MITEL – SIP CoE

Technical
Configuration Notes

Configure MiVoice Business 7.2 with MBG for use with Twilio SIP Trunking

APRIL 2016

SIP COE 16-4940-00441

TECHNICAL CONFIGURATION NOTES

Mitel

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Configure MiVoice Business 7.2 for use with Twilio SIP Trunking
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Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the MiVoice Business (MiVB) 7.2 to connect to Twilio SIP trunking. Different components can be configured in various configurations depending on your VoIP solution. This document covers a basic setup with required option setup.

Interop History

Version	Date	Reason
1	April 2016	Initial Interop with Twilio SIP trunking and MiVoice Business

Interop Status

This Interop of Twilio with MiVoice Business 7.2 has been given a Compatible Certification status. This SIP trunk will be included in the SIP CoE Reference Guide.

COMPATIBLE	The most common certification which means MiVoice Business has been tested and/or validated by the Mitel SIP CoE team. Product support will provide all necessary support related to the interop, but issues unique or specific to the 3rd party will be referred to the 3rd party as appropriate.

Software & Hardware Setup

The table below provides the hardware and software specifications used to generate SIP audio calls, both point to point and conference calls, using Twilio SIP trunking connected to MiVoice Business 7.2.

Manufacturer	Variant	Software Version
Mitel	MiVoice Business	Release 7.2 (13.2.0.17)
Mitel	MiVoice Border Gateway (Trunking)	v9.1.1.41
Mitel	MiVoice Border Gateway (Teleworker)	v9.0.27.0
Mitel	NuPoint Unified Messenging	v17.0.0.24.01
Mitel	MiCollab Audio, Web and Video Conferencing	v5.0.3.33
Mitel	53xx Series IP Sets	v06.03.01.05
Mitel	68xx Series SIP Sets	v4.2.0.181
Twilio	SIP Trunking Service	As of April 2016

Tested Features

The table below provides an overview of the features tested during the Interoperability test cycle and not a detailed view of the test cases. Please see the SIP Trunk Side Interoperability Test Plan APTest **608** for detailed test cases and results.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call through SIP service provider and their PSTN gateway, call holding, transferring, conferencing, busy calls, long calls durations, variable codec	☑
Automatic Call Distribution	Making calls to an ACD environment with RAD treatments, Interflow and Overflow call scenarios and DTMF detection.	☑
NuPoint Voicemail	Terminating calls to a NuPoint voicemail boxes and DTMF detection.	☑
Packetization	Forcing the Mitel MIVB to stream RTP packets through its E2T card at different intervals, from 10ms to 60ms	N/A
Personal Ring Groups (PRG)	Receiving calls through MiVoice Business and their PSTN gateway to a personal ring group. Also moving calls to/from the prime member and group members.	☑
Teleworker	Making and receiving a call through MiVoice Business and their PSTN gateway to and from Teleworker extensions.	☑
Video	Making and receiving a call through MiVoice Business with video capable devices.	N/A
Fax	Use of G.711 for fax calls.	☑

☑ - No issues found X - Issues found, cannot recommend to use ⚠ - Issues found

Device Limitations and Known Issues

This is a list of problems or not supported features when using MiVB with Twilio SIP trunking.

Feature	Problem Description
Basic Call	Twilio only supports G.711.
Packetization	Twilio only supports 20ms packetization rate.
Fax	Twilio does not support T.38

Network Topology

This diagram shows how the testing network is configured for reference.

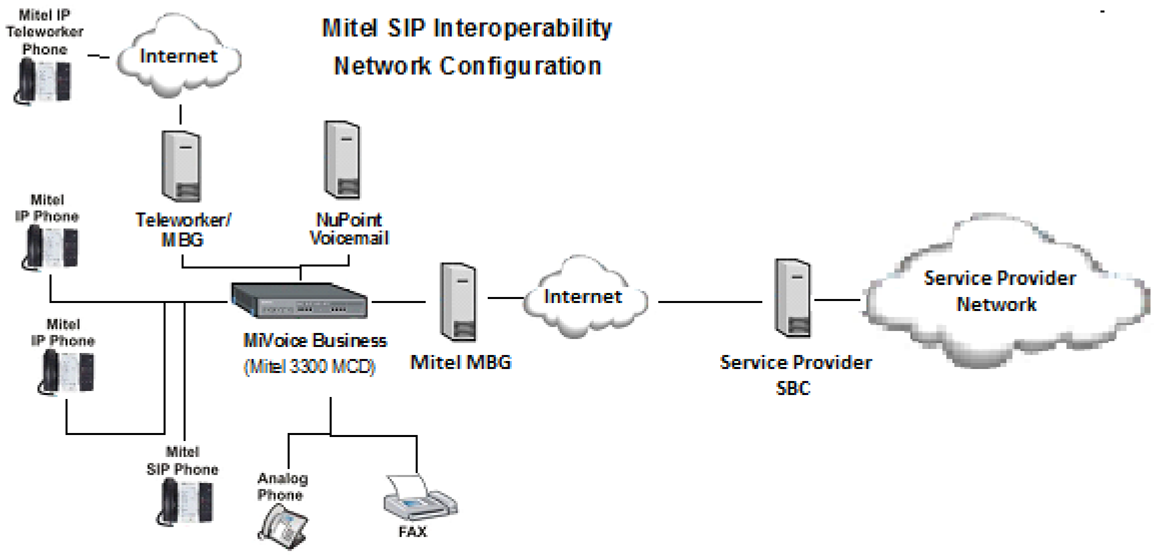

Figure 1 – Network Topology

Configuration Notes

This section is a description of how the SIP Interop was configured. These notes should give a guideline how a device can be configured in a customer environment and how the MiVB programming was configured in our test environment.

Disclaimer: Although Mitel has attempted to setup the interop testing facility as closely as possible to a customer premise environment, implementation setup could be different onsite. YOU MUST EXERCISE YOUR OWN DUE DILIGENCE IN REVIEWING, planning, implementing, and testing a customer configuration.

MiVoice Business Configuration Notes

The following information shows how to configure a MiVoice Business 7.2 to interconnect with Twilio SIP trunking.

Network Requirements

- There must be adequate bandwidth to support the VoIP network. As a guide, the Ethernet bandwidth is approx 85 Kb/s per G.711 voice session and 29 Kb/s per G.729 voice session (assumes 20ms packetization). As an example, for 20 simultaneous SIP sessions, the Ethernet bandwidth consumption will be approx 1.7 Mb/s for G.711 and 0.6Mb/s. Almost all Enterprise LAN networks can support this level of traffic without any special engineering. Please refer to the MiVoice Business Engineering guidelines on the Mitle eDocs Website (http://edocs.mitel.com) for further information.
- For high quality voice, the network connectivity must support a voice-quality grade of service (packet loss <1%, jitter < 30ms, one-way delay < 80ms).

Assumptions for MiVoice Business Programming

- The SIP signaling connection uses UDP on Port 5060.

Licensing and Option Selection – SIP Licensing

Ensure that the MiVB is equipped with enough SIP trunking licenses for the connection to Twilio. This can be verified within the License and Option Selection form.

Enter the total number of licenses in the SIP Trunk Licences field. This is the maximum number of SIP trunk sessions that can be configured in the MiVB to be used with all service providers, applications and SIP trunking devices.

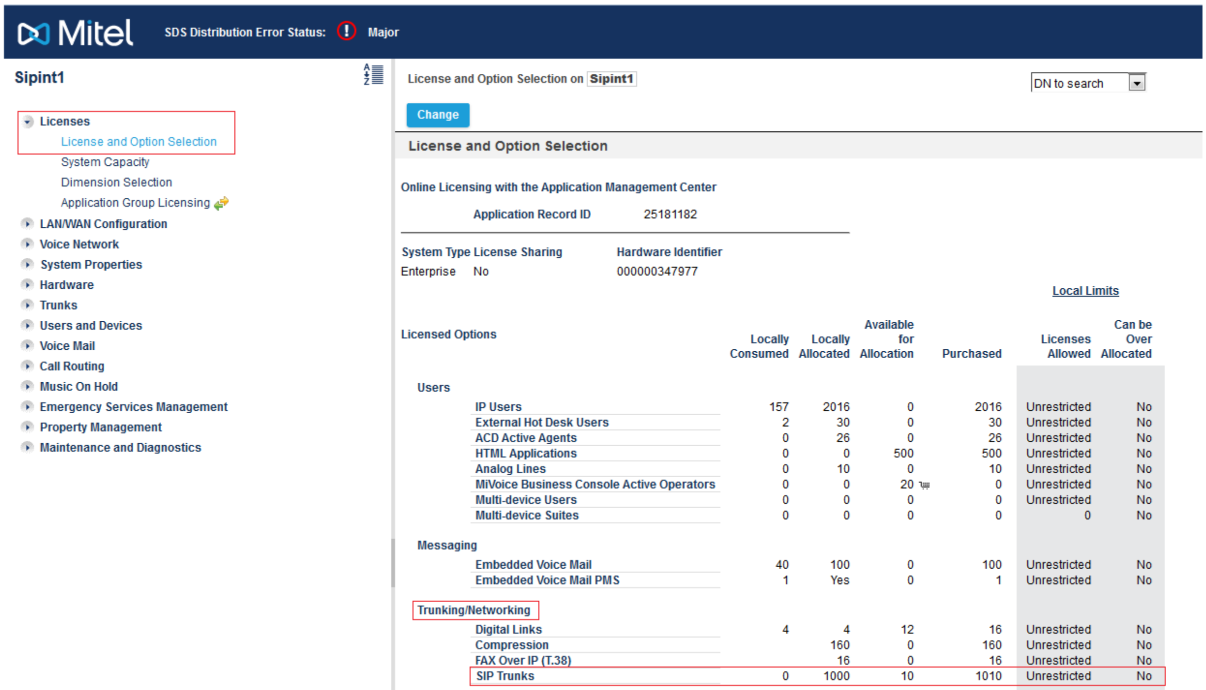

Figure 2 – License and Option Selection

Class of Service Options

The **Class of Service Options form** is used to create or edit a Class of Service and specify the associated options. Classes of Service, identified by Class of Service numbers, are referenced in the Trunk Attributes form for SIP trunks.

Many different options may be required for your site deployment, but ensure that **Public Network Access via DPNSS** Class of Service Option is configured for all devices that make outgoing calls through the SIP trunks in the MiVB.

- Public Network Access via DPNSS set to **Yes**

If use FAX equipment ensure that the following options are enabled,

- Campon Tone Security/FAX Machine set to **Yes**
- Busy Override Security set to **Yes**

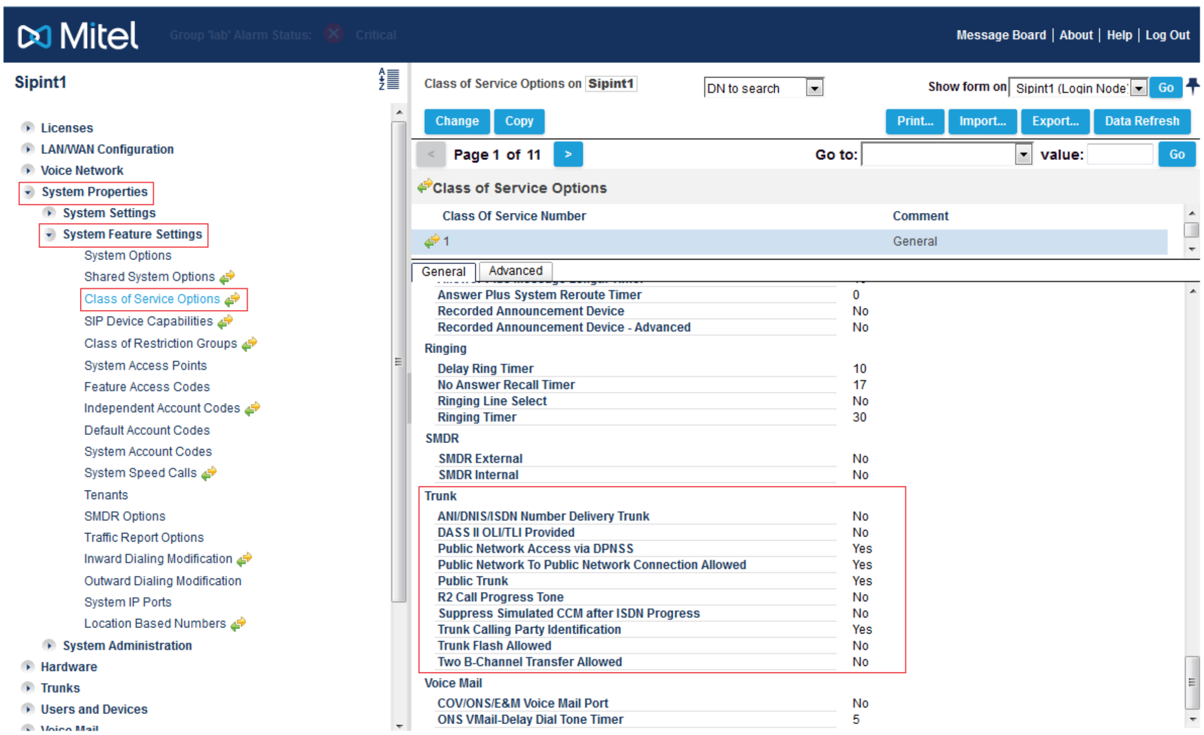

Figure 3 – Class of Service Options

Network Elements

Create a network element for the SIP peer Twilio as shown in **Figure 4**. The IP address or FQDN will be provided by Twilio.

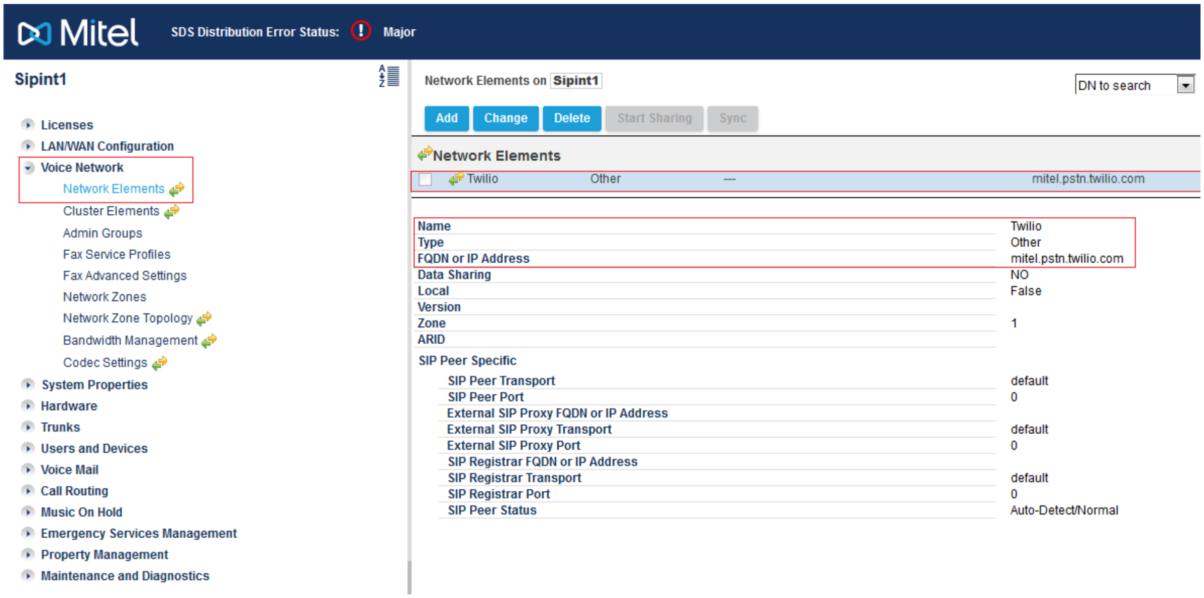

Figure 4 – Network Element for SIP Peer

Create a network element for the Mitel MBG as shown in **Figure 5**. The IP address entered here is that of the MBG.

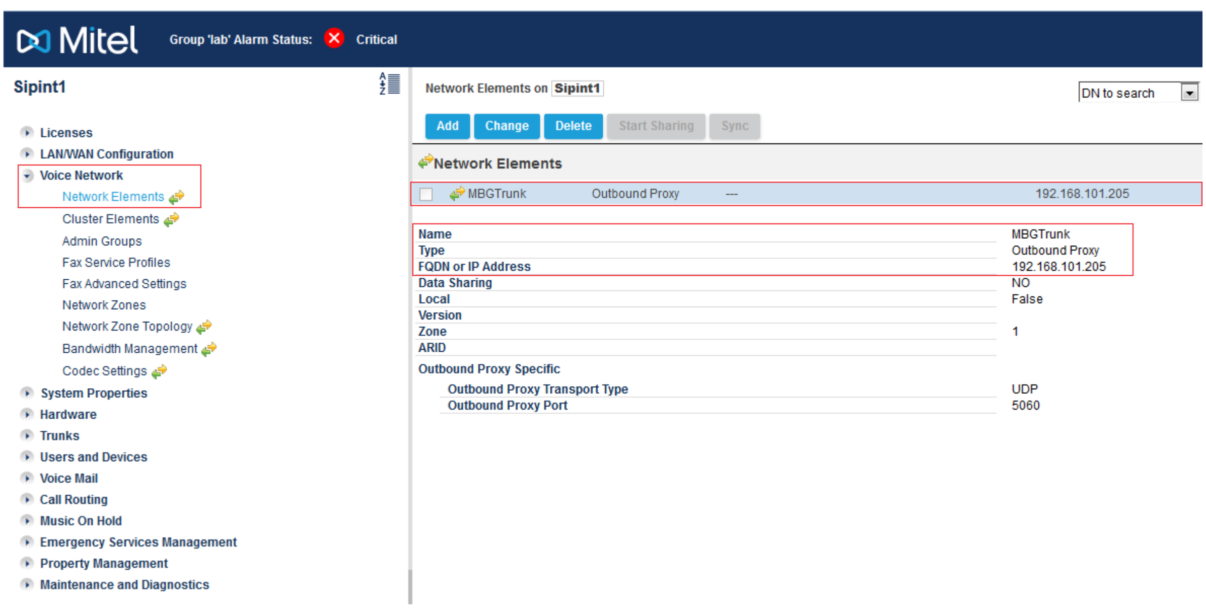

Figure 5 – Network Element for MBG

Trunk Attributes

Use **Trunk Attributes** form to configure Trunk Service Number. In this example, the Trunk Service Number **12** will be used to direct incoming calls to an answer point in MiVoice Business.

Program the **Non-dial In** or **Dial In Trunks** (DID) according to the site requirements and what type of service was ordered from your service provider.

Figure 6 below shows configuration for incoming DID calls. The MiVoice Business will absorb the first 4 digits of the DID number received from Twilio's SIP trunk leaving 7 digits for MiVoice Business to translate and ring the 4-digit extension.

For example, the Twilio SIP trunk delivers numbers 1-613-519-2701 through the SIP trunk to MiVoice Business, which will absorb the first 4 digits (1-613) leaving the remaining 7 digits (519-2701) to route the call. Number 519-2701 must be programmed as a valid dialable number in the MiVoice Business, ie. System Speed Call number to associate 519-2701 with an extension in MiVB. Please refer to MiVoice Business 7.2 System Administration documentation for further programming information.

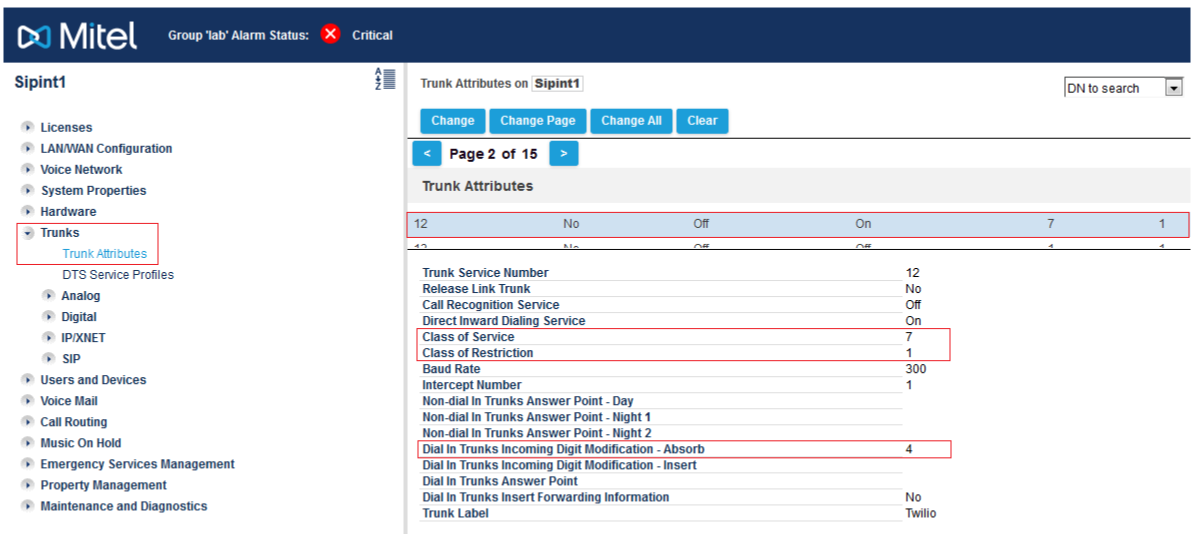

Figure 6 – Trunk Service Assignment

SIP Peer Profile

The recommended connectivity via SIP Trunking does not require additional physical interfaces. IP/Ethernet connectivity is part of the base MiVoice Business Platform. The SIP Peer Profile should be configured as shown in **Figures 7 - 12**.

Basic tab:

Network Element: The selected SIP Peer Profile needs to be associated with the previously created Twilio Network Element.

Registration User Name: Twilio does not support SIP trunk registration at this time so this field was left blank.

Address Type: Use 'IP Address'.

Outbound Proxy Server: Select the Network Element previously configured for the outbound proxy server (MBG),

Trunk Service: Enter the trunk service previously configured.

SMDR: If Call Detail Records (CDR) are required for SIP Trunking, the SMDR Tag should be configured (by default there is no SMDR and this field is left blank).

Maximum Simultaneous Calls: Configure this entry to be the maximum number of SIP trunks provided by Twilio.

Username and Password: Twilio supports digest based authentication for outgoing calls only, inbound is not supported at this time. If you chose to use outbound call authentication fill in these two fields with the username and password you create with Twilio.

NOTE: Ensure the remaining SIP Peer profile configuration options are similar to the screenshots below.

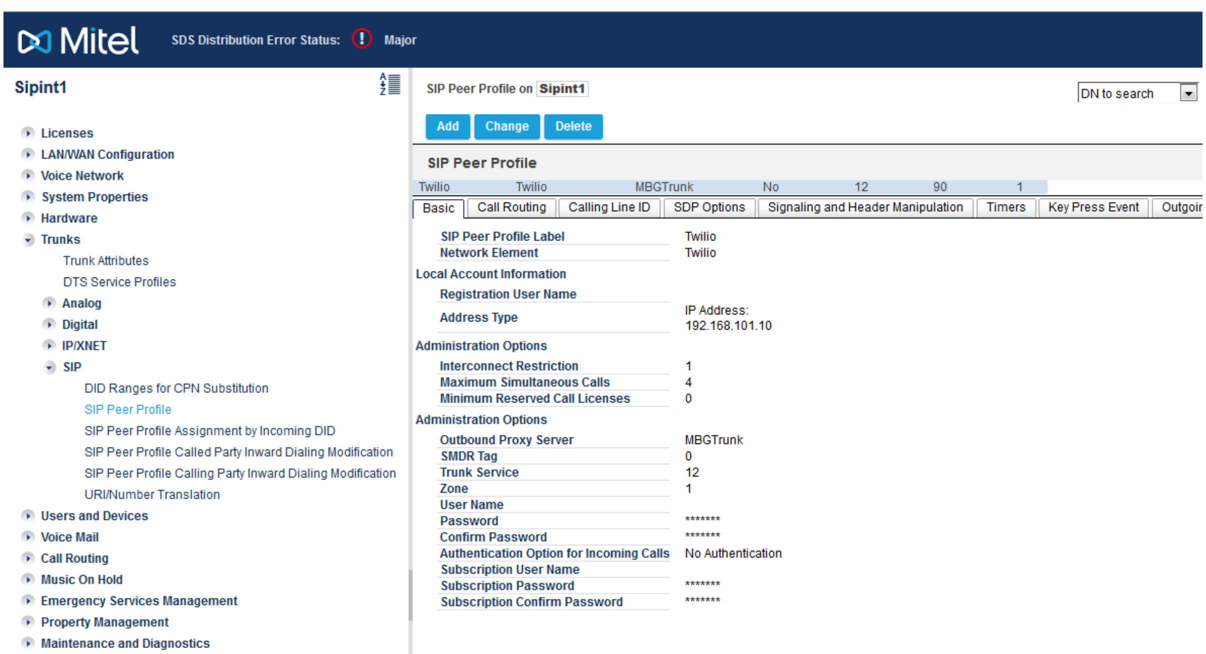

Figure 7 – SIP Peer Profile - Basic

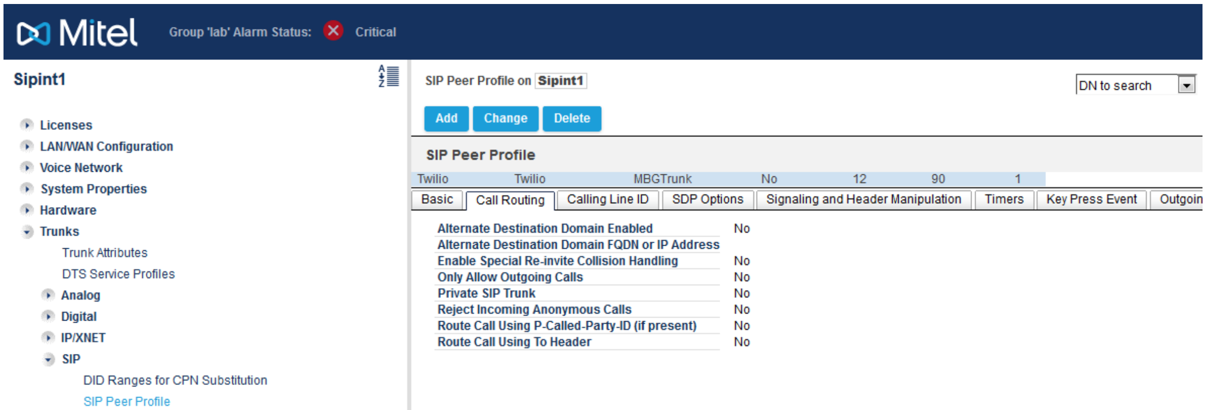

Figure 7 – SIP Peer Profile - Call Routing

Calling Line ID tab:

The ‘Default CPN’ (Calling Party Number) is applied to all outgoing calls; unless there is a match in the ‘Outgoing DID Ranges’ of the SIP Peer profile. This number must be one of the numbers supplied by Twilio.

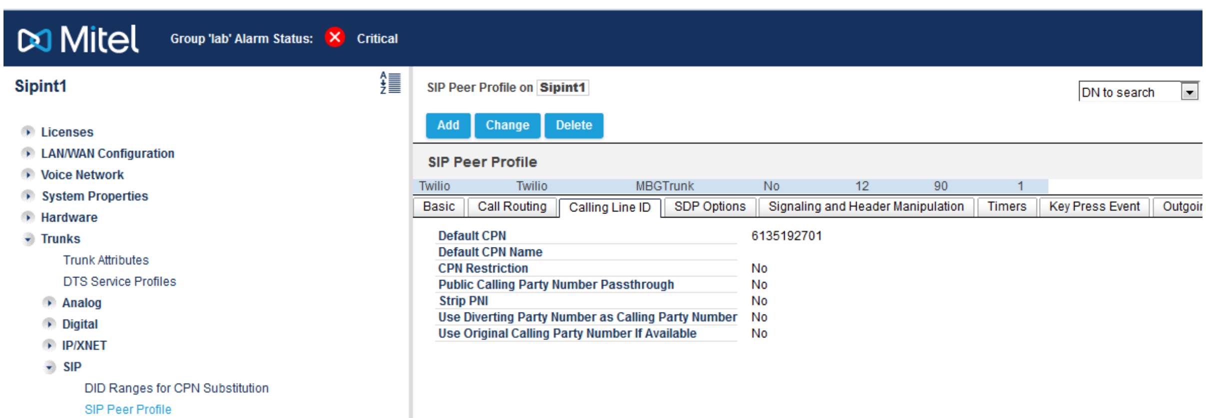

Figure 8 – SIP Peer Profile - Calling Line ID

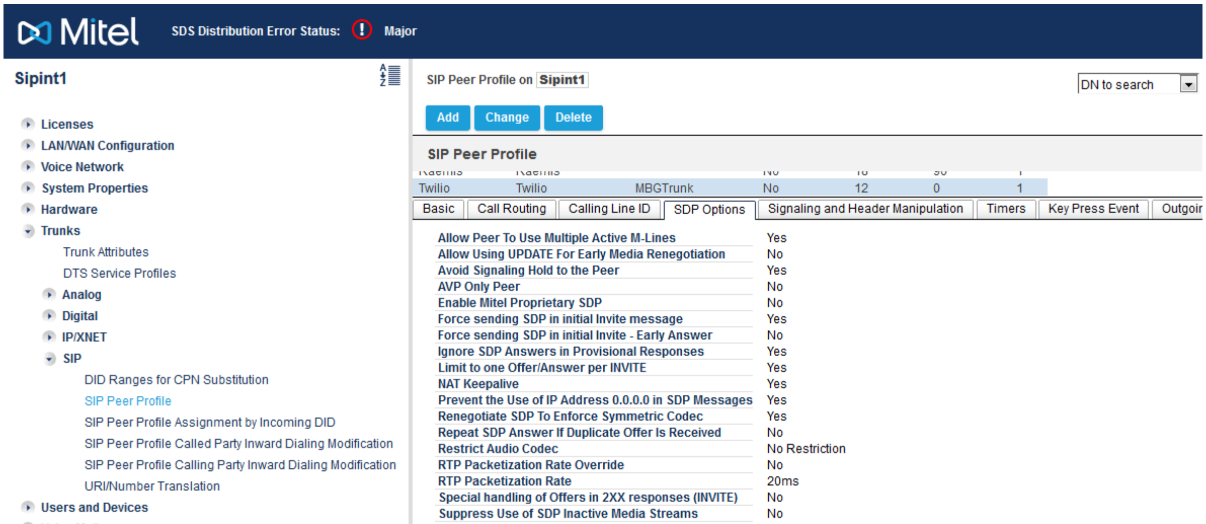

Figure 9 – SIP Peer Profile - SDP Options

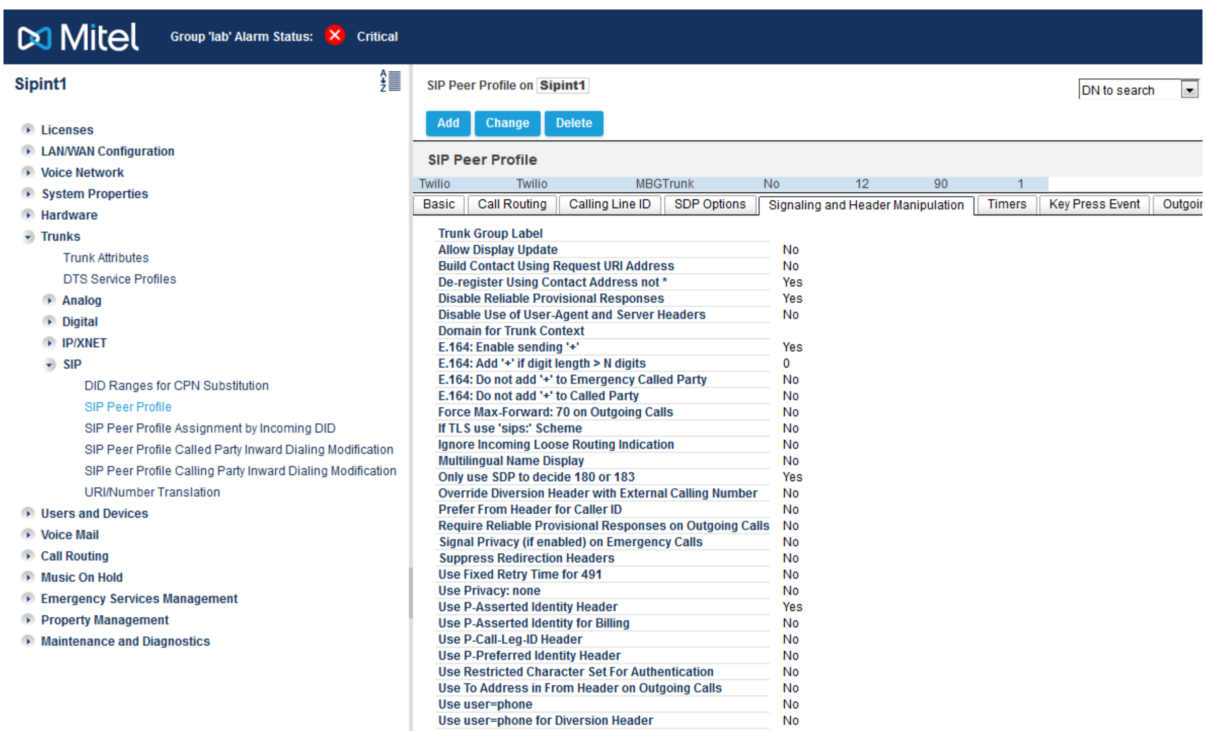

Figure 10 – SIP Peer Profile - Signaling and Header Manipulation

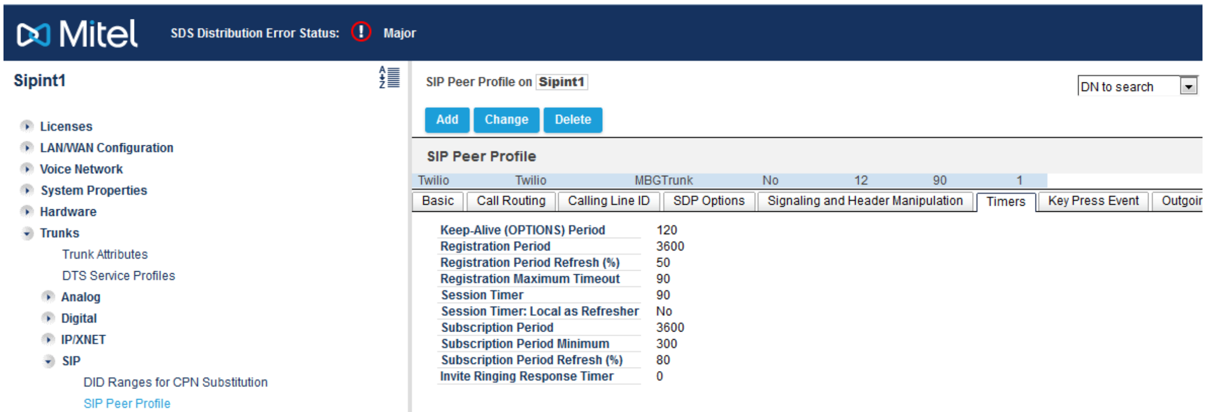

Figure 11 – SIP Peer Profile – Timers

SIP Peer Profile Assignment by Incoming DID

This form is used to assign incoming digits from Twilio. DID range numbers assigned by Twilio are associated to a particular SIP peer.

Enter one or more telephone numbers. The maximum number of digits per telephone number is 26. You can enter a mix of ranges and single numbers (for example, "6135554000-6135554400, 6135554500"). The entire field width is limited to 60 characters.

Use a comma to separate telephone numbers and ranges. Use a dash (-) to indicate a range of telephone numbers. The first and last characters cannot be a comma or a dash. If the numbers do not fit within the 60 character maximum, you can create a new entry for the same profile.

Use a '*' to reduce the number of entries that need to be programmed. This is a type of "prefix identifier", and cannot be used as a range with '-'. For example, the string "11*" would be used to associate a peer with any number in the range from 110 up to the maximum digits per telephone number (In this case, 11999999999999999999999999.) Note that the string "11" by itself would not count as a match, as the '*' represents 1 or more digits.

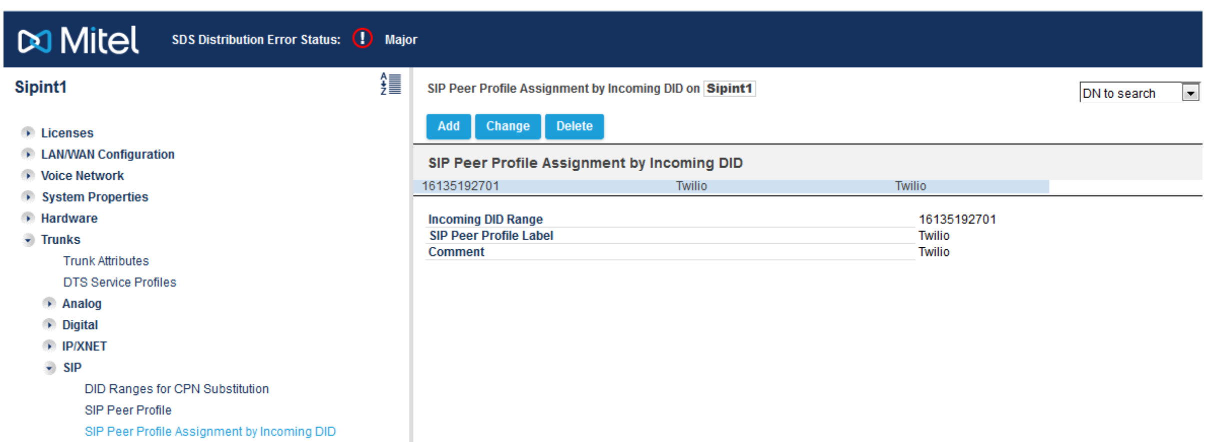

Figure 13 – SIP Peer Profile – Assignment by Incoming DID

ARS Digital Modification Plans

Ensure that ARS Digit Modification for outgoing calls on the SIP trunk to Twilio absorbs or inject additional digits according to your dialing plan. In this example, we will be absorbing 3 digits, for example 910 prefix to dial out.

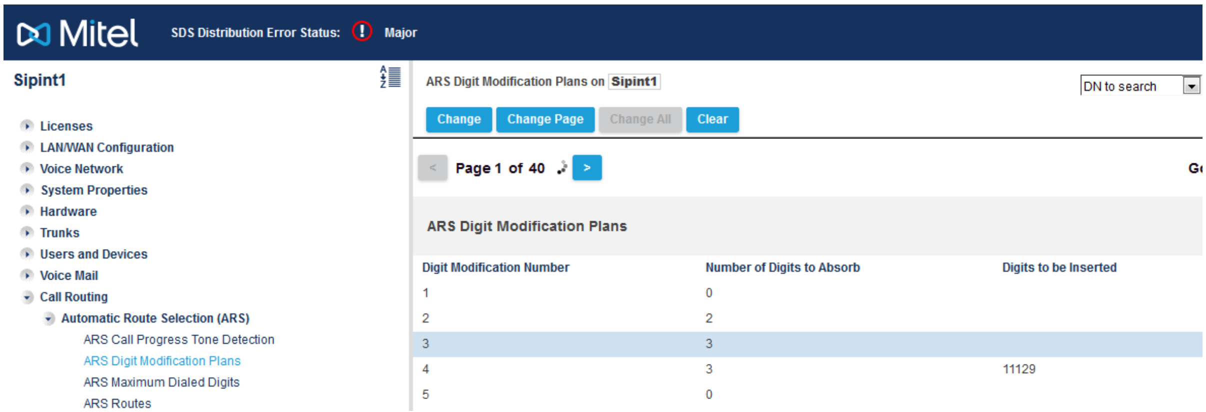

Figure 14 – ARS Digit Modification Plans

ARS Routes

Create a route for SIP Trunks connecting to Twilio. In this example, the SIP trunk is assigned to Route Number **16**. Choose **SIP Trunk** as a routing medium and choose the **SIP Peer Profile** and **Digit Modification** entry created earlier.

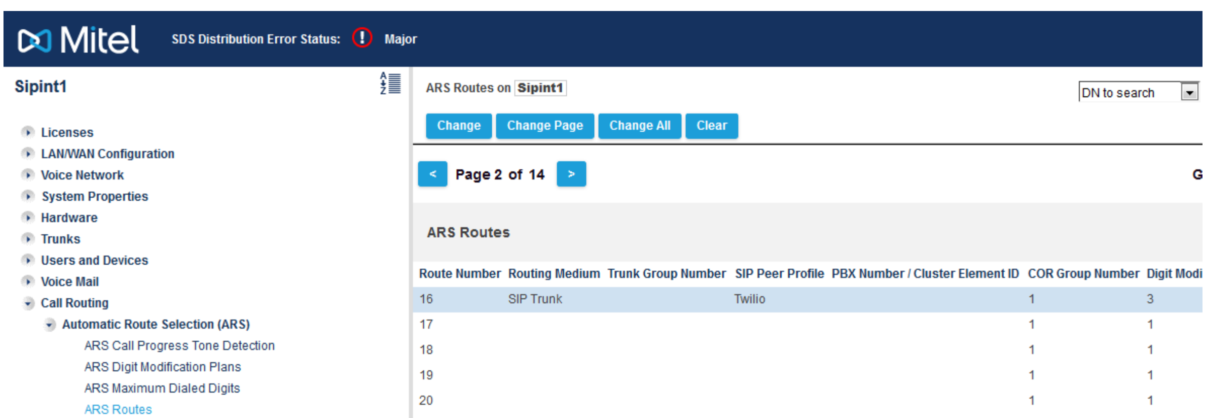

Figure 15 – ARS Routes

ARS Digits Dialed

ARS initiates the routing of trunk calls when certain digits are dialed from a station. In this example, when a user dials **9101613** followed by 7 digits, the call will be routed to Twilio via Route **16**.

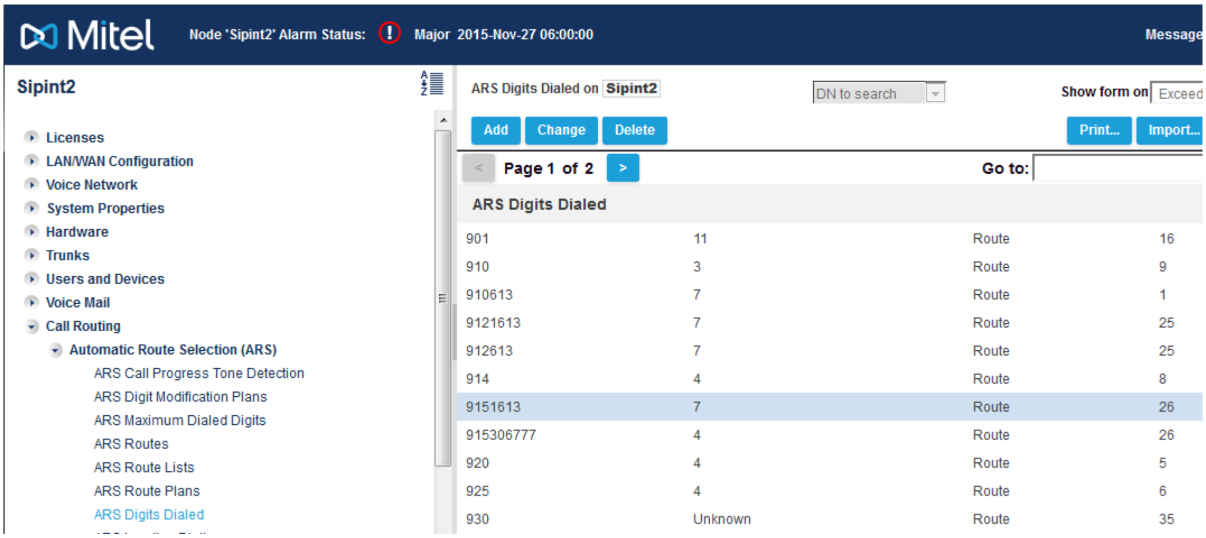

Figure 16 – ARS Digit Dialed

MiVoice Border Gateway Configuration Notes

MBG SIP Options

To enable SIP on the MiVoice Border Gateway (MBG),

- Login to Server Manager of MBG
- Select **Mitel Border Gateway** under Applications
- Select **System Configuration** tab
- Click on **Settings**
- Scroll down to the **SIP Options** section, see **Figure 17**
- Ensure the necessary transport protocols are selected, Twilio uses UDP

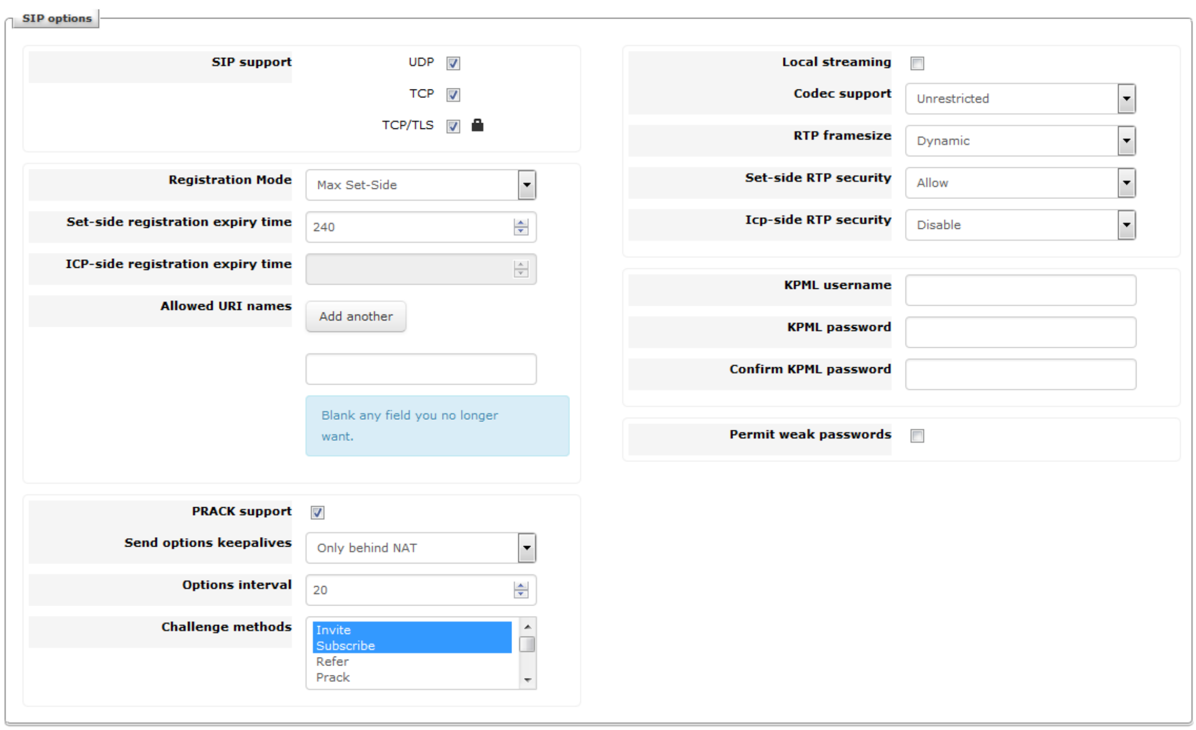

Figure 17 – MBG - SIP Settings

Adding MiVoice Business to MBG

To configure MiVoice Business into Mitel Border Gateway (MBG),

- Login to Server Manager of MBG
- Select **Mitel Border Gateway** under Applications
- Select **Service Configuration** tab
- Click on **ICPs**
- Add ICP by clicking the **'+'** symbol under 'ICP Information'
- Enter a name for MiVoice Business, example: SIPINT1
- Enter the IP address of MiVoice Business
- Select the type as MiVoice Business

Figure 18 – Configuration - ICP Setup

SIP Trunk Configuration

To configure Twilio SIP trunking into the Mitel Border Gateway (MBG),

- Under the **Service Configuration** tab of MBG, click on **SIP Trunking**
- Add a SIP Trunk by clicking on the **'+'** under 'SIP Trunk Information' and Enter the SIP trunk's details as shown.

  Name: Enter the trunk name, example: Twilio

  Remote trunk endpoint address: Enter the public IP address or FQDN of the provider's switch or gateway. This address will be provided to you by Twilio.

  Local/Remote RTP framesize (ms): Leave as the default 'Auto'.

  PRACK: Twilio does not currently support PRACK so set this to 'Disabled'.

  Routing rule one: Allows routing of any digits to the selected MiVB

  The rest of the settings are optional and could be configured if required
- Click **Save**

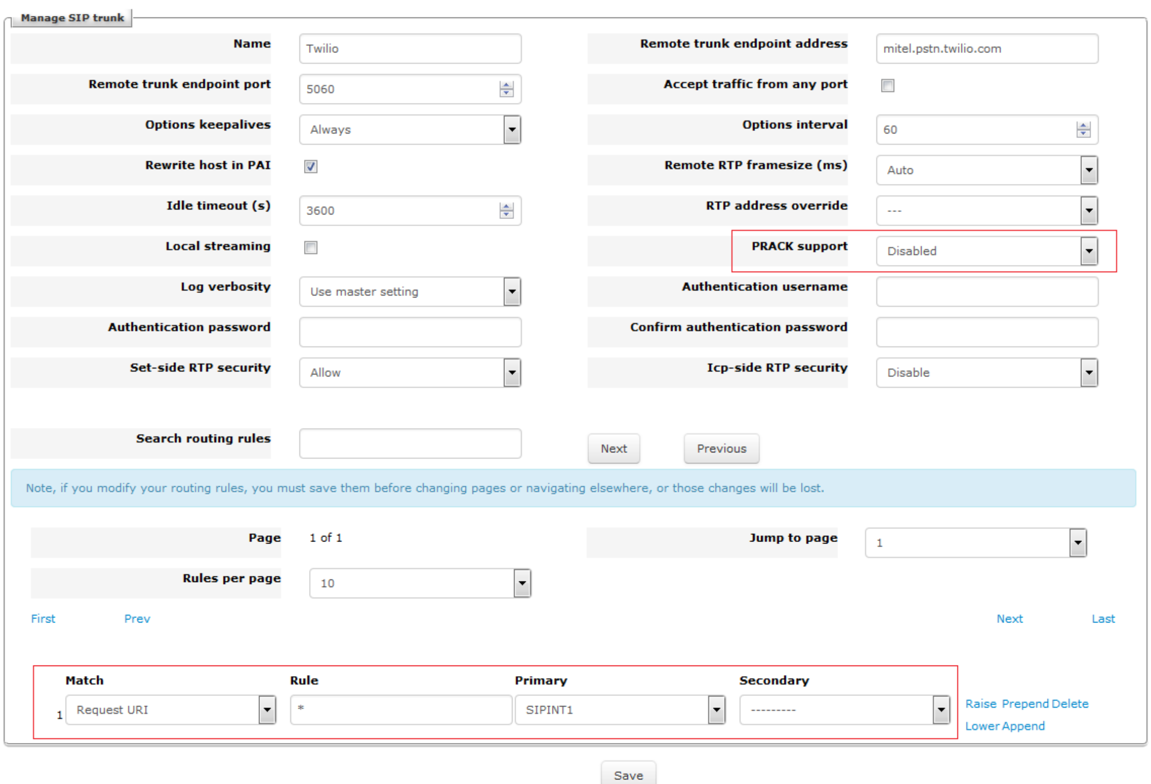

Figure 19 – Services - SIP Trunking setup

Powering connections

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