212>
Supported: 100rel
Content-Type: application/sdp
Content-Length: 360

v=0
o=0001 33852938 33852938 IN IP4 192.168.12.212
s=SIP Call
c=IN IP4 192.168.12.212
t=0 0
m=image 1034 udpt1 t:38
a=T38FaxVersion:0
a=T38FaxBitRate:9600
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCP
a=T38FaxHasBuffer:400
a=T38FaxMaxDatagram:280
a=T38FaxUDPECC:t:38UDPRedundancy
    
```

T.38 Fax Simulation Call Generation

- The following message sequence is displayed for T.30 pass through fax generation.



Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Event	Result	Total Iterati...	Completed Iter...
1	SipRegistrationControl.gls	Profile0001		Start		None		Unknown	1	0
2	SipCallControl.gls	Profile0001	GL-MAPS_3_18742219-1019-2704@192.168.12.212	Stop	Fax Session Successful	SIP_TerminateCall		Pass	1	0

Message Sequence (MAPS to DUT):

- INVITE → 16:38:39.142.1603
- 100 Trying ← 16:38:39.150.8056
- 180 Ringing ← 16:38:39.152.5612
- 200 OK ← 16:38:39.190.6417
- ACK → 16:38:39.195.242
- Fax Status :: Send Fax Stated → 16:38:39.196.8880
- 33600 Rate of V34 selected after MPh exch. ← 16:39:13.146.8807
- V21 Signal Done ← 16:39:13.147.822
- CSI(Called Subscriber Identification) ← 16:39:13.147.2981
- DIS(Digital Identification Signal) ← 16:39:13.147.4801
- ECM mode Selected in DCS ← 16:39:13.147.6568
- MMR Encoding selected in DCS ← 16:39:13.147.8331

SIP Log:

```

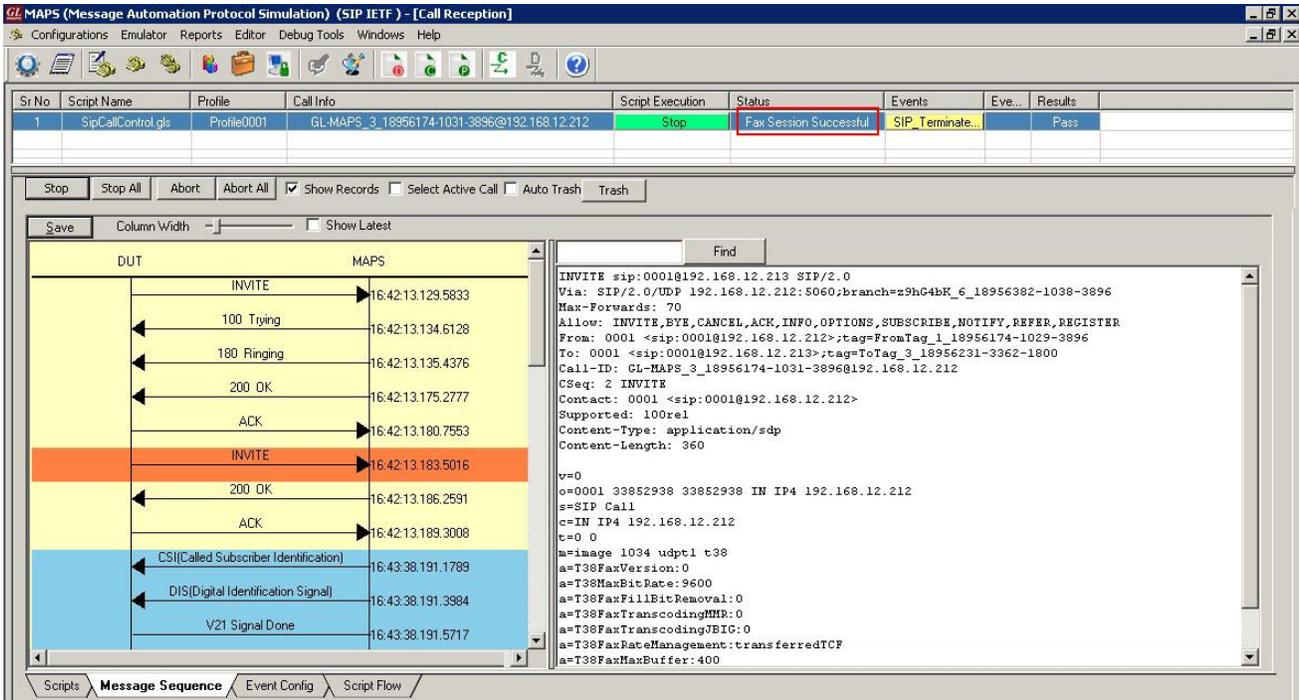
INVITE sip:0001@192.168.12.213 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.212:5060;branch=z9hG4bK_4_18742219-1020-2704
Max-Forwards: 70
Allow: INVITE,BYE,CANCEL,ACK,INFO,OPTIONS,SUBSCRIBE,NOTIFY,REFER,REGISTER
From: 0001 <sip:0001@192.168.12.212>;tag=FromTag_1_18742219-1017-2704
To: 0001 <sip:0001@192.168.12.213>
Call-ID: GL-MAPS_3_18742219-1019-2704@192.168.12.212
CSeq: 1 INVITE
Contact: 0001 <sip:0001@192.168.12.212>
Supported: 100rel
Content-Type: application/sdp
Content-Length: 247

v=0
o=0001 33852938 33852938 IN IP4 192.168.12.212
s=SIP Call
c=IN IP4 192.168.12.212
t=0 0
m=audio 1030 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fatp:101 0-15
a=rtpmime:audio
a=sendrecv
    
```

T.30 Pass Through Fax Simulation Call Generation

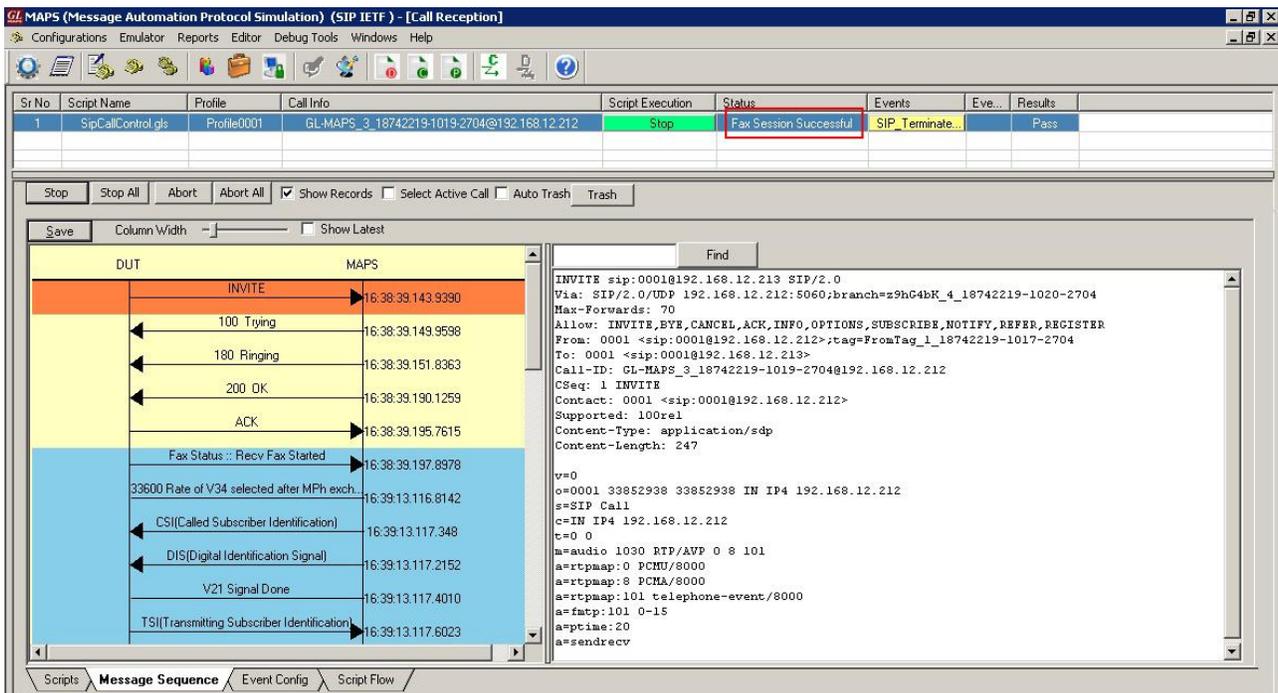
- On the first MAPS™ instance main window, click on **Call Reception**  icon and observe the calls being received.

Note: Uncheck the **'Show Latest'** box available in the middle toolbar, to scroll and view the complete message sequence flow.



T.38 Fax Simulation Call Reception

- The following message sequence is displayed for T.30 pass through fax reception



T.30 Pass Through Fax Simulation Call Reception

- The following is the sample **Recorded Fax (*.tif)** file which is saved in the C:\Program Files\GL Communications Inc\MAPS-SIP\FaxFiles\Recv folder on the MAPS™ SIP call reception side.

Note: Same sample Fax (*.tif) file is transmitted for both T.30 pass through mode and T.38 fax simulation.



Sample Recorded Fax File