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MiVB - Configure MiVoice Business 10.2 for use with phonestar* SIP Trunking Without MBG

Description: This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVB to connect to phonestar* SIP Trunking without MBG.

Environment: MiVoice Business 10.2 (10.2.0.54), Mitel 69xx/69xxw MiNET 03.00.00.050 and Mitel 69xx/68xx SIP 6.4.0.3022

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Mitel Technical Configuration Notes – Configure MiVoice Business 10.2 for use with phonestar* SIP Trunking without MBG

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Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVB to connect to phonestar* SIP Trunking without MBG. The different devices 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	February, 2025	Initial interop certification of MiVoice Business 10.2
		with phonestar* SIP Trunking without MBG

Interop Status

The Interop of phonestar* SIP Trunking has been given a Certification status. The phonestar* will be included in the Mitel Interoperability Reference Guide (IRG). The status of phonestar* SIP Trunking achieved is:



Software & Hardware Setup

This was the test setup to generate a basic SIP call between phonestar* SIP Trunking and the MiVB without MBG.

Note – Although this testing was performed on the below tested variants, the scope of this testing can be extended to other product variants that work with the same firmware. The list of components for which this testing can be considered applicable is given in the "Additional Applicable Variants" column of the following table –

Manufacturer	Tested Variants	Software Version	Additional Applicable Variants
Mitel	MiVoice Business	10.2 (10.2.0.54)	NA
Mitel	69xx/69xxw MiNET	03.00.00.050	NA
Mitel	69xx/68xx SIP	6.4.0.3022	NA

Tested Features

This is an overview of the features tested during the Interop test cycle and not a detailed view of the test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call through Service Provider phonestar* and their PSTN gateway, call holding, transferring, conferencing, busy calls, long calls durations	1
Automatic Call	Making calls to an ACD environment with RAD treatments,	
Distribution	Interflow and Overflow call scenarios and DTMF detection	-
DTMF Signal	Sending DTMF after call setup (i.e. mailbox password)	 ✓
Personal Ring	Receiving calls through Service Provider phonestar* and their	
Groups	PSTN gateway to a personal ring group. Also moving calls to/from	
	the prime member and group members	
Packetization	Forcing the Mitel MiVB to stream RTP packets at different intervals, from 10ms to 30ms	✓
External Hot Desking	Receiving calls through Service Provider phonestar* and their PSTN gateway to PRG with EHDU. Including moving calls to/from the prime member of the PRG with the EHDU	1
Codec	Different codecs support (G711 and G722)	
TLS/SRTP	Making and receiving calls through Service Provider phonestar* via TLS/SRTP	1
Fax	G711 & T.38 Fax Calls	
Video	Making and receiving a call through Service Provider phonestar* with video capable devices	X
Resiliency	Fail-over and Fail-back scenarios between MiVB, MBG and the service provider phonestar* SBC setup	NA

✓ - No issues found × - Issues found, cannot recommend using △ - Issues found NA – Not Applicable

Device Limitations and Known Issues

This is a list of problems or unsupported features when phonestar* SIP Trunk is connected to the MiVB.

Feature	Problem Description
Video Calls	phonestar* does not support video calling.
	Recommendation : Please contact phonestar* support for updates for supporting of the video calling.
Codec (G729)	phonestar* does not support G729 codec calling.
	Recommendation : Please contact phonestar* support for more information on this.
Resiliency	phonestar* does not support DNS SRV. For redundancy, phonestar* uses a virtual IP concept which failovers to another host.
	Recommendation : Please contact phonestar* support for getting more information on this.
Long Duration Calls	The long-duration calls are failing with BYE from phonestar* side after 30mins when enabling session timers on the MiVB and recommending that to disable session timers on the MiVB in order to avoid issues.
	Recommendation : Please disable session timer (set value to "0") on the MiVB or contact phonestar* support for more information on this.
Supervised Transfer	In case of incoming PSTN call to Mitel Phone (DID) and Mitel Phone doing the supervised transfer the call to another PSTN user scenario is failing and observed two-way audio issues after the call is transferred over G722/Opus codecs. Hence, please restrict the codec to G711 on the MiVB SIP Peer Profile in order to work successfully.
	Recommendation : Please contact phonestar* support for getting more information on this.
Blind Transfer	In case of incoming PSTN call to Mitel Phone (DID) and Mitel Phone doing the blind transfer the call to another PSTN user scenario is failing and observed two-way audio issues after the call is transferred over G722/Opus codecs. Hence, please restrict the codec to G711 on the MiVB SIP Peer Profile in order to work successfully.
	Recommendation : Please contact phonestar* support for getting more information on this.
PRACK	phonestar* does not support PRACK and responding with 420 Bad Extension as Unsupported: 100rel.
	Recommendation : Please contact phonestar* support for getting more information on this.

Network Topology



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 phonestar* SIP Trunking connected with 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.

MiVB Configuration Notes

The following steps show how to program a MiVB to interconnect with phonestar* with or without an asterisk SIP Trunking without MBG.

Configuration Template

A configuration template can be found in the same Mitel Knowledge Management System (KMS) article as this document. The template is a Microsoft Excel spreadsheet (.csv format) **solely** consisting of the SIP Peer profile option settings used during interop testing. All other forms should be programmed as indicated below. Importing the template can save you considerable configuration time and reduce the likelihood of data-entry errors. Refer to the MiVB documentation on how the Import functionality is used.

Network Requirements

- There must be adequate bandwidth to support the voice over IP. 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 MiVB Engineering guidelines 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 MiVB 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 phonestar* with or without an asterisk SIP Trunking. 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 🕴 MiVoice	e Busir	ness				Node Alarm Status: Cte	ar 2024-Dec-13 13:40:50	Ģ	?	≣ %₀	
Primary	ź≣	License and Option Selection on Primary		Search DN 🗸					Show form	on Not Accessibl	,e
Licenses		Change						Print	Import	Export	
License and Option Selection		License and Option Selection									
System Capacity Dimension Selection		Online Licensing with the Application Management C	enter								
Application Group Licensing 🎺		Application Record ID 47105408									
LAN/WAN Configuration Voice Network		System Type	License Sharing		Hardware Identi	fier					
System Properties		Enterprise	No		d5c59edc-f21e-4	lddb-82f0-8dacb78228cc					
Hardware Trunks		Licensed Options		l ocally Consumed	Locally Allocated	Available for Allocation	Purchased	Licenses Allowed	Loc	al Limits	ated
Users and Devices		Users									
Integrated Directory Services Voice Mail		IP Users		21	510	0	510	Unrestricted		Yes	
Call Routing		External Hot Desk Users		1	50	0	50	Unrestricted		Yes	
Music On Hold Emergency Services Management		ACD Active Agents		0	10	0	10	Unrestricted		No	
Property Management		HTML Applications		0	500	0	500	Unrestricted		Yes	
Maintenance and Diagnostics		Single Line Users		0	50	0	50	Unrestricted		Yes	
		MiVoice Business Console Active Operators		0	10	0	10	Unrestricted		No	
		Multi-device Users		0	200	0	200	Unrestricted		Yes	
		Multi-device Suites		0	0	0	0	0		No	
		Messaging									
		Embedded Voice Mail		22	100	0	100	Unrestricted		Yes	
		Embedded Voice Mail PMS		0	No	1 3		Unrestricted		Yes	
		Trunking / Networking									
		Digital Links		0	0	2 1	w 0	Unrestricted		Yes	_
		SIP Trunks		0	100	0	100	Unrestricted		Yes	

Figure 2 – License and Option Selection

Class of Service Assignment

The Class of Service Options Assignment form is used to create or edit a Class of Service and specify its options. Classes of Service, identified by Class of Service numbers, are referenced in the Trunk Service Assignment 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
- Campon Tone Security/FAX Capable set to Yes

Mitel MiVoid	ice Business	Node Alarm Status: Clear 2025 Feb 19 07:08:32
Local_224	Class of Service Options on Local_224 Search DN V	Sho
Licenses	Change Copy	Print.
LAN/WAN Configuration Voice Network System Properties	Vage 1 of 11 2 00 to value	
System Settings System Feature Settings	Class Of Service Number	Comment
System Options Shared System Options 🛷	Central Advanced	r nonestar
Class of Service Options	Recorded Announcement Device - Advanced	No
Class of Restriction Groups 🞺	Allow Recall after Transfer	No
System Access Points 🍻 Feature Access Codes 🞺	No Answer Recall Timer	17
Independent Account Codes 🥔 Default Account Codes 📣	Ringing Line Select Ringing Timer	No 180
System Account Codes 🎺 System Speed Calls 💣	SMDR SMDR External	No
Tenants	SMDR Internal Trunk	No
Traffic Report Options 🎺	ANIDNIS1SDN Number Delivery Trunk DASS II OLIITLI Provided	No No
Inward Dialing Modification	Public Network Access via DPNSS Public Network To Public Network Connection Allowed	Yes
System IP Ports 🧼 Location Based Numbers 🖨	Public Trunk	Yes
System Administration	Suppress Simulated CCM after ISDN Progress	NO
Trunks	Trunk Calling Party Identification Trunk Flash Allowed	Yes No
Users and Devices Integrated Directory Services	Two B-Channel Transfer Allowed Voice Mail	No
Voice Mail Call Routing	COV/ONS/E&M Voice Mail Port ONS VMail-Delay Dial Tone Timer	No 5

Figure 3 – Class of Service

Network Element Assignment

Create a network element for Service Provider phonestar* with or without an asterisk SIP Trunking. In this example, the soft switch is reachable by an IP Address/FQDN and is defined as "phonestar" in the network element assignment form. **The FQDN or IP addresses of the SIP Peer (Network Element), the External SIP Proxy and Registrar are provided by your service provider**.

If your service provider phonestar* trusts your network connection by asking for your gateway external IP address, then programming the IP address for the SIP Peer, Outbound Proxy and Registrar is not required for SIP trunk integration. This will need to be verified with your service provider.

✓ Network Elements	
Name	Phonestar
Туре	Other 🗸
FQDN or IP Address	ps15023ip01.trunk.phor
Local	False
Version	
Zone	1
SIP Peer	
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	ps15023ip01.trunk.phor
SIP Registrar Transport	default 🗸
SIP Registrar Port	0
SIP Peer Status	Auto-Detect/Normal 🗸
	Save Cancel

Figure 4 – Network Element Assignment

Trunk Attributes

This is configured in the Trunk Attributes form. In this example the Trunk Attributes is defined for Trunk Service Number **1** which will be used to direct incoming calls to an answer point in the Mitel MiVB.

🤣 Trunk Attributes	
Trunk Service Number	1
Release Link Trunk	No 🗸
Call Recognition Service	∽ 110
Direct Inward Dialing Service	Off On
Caller Based Routing Service	Off ○ On
Class of Service	1
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	0
Dial In Trunks Incoming Digit Modification - Insert	
Dial In Trunks Answer Point	
Dial In Trunks Insert Forwarding Information	● No ◯ Yes
Trunk Label	Phonestar
	Save Cancel

Figure 5 – Trunk Attributes

SIP Peer Profile

The recommended connectivity via SIP trunking does not require additional physical interfaces. IP/Ethernet connectivity is a part of the base MiVB Platform. The SIP Peer Profile should be configured with the following options:

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

Registration User Name: Provide SIP trunk user name that given by phonestar.

Address Type: Select IP address.

Calling Line ID: The default CPN is applied to all calls unless there is a match in the "Outgoing DID Ranges" of the SIP Peer Profile. Do not use a Default CPN if you want public numbers to be preserved through the SIP interface. Add private numbers into the DID ranges for CPN Substitution form (see DID Ranges for CPN Substitution). Then select the appropriate numbers in the Outgoing DID Ranges in this form (SIP Peer Profile).

Trunk Service Assignment: Enter the trunk service assignment previously configured.

SMDR: If Call Detail Records 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: This entry should be configured to maximum number of SIP trunks provided by Service Provider phonestar*.

NOTE: Ensure the remaining SIP Peer profile policy options are like the screen capture below.

▶ Mitel MiVoice Business		Node Alarm Status: Clear 2025-Feb-19 07:08:32	□? ॿ १₀ ©
Local_224	SIP Peer Profile on Local 224 Search DN v		Show form on Not Accessible
Licenses LANWAN Configuration	Add Change Delete SIP Peer Profile		Print Import Export Data
Voice Network System Properties Hardware	Network Element SIP Peer Profile Label Outbound Proxy Server Phonestar Phonestar	CPN Restriction Trunk Service No 1	Session Timer Zone 0 1
Trunks Trunk Attributes &	Resc. Call Routing Calling Line ID SDP Options Signaling and Header Manipulation Timers Key Press Event Outgoing DID Ranges SIP Peer Profile Label	Profile Information Phonestar	
IP/XNE I SIP DID Ranges for CPN Substitution	Network Element Local Account Information	Phonestar	
SIP Peer Profile SIP Peer Profile Assignment by Incoming DID	Registration User Name Address Type Address Type	1502301 IP Address: 115.110.136.86	
SIP Peer Profile Called Party Inward Dialing Modification SIP Peer Profile Calling Party Inward Dialing Modification	Instructione Restriction Maximum Simultaneous Calls	1 20	
SIP Peer Profile Called Party Outward Dialing Modification URI/Number Translation	Minimum Reserved Call Licenses Outbound Proxy Server	0	
Users and Devices Integrated Directory Services	SMOR Tag Trunk Service	0	
Call Routing	Zone Authentication Options	1	1
Music On Hold Emergency Services Management Property Management	User Name Pessword Comm Password	1502301	
Maintenance and Diagnostics	Authentication Option for Incoming Calls Subscription User Name	No Authentication	1
	Subscription Password Subscription Confirm Password Gateware Oncions	******	
	Digital Trunk Licenses Maximum Digital/Analog Channels	0	

Figure 6 – SIP Peer Profile Assignment - Basic

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information	
Alte	rnate Destinat	ion Domain Enal	bled					,	No
Alte	mate Destinat	ion Domain FQD	N or IP Address	5					
Ena	ble Special Re	invite Collision	Handling					•	No
Onl	y Allow Outgo	ng Calls						٨	No
Priv	ate SIP Trunk							١	No
Rej	ect Incoming A	nonymous Calls						١	No
Rer	oute Incoming	Calls With 486 F	Responses Whe	n Trunks Are Congested				٨	No
Rer	oute Outgoing	Calls On 500 Re	sponses					Ν	No
Rot	te Call Using	-Called-Party-ID	(if present)					Y	Yes
Rou	ite Call Using	o Header						١	No

Figure 7 – SIP Peer Profile Assignment - Call Routing

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information	
Def	ault CPN								
Def	ault CPN Name	e							
CP	Restriction								1
Ov	erride From He	ader with Defaul	t CPN						1
Pul	lic Calling Par	ty Number Pass	hrough						1
Str	p PNI								1
Use	Diverting Par	ty Number as Ca	lling Party Num	ber					ħ
Use	Original Calli	ng Party Number	If Available						1

Figure 8 – SIP Peer Profile Assignment - Calling Line ID

Basi	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information	
А	low Peer To Use	Multiple Active I	M-Lines						Yes
A	low Using UPDA	TE For Early Me	dia Renegotiatio	on					No
A	oid Signaling H	old to the Peer							Yes
A	/P Only Peer								Yes
E	able Mitel Prop	ietary SDP							No
F	rce sending SD	P in initial Invite	message						Yes
F	rce sending SD	P in initial Invite	- Early Answer						No
Ig	nore SDP Answe	rs in Provisional	l Responses						Yes
IP	Media Default								ipv4
L	mit to one Offer/	Answer per INVI	TE						Yes
N	AT Keepalive								Yes
P	event Codec Se	ection on Answe	er.						No
P	event the Use of	IP Address 0.0.0	0.0 in SDP Mess	ages					Yes
R	ject Call withou	t telephone-even	t payload						No
R	enegotiate SDP 1	o Enforce Symm	netric Codec						No
R	peat SDP Answ	er If Duplicate Of	fer Is Received						No
R	strict Audio Co	lec							No Restriction
R	P Packetization	Rate Override							No
R	P Packetization	Rate							20ms
S	ecial handling o	f Offers in 2XX r	esponses (INVI	TE)					No
S	ppress Use of S	DP Inactive Med	ia Streams						Yes

Figure 9 – SIP Peer Profile Assignment - SDP Options

Basi	c Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information		
Т	runk Group Lab	el								
A	llow Display Up	date							No	
В	uild Contact Us	ing Request URI	Address						No	
D	e-register Using	Contact Addres	s not *						Yes	
D	isable Reliable	Provisional Resp	onses						No	
D	isable Use of U	ser-Agent and Se	rver Headers						No	
D	iscard Received	I P-Asserted-Iden	tity Headers						No	
D	omain for Trunk	Context								
E	mergency Call I	Headers							CESID in From,	[and PAI]
E	.164: Enable se	nding '+'							Yes	
E	.164: Add '+' if o	ligit length > N di	gits						0	
E	.164: Do not ad	d '+' to Emergenc	y Called Party						No	
E	.164: Do not ad	d '+' to Called Par	ty						No	
F	orce Max-Forwa	rd: 70 on Outgoir	ng Calls						No	
If	TLS use 'sips:'	Scheme							No	
10	nore Incoming	Loose Routing In	dication						No	
Ir	clude Diversion	Header for EHD	U						No	
N	ode for Out-of-	Band DTMF							RFC 4733 DTM	IF
N	ultilingual Nam	e Display							No	
0	nly use SDP to	decide 180 or 183	\$						Yes	
Р	refer From Head	der for Caller ID							No	
Q	.850 Reason He	aders							No	
R	equire Reliable	Provisional Resp	onses on Outg	oing Calls					No	
S	uppress Incomi	ng Name							No	
5	uppress Redire	ction Headers							No	
U	se Fixed Retry	Time for 491							No	
U	se Privacy: non	e							No	
Us	e Privacy: non	e								No
Us	e P-Asserted I	dentity Header								Yes
Us	e P-Asserted I	dentity for Billin	g							No
Us	e P-Call-Leg-I) Header								No
Us	e P-Early-Med	ia Header								No
Us	e P-Preferred I	dentity Header								No
Us	e Restricted C	haracter Set Fo	r Authenticatio	on						No
Us	e To Address i	n From Header	on Outgoing (Calls						No
Us	e user-phone									No
Us	e user=phone	for Diversion H	eader							No
Us	er-Defined Hea	ader Name								
Us	er-Defined Hea	der Value								

Figure 10 – SIP Peer Profile Assignment - Signaling and Header Manipulation

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information	
Ke	ep-Alive (OPTIC	ONS) Period							120
Re	gistration Perio	d							3600
Re	gistration Perio	d Refresh (%)							50
Re	gistration Maxi	mum Timeout							90
Se	ssion Timer								0
Se	ssion Timer: Lo	cal as Refresher							No
Su	bscription Peri	bd							3600
Su	bscription Peri	od Minimum							300
Su	bscription Peri	od Refresh (%)							80



Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information		
Alle	w Inc Subscrip	tions for Local [Digit Monitoring	1						No
Alle	ow Out Subscri	ptions for Remo	e Digit Monitor	ing						No
For	ce Out Subscri	ptions for Remo	te Digit Monitor	ing						No
Re	quest Outboun	I Proxy to Handle	e Out Subscrip	tions						No
KP	ML Transport									default
KP	ML Port									0

Figure 12 – SIP Peer Profile Assignment - Key Press Event

Basi	c Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information
								Update
In	dex			DID Range			CPN	Substitution

Figure 13 – SIP Peer Profile Assignment - Outgoing DID Ranges

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information
Cre	ator							
Dat	e Created							
Cre	ated with Vere	ion						
- Cie	aleu with versi	on						
501	vice Provider							
Ver	dor Notes							

Figure 14 – SIP Peer Profile Assignment - Profile Information

SIP Peer Profile Assignment by Incoming DID

This form is used to associate DID range numbers from Service Provider phonestar* with or without an asterisk SIP trunk to a particular SIP Peer profile. The configured settings here help matching the incoming DID numbers with the SIP Peer Profile when call is arriving from anonymous caller. 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, "41445896887,41916896751"). 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 characters maximum, you can create a new entry for the same profile.

Mitel MiVoice Busine	ess		Node Alarm Status: Clear 2025-Feb-19 07	08:32	D	?		۶ _®	1	Đ
Local_224	S	SIP Peer Profile Assignment by Incoming DID on Local_224	earch DN 🗸		Sho	v form on	Not Acce	ssible	¥ 60	Ŧ
Licenses	Î	Add Change Delete			Print	Impo	rt E	xport	Data R	tefresh
Voice Network System Properties		Incoming DID Range	SIP Peer Profile Label Phonestar		_	_	Commer	nt	_	_
Hardware Trunks Trunk Attributes	Ī	Incoming DID Range		41445896887				ST.		_
IP/XNET SIP		Comment		Phonestar						
DID Ranges for CPN Substitution SIP Peer Profile										
SIP Peer Profile Assignment by Incoming DID SIP Peer Profile Called Party Inward Dialing Modification	on									
SIP Peer Profile Calling Party Inward Dialing Modificat	ion									

Figure 15 – SIP Peer Profile Assignment by Incoming DID

ARS Digit Modification Plans

Ensure that Digit Modification for outgoing calls on the SIP trunk to phonestar* SIP Trunking absorbs or injects additional digits according to your dialling plan. In this example, we will be absorbing 3 digits.

🕅 Mitel 🕴 MiVoic	e Business				Node Alarm Status:	Clear 2025-Feb-19
Local_224	ARS D	git Modification Plans or	Local_224	Search DN	,	
Licenses	Cha	nge Change Page	Change All Clear	l		
LAN/WAN Configuration		Page 1 of 55 >	Go to	Value	Go	
Voice Network System Properties	🥔 A	RS Digit Modificatio	on Plans			
Hardware	Digit	Modification Number	Number	of Digits to Absorb		Digits to be Inserted
Trunks	1		3			
Users and Devices	2		0			
Integrated Directory Services	3		0			
Voice Mail	4		0			
Call Routing	5		0			
ARS Call Progress Tone Detection	6		0			
ARS Digit Modification Plans	7		0			
ARS Maximum Dialed Digits 🥔	8		0			
ARS Routes	9		0			

Figure 16 – Digit Modification Assignment

ARS Routes

Create a route for SIP trunks connecting a trunk to phonestar* with or without an asterisk SIP Trunking. In this example, the SIP trunk is assigned to Route Number **1**. Choose SIP Trunk as a routing medium and choose the SIP Peer Profile and Digit Modification entry created earlier.

Mitel MiVoice Busir	Node Alarm Status:							
Local_224	ARS Routes on Local_224 Search DN V							
Licenses LAN/WAN Configuration	Change Change All Clear Change Change							
Voice Network System Properties	ARS Routes							
Hardware	Route Number 1 Routing Medium SIP Trunk							
Users and Devices	Trunk Group Number							
Voice Mail	PBX Number / Cluster Element ID							
Call Routing Automatic Route Selection (ARS)	COR Group Number 1							
ARS Call Progress Tone Detection 🎺 ARS Digit Modification Plans 🞺	Digits Before Outpulsing							
ARS Maximum Dialed Digits 💣	Route Type PSTN Access Via DPNSS ▼ Compression Off							
ARS Route Lists ARS Route Plans								
ARS Digits Dialed	Save Cancel							

Figure 17 – SIP Trunk Route Assignment

ARS Digits Dialed

ARS initiates the routing of SIP trunk calls when certain digits are dialed from a station. In this example, when a user dials 111918123347168, the call will be routed to phonestar* SIP Trunking.

Mitel MiVoice Busin	ess				Node Alarm Statu:
Local_224	ARS Digits Dialec	d on Local_224		Search DN 🗸]
	Change				
Licenses	Change Range	e Programming - AR	S Digits Dialed	Help	
Voice Network	This form allows yo	ou to change one or more r	ecords, starting at th	ne following record:	
System Properties	Digits Dialed N	lumber of Digits to Follow	Termination Type	Termination Number	
Trunks	111 U	Inknown	Route	1	
Users and Devices					1
Integrated Directory Services	1. Enter the numb	ber of records to change:	1		
Voice Mail	2. Define the Cha	ange Range Programming I	Pattern:		
Call Routing	Field Name	Change	Value to char	ige Incre	ment by
Automatic Route Selection (ARS)	Digits Dialed	Change to	× 111		
ARS Call Progress Tone Detection 🥔	Number of Digit	s to Follow			
ARS Digit Modification Plans					
ARS Maximum Dialed Digits 🛹	Termination Typ	Change to	✓ Route ✓	-	
ARS Routes	Termination Nur	mber Change to	√ 1		
ARS Route Lists	4				×
ARS Route Plans				Preview Save	Cancel
ARS Digits Dialed					

Figure 18 – ARS Digit Dialed Assignment

<u>Note</u> – This configuration is just a reference. Customers can always change the configuration according to their requirements. If any customization is required, the customer should refer to the product guides which provides detailed information.

Fax Configuration

Service Provider phonestar* with or without an asterisk uses the inter-zone FAX profile. This form allows you to define the settings for FAX communication over the IP network. You can modify the default settings for the:

- Inter-zone FAX profile: defines the FAX settings between different zones in the network. There is only one Inter-zone FAX profile; it applies to all inter-zone FAX communication. It defaults to V.29, 7200bps. It defines the settings for FAX Relay (T.38) FAX communication.
- Intra-zone FAX profile: defines the FAX settings within each zone in the network.
 - Profile 1 defines the settings for G.711 pass through communication.
 - Profile 2 to 64 define the settings for FAX Relay (T.38) FAX communication.
 - All zones default to G.711 pass through communication (Profile 1).

🕅 Mitel 🕴 міve	oice Bu	siness	Node	e Alarm Status: Clear 2025-Feb-19 07:08:32	D	? 🗉	୵ୢୄଡ଼	③ ₽
Local_224	ź	Fax Service Profiles on Local_224	Search DN 🗸		Show	form on Not Acc	cessible	~ Go †
Licenses	Î	Change			Print	Import	Export	Data Refresh
LAN/WAN Configuration		🗳 Inter-Zone Fax Profile						
Network Elements 🎺		Maximum Fax Rate		14400 (V.17,	14400bps)			
Cluster Elements 🧬 Analog Gateway Servers		High Speed Redundancy Low Speed Redundancy		1				
Admin Groups Fax Service Profiles 🖨		Error Correction Mode (ECM)		Disabled				
Fax Advanced Settings			. Mahar					
Network Zone Topology 💉		Page 1 or / So to	Value	Go Change Member Chang	ge Page Members	Change All M	Members	Clear Member
Codec Settings		💉 Intra-Zone Fax Service Profiles						
Mass Audio Notification 47	_	Profile Maximum Fax Rate High Speed	Redundancy Low Speed Redundancy	Error Correction Mode NSF Override	NSF Vendor Code \	Value NSF Co	ountry Code V	alue Label
Hardware Trunks		1 2 14400 (V.17, 14400bps) 1	- 3	Disabled Disabled	-	-		G.711 T.38
Users and Devices	_	3 .						

Figure 19 - Fax Configuration

Glossary

MiVoice Business	MiVB
MiNET Interface	MINET
Mitel Solutions Alliance	MSA
Knowledge Management System	KMS
Class of Service	COS
Automatic Route Selection	ARS
Personal Ring Group	PRG
External Hot Desk User	EHDU
Automatic Call Distribution	ACD