

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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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	<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.

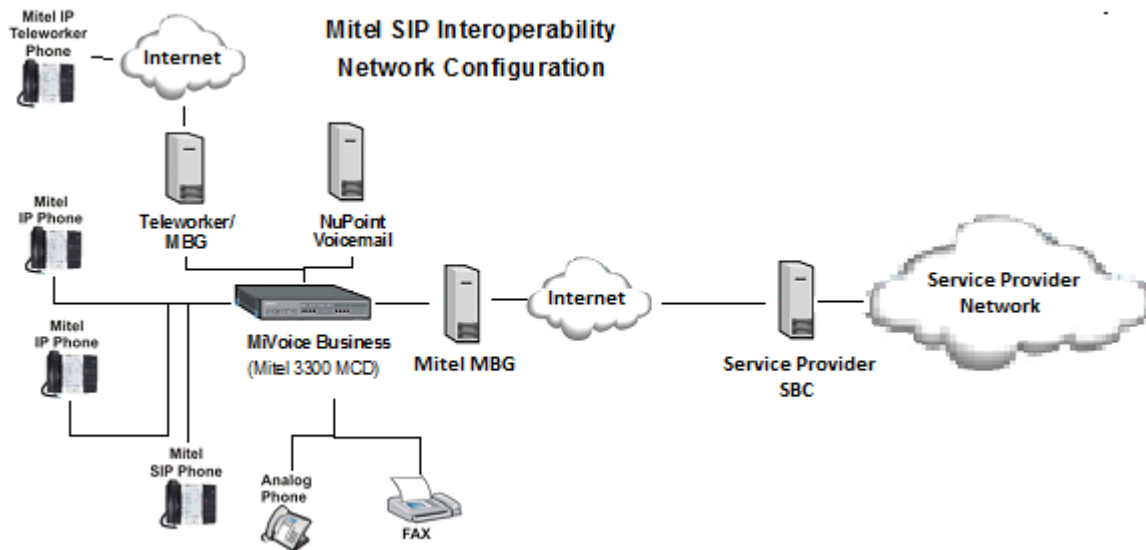


Figure 1 – Network Topology

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.

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 [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
Users						
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
Messaging						
Embedded Voice Mail	40	100	0	100	Unrestricted	No
Embedded Voice Mail PMS	1	Yes	0	1	Unrestricted	No
Trunking/Networking						
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**

The screenshot displays the Mitel SIPint1 web interface. The left sidebar shows the navigation menu with 'System Properties' > 'System Settings' > 'System Feature Settings' > 'Class of Service Options' highlighted. The main content area shows the 'Class of Service Options' configuration page for 'Sipint1'. The page includes a search bar, navigation buttons (Change, Copy, Print, Import, Export, Data Refresh), and a table of options. The table has columns for 'Class Of Service Number' and 'Comment'. The 'Class Of Service Number' is '1' and the 'Comment' is 'General'. The table is divided into sections: General, Advanced, Ringing, SMDR, Trunk, and Voice Mail. The 'Trunk' section is highlighted with a red box and contains the following options:

Option	Value
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

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 web interface. The left sidebar is expanded to 'Voice Network' > 'Network Elements'. The main panel shows 'Network Elements on Sipint1' with a table listing one element: 'Twilio' (Type: Other, FQDN or IP Address: 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 web interface. The left sidebar is expanded to 'Voice Network' > 'Network Elements'. The main panel shows 'Network Elements on Sipint1' with a table listing one element: 'MBGTrunk' (Type: Outbound Proxy, FQDN or 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.

Group 'lab' Alarm Status: ✖ Critical

Sipint1

Trunk Attributes on Sipint1

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 shows the Mitel SIPint1 configuration interface. The top navigation bar includes the Mitel logo, a status indicator for 'SDS Distribution Error Status: Major', and the user 'Sipint1'. The left sidebar shows a tree view of configuration categories, with 'SIP' expanded to show 'SIP Peer Profile'. The main content area displays the configuration for the 'SIP Peer Profile' on the 'Basic' tab. The configuration is organized into several sections:

- Local Account Information:**
 - Registration User Name: (blank)
 - Address Type: IP Address: 192.168.101.10
- Administration Options:**
 - Interconnect Restriction: 1
 - Maximum Simultaneous Calls: 4
 - Minimum Reserved Call Licenses: 0
- Administration Options (continued):**
 - Outbound Proxy Server: MBGTrunk
 - SMDR Tag: 0
 - Trunk Service: 12
 - Zone: 1
 - User Name: (blank)
 - Password: (masked with asterisks)
 - Confirm Password: (masked with asterisks)
 - Authentication Option for Incoming Calls: No Authentication
 - Subscription User Name: (blank)
 - Subscription Password: (masked with asterisks)
 - Subscription Confirm Password: (masked with asterisks)

Figure 7 – SIP Peer Profile - Basic

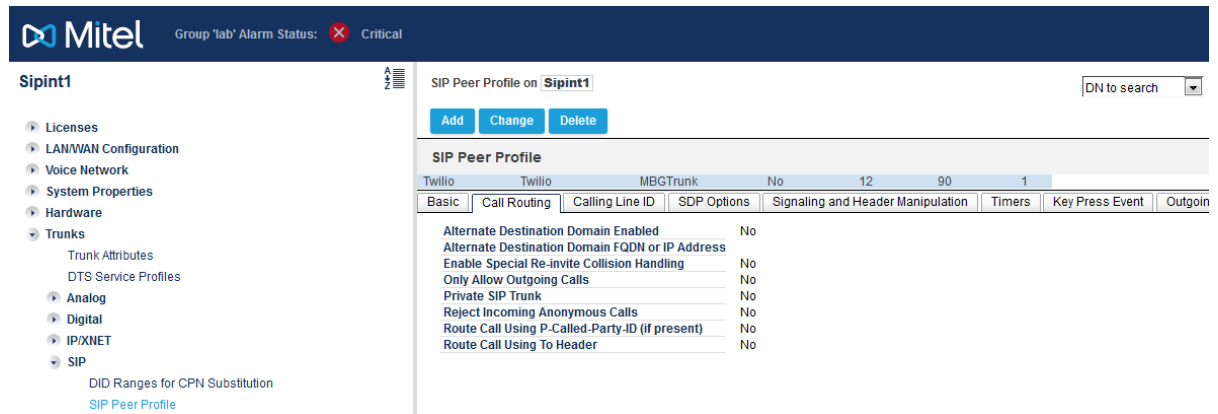


Figure 7 – SIP Peer Profile - Call Routing

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.

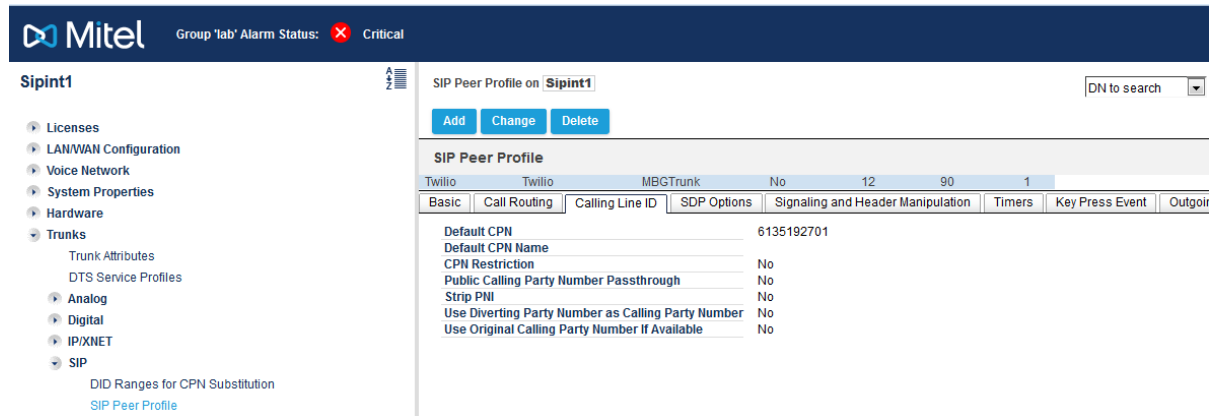


Figure 8 – SIP Peer Profile - Calling Line ID

SIP Peer Profile on **Sipint1**

Buttons: Add, Change, Delete

SIP Peer Profile	
Twilio	Twilio
Basic	Call Routing
Calling Line ID	SDP Options
Signaling and Header Manipulation	Timers
Key Press Event	Outgoing

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

SIP Peer Profile on **Sipint1**

Buttons: Add, Change, Delete

SIP Peer Profile	
Twilio	Twilio
Basic	Call Routing
Calling Line ID	SDP Options
Signaling and Header Manipulation	Timers
Key Press Event	Outgoing

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

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.

The screenshot shows the Mitel SIPint1 interface with a navigation menu on the left and a main content area. The main content area displays 'ARS Digit Modification Plans on Sipint1'. Below this, there are buttons for 'Change', 'Change Page', 'Change All', and 'Clear'. A pagination control shows 'Page 1 of 40'. The main table is titled 'ARS Digit Modification Plans' and contains the following data:

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.

The screenshot shows the Mitel SIPint1 interface with a navigation menu on the left and a main content area. The main content area displays 'ARS Routes on Sipint1'. Below this, there are buttons for 'Change', 'Change Page', 'Change All', and 'Clear'. A pagination control shows 'Page 2 of 14'. The main table is titled 'ARS Routes' and contains the following data:

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 configuration interface. The left sidebar lists various system settings, with 'Call Routing' expanded to show 'Automatic Route Selection (ARS)'. The main content area is titled 'ARS Digits Dialed on Sipint2' and contains a table of configurations. The table has columns for the dialed digits, the number of occurrences, the route name, and the route ID. The row for '9151613' is highlighted in blue, indicating it is the current selection.

ARS Digits Dialed	Count	Route	Route ID
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

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	Hostname or IP address	192.168.101.10
Type	MiVoice Business	Installer password	
SIP capabilities	UDP	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**

The screenshot displays the 'Manage SIP trunk' configuration interface. It is divided into two columns of settings. The left column includes fields for Name (Twilio), Remote trunk endpoint port (5060), Options keepalives (Always), Rewrite host in PAI (checked), Idle timeout (3600s), Local streaming (unchecked), Log verbosity (Use master setting), Authentication password, Set-side RTP security (Allow), and Search routing rules. The right column includes Remote trunk endpoint address (mitel.pstn.twilio.com), Accept traffic from any port (unchecked), Options interval (60), Remote RTP framesize (Auto), RTP address override (---), PRACK support (Disabled), Authentication username, Confirm authentication password, and Icp-side RTP security (Disable). Below the settings is a routing rule table with one rule: Match 'Request URI', Rule '*', Primary 'SIPINT1', and Secondary '-----'. A 'Save' button is located at the bottom center.

Figure 19 – Services - SIP Trunking setup

