

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



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## Mitel Technical Configuration Notes:

Configure MiVoice Business 7.2 for use with Twilio SIP Trunking  
April 2016 – 16-4940-00441

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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.

	<p>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.</p>
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## 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 Messaging	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     - 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.

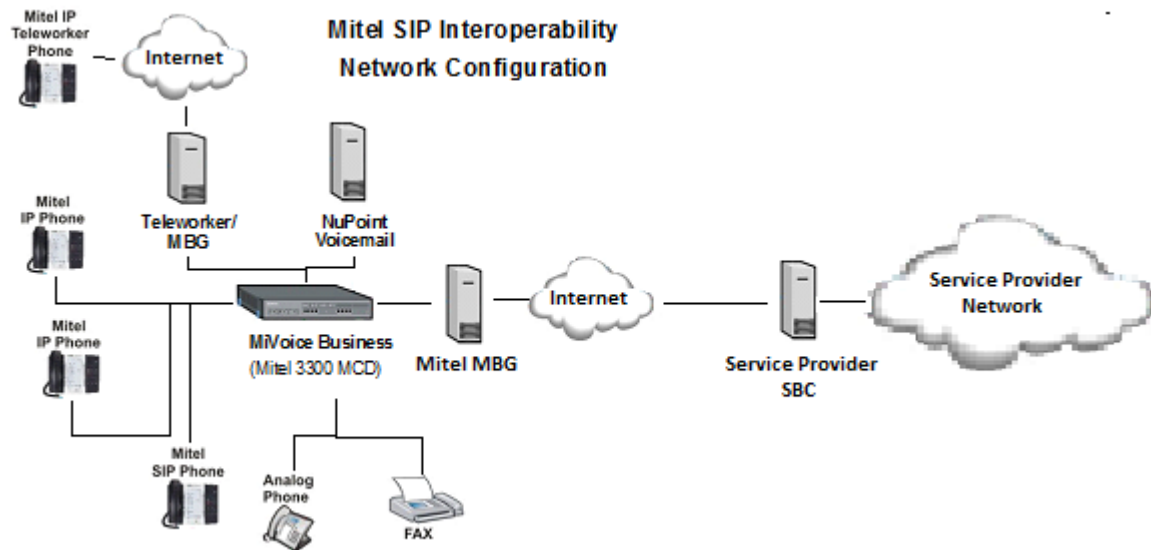


Figure 1 – Network Topology

## 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.

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**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.**

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## 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 [Mitel eDocs Website](http://edocs.mitel.com) (<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.

**Mitel** SDS Distribution Error Status: ! Major

**Sipint1** License and Option Selection on **Sipint1** DN to search

[Change](#)

### License and Option Selection

Online Licensing with the Application Management Center

Application Record ID: 25181182

System Type License Sharing: Enterprise No Hardware Identifier: 000000347977

Licensed Options	Locally Consumed	Locally Allocated	Available for Allocation	Purchased	Local Limits	
					Licenses Allowed	Can be Over Allocated
<b>Users</b>						
IP Users	157	2016	0	2016	Unrestricted	No
External Hot Desk Users	2	30	0	30	Unrestricted	No
ACD Active Agents	0	26	0	26	Unrestricted	No
HTML Applications	0	0	500	500	Unrestricted	No
Analog Lines	0	10	0	10	Unrestricted	No
MiVoice Business Console Active Operators	0	0	20	0	Unrestricted	No
Multi-device Users	0	0	0	0	Unrestricted	No
Multi-device Suites	0	0	0	0	0	No
<b>Messaging</b>						
Embedded Voice Mail	40	100	0	100	Unrestricted	No
Embedded Voice Mail PMS	1	Yes	0	1	Unrestricted	No
<b>Trunking/Networking</b>						
Digital Links	4	4	12	16	Unrestricted	No
Compression		160	0	160	Unrestricted	No
FAX Over IP (T.38)		16	0	16	Unrestricted	No
SIP Trunks	0	1000	10	1010	Unrestricted	No

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**

**Sipint1** Group: lab Alarm Status: Critical Message Board | About | Help | Log Out

Class of Service Options on **Sipint1** DN to search Show form on Sipint1 (Login Node) Go

Change Copy Print... Import... Export... Data Refresh

Page 1 of 11 Go to: value: Go

### Class of Service Options

Class Of Service Number	Comment
1	General

General Advanced

Answer Plus System Reroute Timer	0
Recorded Announcement Device	No
Recorded Announcement Device - Advanced	No
<b>Ringing</b>	
Delay Ring Timer	10
No Answer Recall Timer	17
Ringing Line Select	No
Ringing Timer	30
<b>SMDR</b>	
SMDR External	No
SMDR Internal	No
<b>Trunk</b>	
ANI/DNIS/ISDN Number Delivery Trunk	No
DASS II OLI/TLI Provided	No
Public Network Access via DPNSS	Yes
Public Network To Public Network Connection Allowed	Yes
Public Trunk	Yes
R2 Call Progress Tone	No
Suppress Simulated CCM after ISDN Progress	No
Trunk Calling Party Identification	Yes
Trunk Flash Allowed	No
Two B-Channel Transfer Allowed	No
<b>Voice Mail</b>	
COV/ONS/E&M Voice Mail Port	No
ONS VMail-Delay Dial Tone Timer	5

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.

The screenshot shows the Mitel SIPint1 interface. On the left, the 'Voice Network' menu is expanded, and 'Network Elements' is selected. The main panel displays 'Network Elements on Sipint1'. A table lists the network elements, with one entry for 'Twilio' of type 'Other' and FQDN 'mitel.pstn.twilio.com'. Below the table, the configuration details for the 'Twilio' element are shown:

Name	Twilio
Type	Other
FQDN or IP Address	mitel.pstn.twilio.com
Data Sharing	NO
Local	False
Version	
Zone	1
ARID	
SIP Peer Specific	
SIP Peer Transport	default
SIP Peer Port	0
External SIP Proxy FQDN or IP Address	
External SIP Proxy Transport	default
External SIP Proxy Port	0
SIP Registrar FQDN or IP Address	
SIP Registrar Transport	default
SIP Registrar Port	0
SIP Peer Status	Auto-Detect/Normal

**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.

The screenshot shows the Mitel SIPint1 interface. On the left, the 'Voice Network' menu is expanded, and 'Network Elements' is selected. The main panel displays 'Network Elements on Sipint1'. A table lists the network elements, with one entry for 'MBGTrunk' of type 'Outbound Proxy' and IP address '192.168.101.205'. Below the table, the configuration details for the 'MBGTrunk' element are shown:

Name	MBGTrunk
Type	Outbound Proxy
FQDN or IP Address	192.168.101.205
Data Sharing	NO
Local	False
Version	
Zone	1
ARID	
Outbound Proxy Specific	
Outbound Proxy Transport Type	UDP
Outbound Proxy Port	5060

**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.

**Mitel** Group 'tab' Alarm Status: ✖ Critical

**Sipint1**

Trunk Attributes on **Sipint1** DN to search

[Change](#) [Change Page](#) [Change All](#) [Clear](#)

**Page 2 of 15**

**Trunk Attributes**

12	No	Off	On	7	1
43	No	Off	Off	4	4
Trunk Service Number 12					
Release Link Trunk No					
Call Recognition Service Off					
Direct Inward Dialing Service On					
Class of Service 7					
Class of Restriction 1					
Baud Rate 300					
Intercept Number 1					
Non-dial In Trunks Answer Point - Day					
Non-dial In Trunks Answer Point - Night 1					
Non-dial In Trunks Answer Point - Night 2					
Dial In Trunks Incoming Digit Modification - Absorb 4					
Dial In Trunks Incoming Digit Modification - Insert					
Dial In Trunks Answer Point					
Dial In Trunks Insert Forwarding Information No					
Trunk Label Twilio					

**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.

The screenshot displays the Mitel SIPint1 configuration interface. The left sidebar shows a navigation tree with categories like Licenses, LAN/WAN Configuration, Voice Network, System Properties, Hardware, Trunks, SIP, Users and Devices, Voice Mail, Call Routing, Music On Hold, Emergency Services Management, Property Management, and Maintenance and Diagnostics. The main area is titled 'SIP Peer Profile on Sipint1' and includes tabs for Basic, Call Routing, Calling Line ID, SDP Options, Signaling and Header Manipulation, Timers, Key Press Event, and Outgoing. The 'Basic' tab is active, showing fields for SIP Peer Profile Label (Twilio), Network Element (Twilio), Local Account Information (Registration User Name, Address Type: IP Address: 192.168.101.10), Administration Options (Interconnect Restriction: 1, Maximum Simultaneous Calls: 4, Minimum Reserved Call Licenses: 0), and another set of Administration Options (Outbound Proxy Server: MBGTrunk, SMDR Tag: 0, Trunk Service: 12, Zone: 1, User Name, Password, Confirm Password, Authentication Option for Incoming Calls: No Authentication, Subscription User Name, Subscription Password, Subscription Confirm Password).

Figure 7 – SIP Peer Profile - Basic

The screenshot shows the Mitel SIP Peer Profile configuration page for 'Sipint1'. The left sidebar contains a navigation tree with categories: Licenses, LAN/WAN Configuration, Voice Network, System Properties, Hardware, and Trunks. Under Trunks, there are sub-items: Trunk Attributes, DTS Service Profiles, Analog, Digital, IP/XNET, and SIP. The SIP item is expanded, showing 'DID Ranges for CPN Substitution' and a link to 'SIP Peer Profile'. The main content area is titled 'SIP Peer Profile on Sipint1' and includes 'Add', 'Change', and 'Delete' buttons. Below this is a table with columns: Twilio, Twilio, MBGTrunk, No, 12, 90, 1. The 'Call Routing' tab is selected, showing a list of settings: Alternate Destination Domain Enabled (No), Alternate Destination Domain FQDN or IP Address (No), Enable Special Re-invite Collision Handling (No), Only Allow Outgoing Calls (No), Private SIP Trunk (No), Reject Incoming Anonymous Calls (No), Route Call Using P-Called-Party-ID (if present) (No), and Route Call Using To Header (No).

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.

The screenshot shows the Mitel SIP Peer Profile configuration page for 'Sipint1', specifically the 'Calling Line ID' tab. The left sidebar is identical to Figure 7. The main content area is titled 'SIP Peer Profile on Sipint1' and includes 'Add', 'Change', and 'Delete' buttons. Below this is a table with columns: Twilio, Twilio, MBGTrunk, No, 12, 90, 1. The 'Calling Line ID' tab is selected, showing a list of settings: Default CPN (6135192701), Default CPN Name (No), CPN Restriction (No), Public Calling Party Number Passthrough (No), Strip PNI (No), Use Diverting Party Number as Calling Party Number (No), and Use Original Calling Party Number If Available (No).

Figure 8 – SIP Peer Profile - Calling Line ID

**Mitel** SDS Distribution Error Status: ! Major

**Sipint1**

- Licenses
- LAN/WAN Configuration
- Voice Network
- System Properties
- Hardware
- Trunks
  - Trunk Attributes
  - DTS Service Profiles
  - Analog
  - Digital
  - IP/XNET
  - SIP
    - DID Ranges for CPN Substitution
    - [SIP Peer Profile](#)
    - SIP Peer Profile Assignment by Incoming DID
    - SIP Peer Profile Called Party Inward Dialing Modification
    - SIP Peer Profile Calling Party Inward Dialing Modification
    - URI/Number Translation
- Users and Devices

SIP Peer Profile on **Sipint1** DN to search

[Add](#) [Change](#) [Delete](#)

**SIP Peer Profile**

Twilio	Twilio	MBGTrunk	No	12	0	1	
Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	
Allow Peer To Use Multiple Active M-Lines	Yes	Allow Using UPDATE For Early Media Renegotiation	No	Avoid Signaling Hold to the Peer	Yes	AVP Only Peer	No
Enable Mitel Proprietary SDP	No	Force sending SDP in initial Invite message	Yes	Force sending SDP in initial Invite - Early Answer	No	Ignore SDP Answers in Provisional Responses	Yes
Limit to one Offer/Answer per INVITE	Yes	NAT Keepalive	Yes	Prevent the Use of IP Address 0.0.0.0 in SDP Messages	Yes	Renegotiate SDP To Enforce Symmetric Codec	Yes
Repeat SDP Answer If Duplicate Offer Is Received	No	Restrict Audio Codec	No Restriction	RTP Packetization Rate Override	No	RTP Packetization Rate	20ms
Special handling of Offers in 2XX responses (INVITE)	No	Suppress Use of SDP Inactive Media Streams	No				

Figure 9 – SIP Peer Profile - SDP Options

**Mitel** Group 'lab' Alarm Status: X Critical

**Sipint1**

- Licenses
- LAN/WAN Configuration
- Voice Network
- System Properties
- Hardware
- Trunks
  - Trunk Attributes
  - DTS Service Profiles
  - Analog
  - Digital
  - IP/XNET
  - SIP
    - DID Ranges for CPN Substitution
    - [SIP Peer Profile](#)
    - SIP Peer Profile Assignment by Incoming DID
    - SIP Peer Profile Called Party Inward Dialing Modification
    - SIP Peer Profile Calling Party Inward Dialing Modification
    - URI/Number Translation
- Users and Devices
- Voice Mail
- Call Routing
- Music On Hold
- Emergency Services Management
- Property Management
- Maintenance and Diagnostics


SIP Peer Profile on **Sipint1** DN to search

[Add](#) [Change](#) [Delete](#)

**SIP Peer Profile**

Twilio	Twilio	MBGTrunk	No	12	90	1	
Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	
Trunk Group Label	No	Allow Display Update	No	Build Contact Using Request URI Address	No	De-register Using Contact Address not *	Yes
Disable Reliable Provisional Responses	Yes	Disable Use of User-Agent and Server Headers	No	Domain for Trunk Context	No	E.164: Enable sending '+'	Yes
E.164: Add '+' if digit length > N digits	0	E.164: Do not add '+' to Emergency Called Party	No	E.164: Do not add '+' to Called Party	No	Force Max-Forward: 70 on Outgoing Calls	No
If TLS use 'sips:' Scheme	No	Ignore Incoming Loose Routing Indication	No	Multilingual Name Display	No	Only use SDP to decide 180 or 183	Yes
Override Diversion Header with External Calling Number	No	Prefer From Header for Caller ID	No	Require Reliable Provisional Responses on Outgoing Calls	No	Signal Privacy (if enabled) on Emergency Calls	No
Suppress Redirection Headers	No	Use Fixed Retry Time for 491	No	Use Privacy: none	No	Use P-Asserted Identity Header	Yes
Use P-Asserted Identity for Billing	No	Use P-Call-Leg-ID Header	No	Use P-Preferred Identity Header	No	Use Restricted Character Set For Authentication	No
Use To Address in From Header on Outgoing Calls	No	Use user=phone	No	Use user=phone for Diversion Header	No		

Figure 10 – SIP Peer Profile - Signaling and Header Manipulation


SDS Distribution Error Status: ! Major

Sipint1

Licenses

LAN/WAN Configuration

Voice Network

System Properties

Hardware

Trunks

Trunk Attributes

DTS Service Profiles

Analog

Digital

IP/XNET

SIP

DID Ranges for CPN Substitution

SIP Peer Profile

SIP Peer Profile on **Sipint1**

Add

Change

Delete

DN to search

SIP Peer Profile

Twilio	Twilio	MBGTrunk	No	12	90	1
Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event
Outgoing						

Keep-Alive (OPTIONS) Period

120

Registration Period

3600

Registration Period Refresh (%)

50

Registration Maximum Timeout

90

Session Timer

90

Session Timer: Local as Refresher

No

Subscription Period

3600

Subscription Period Minimum

300

Subscription Period Refresh (%)

80

Invite Ringing Response Timer

0

### 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, 119999999999999999999999.) Note that the string "11" by itself would not count as a match, as the '\*' represents 1 or more digits.


SDS Distribution Error Status: ! Major

Sipint1

- Licenses
- LAN/WAN Configuration
- Voice Network
- System Properties
- Hardware
- Trunks
  - Trunk Attributes
  - DTS Service Profiles
    - Analog
    - Digital
    - IP/XNET
    - SIP
      - DID Ranges for CPN Substitution
      - SIP Peer Profile
        - SIP Peer Profile Assignment by Incoming DID

SIP Peer Profile Assignment by Incoming DID on **Sipint1**

Add

Change

Delete

SIP Peer Profile Assignment by Incoming DID

16135192701	Twilio	Twilio
Incoming DID Range	16135192701	
SIP Peer Profile Label	Twilio	
Comment	Twilio	

**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.

ARS Digit Modification Plans on **Sipint1** DN to search

[Change](#) [Change Page](#) [Change All](#) [Clear](#)

< Page 1 of 40 >

Digit Modification Number	Number of Digits to Absorb	Digits to be Inserted
1	0	
2	2	
3	3	
4	3	11129
5	0	

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.

ARS Routes on **Sipint1** DN to search

[Change](#) [Change Page](#) [Change All](#) [Clear](#)

< Page 2 of 14 >

Route Number	Routing Medium	Trunk Group Number	SIP Peer Profile	PBX Number / Cluster Element ID	COR Group Number	Digit Modi
16	SIP Trunk		Twilio		1	3
17					1	1
18					1	1
19					1	1
20					1	1

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**.

The screenshot shows the Mitel SIPint2 web interface. The top header includes the Mitel logo, a status message 'Node "Sipint2" Alarm Status: Major 2015-Nov-27 06:00:00', and a 'Message' link. The left sidebar contains a navigation menu with the following items: Licenses, LAN/WAN Configuration, Voice Network, System Properties, Hardware, Trunks, Users and Devices, Voice Mail, and Call Routing. The 'Call Routing' item is expanded, showing sub-items: Automatic Route Selection (ARS), ARS Call Progress Tone Detection, ARS Digit Modification Plans, ARS Maximum Dialed Digits, ARS Routes, ARS Route Lists, ARS Route Plans, and ARS Digits Dialed. The 'ARS Digits Dialed' sub-item is selected, displaying a table of configurations.

ARS Digits Dialed on **Sipint2**

Page 1 of 2

ARS Digits Dialed	Count	Route	Count
901	11	Route	16
910	3	Route	9
910613	7	Route	1
9121613	7	Route	25
912613	7	Route	25
914	4	Route	8
9151613	7	Route	26
915306777	4	Route	26
920	4	Route	5
925	4	Route	6
930	Unknown	Route	35

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

The screenshot shows the 'SIP options' configuration page. It is divided into several sections:

- SIP support:** Includes checkboxes for UDP (checked), TCP (checked), and TCP/TLS (checked with a lock icon).
- Registration Mode:** A dropdown menu set to 'Max Set-Side'.
- Set-side registration expiry time:** A numeric input field set to 240.
- ICP-side registration expiry time:** A numeric input field.
- Allowed URI names:** A section with an 'Add another' button and a text input field. Below it is a blue note: 'Blank any field you no longer want.'
- PRACK support:** A checkbox that is checked.
- Send options keepalives:** A dropdown menu set to 'Only behind NAT'.
- Options interval:** A numeric input field set to 20.
- Challenge methods:** A list box with 'Invite', 'Subscribe', 'Refer', and 'Prack'. 'Invite' is currently selected.
- Local streaming:** A checkbox that is unchecked.
- Codec support:** A dropdown menu set to 'Unrestricted'.
- RTP framesize:** A dropdown menu set to 'Dynamic'.
- Set-side RTP security:** A dropdown menu set to 'Allow'.
- Icp-side RTP security:** A dropdown menu set to 'Disable'.
- KPML username:** A text input field.
- KPML password:** A text input field.
- Confirm KPML password:** A text input field.
- Permit weak passwords:** A checkbox that is unchecked.

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

Manage ICP	
Name	SIPINT1
Type	MiVoice Business
SIP capabilities	UDP
Hostname or IP address	192.168.101.10
Installer password	
Indirect call recording capable	<input type="checkbox"/>

Save

**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**

**Manage SIP trunk**

<b>Name</b>	Twilio	<b>Remote trunk endpoint address</b>	mitel.pstn.twilio.com
<b>Remote trunk endpoint port</b>	5060	<b>Accept traffic from any port</b>	<input type="checkbox"/>
<b>Options keepalives</b>	Always	<b>Options interval</b>	60
<b>Rewrite host in PAI</b>	<input checked="" type="checkbox"/>	<b>Remote RTP framesize (ms)</b>	Auto
<b>Idle timeout (s)</b>	3600	<b>RTP address override</b>	---
<b>Local streaming</b>	<input type="checkbox"/>	<b>PRACK support</b>	Disabled
<b>Log verbosity</b>	Use master setting	<b>Authentication username</b>	
<b>Authentication password</b>		<b>Confirm authentication password</b>	
<b>Set-side RTP security</b>	Allow	<b>Icp-side RTP security</b>	Disable

**Search routing rules**  Next Previous

Note, if you modify your routing rules, you must save them before changing pages or navigating elsewhere, or those changes will be lost.

**Page** 1 of 1 **Jump to page** 1

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Match	Rule	Primary	Secondary
1 Request URI	*	SIPINT1	-----

[Raise](#) [Prepend](#) [Delete](#) [Lower](#) [Append](#)

Save

Figure 19 – Services - SIP Trunking setup

