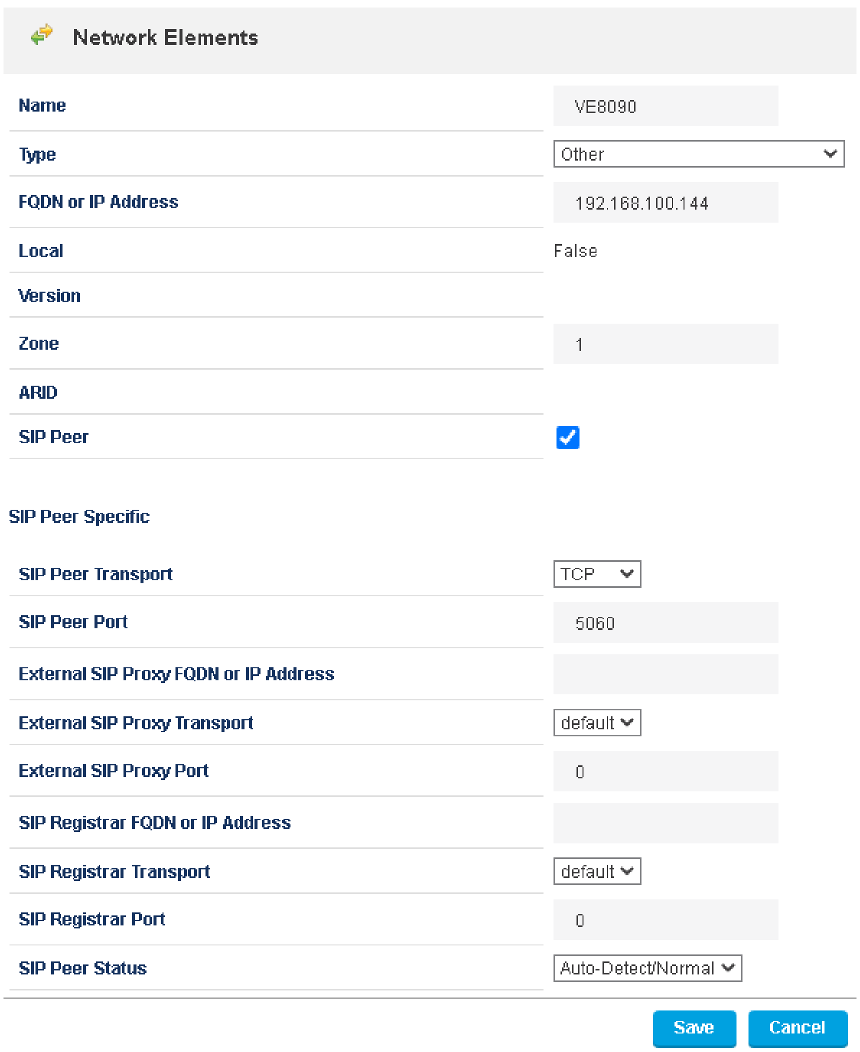
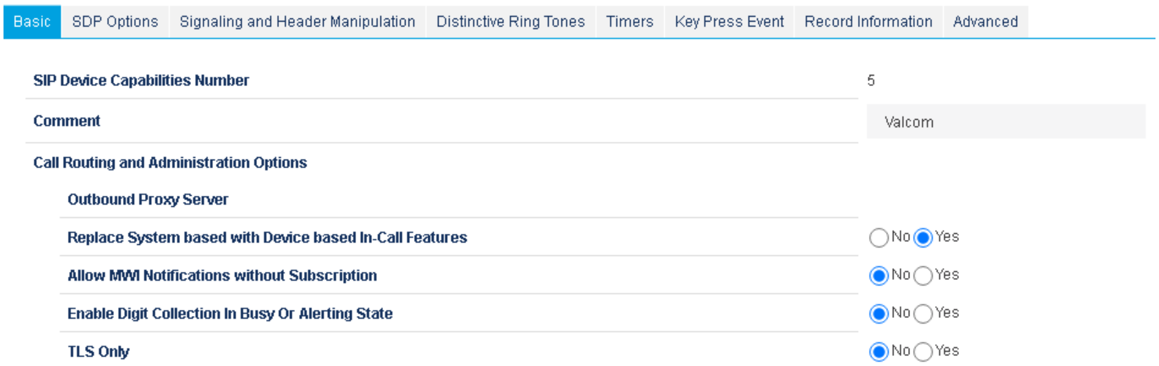
Valcom IP devices can communicate with Mitel MiVoice Business systems over a SIP Trunk connection. The following example illustrates the programming on a MiVoice virtual 3300 ICP release 9.0 SP3 system and a Valcom VE8090 SIP Controller.

The basic process is to create a Network Element representing the Valcom device, define several options in the SIP Device Capabilities, define a SIP Trunk Peer Profile and define the Call Routing to direct calls to the Valcom device. Other settings may be required in the Mitel system for specific site requirements related to your installation.

Login to the MiVoice Business System Administration Tool and navigate to the Voice Network → Network Elements page. Fill in the appropriate fields for Name, IP Address and check the box for SIP Peer. In the Peer Specific area, Transport can be either TCP or UDP. If the default Peer Port of 5060 is changed here, it will also need to be changed in the Valcom VE8090 programming. Save the entries when completed.

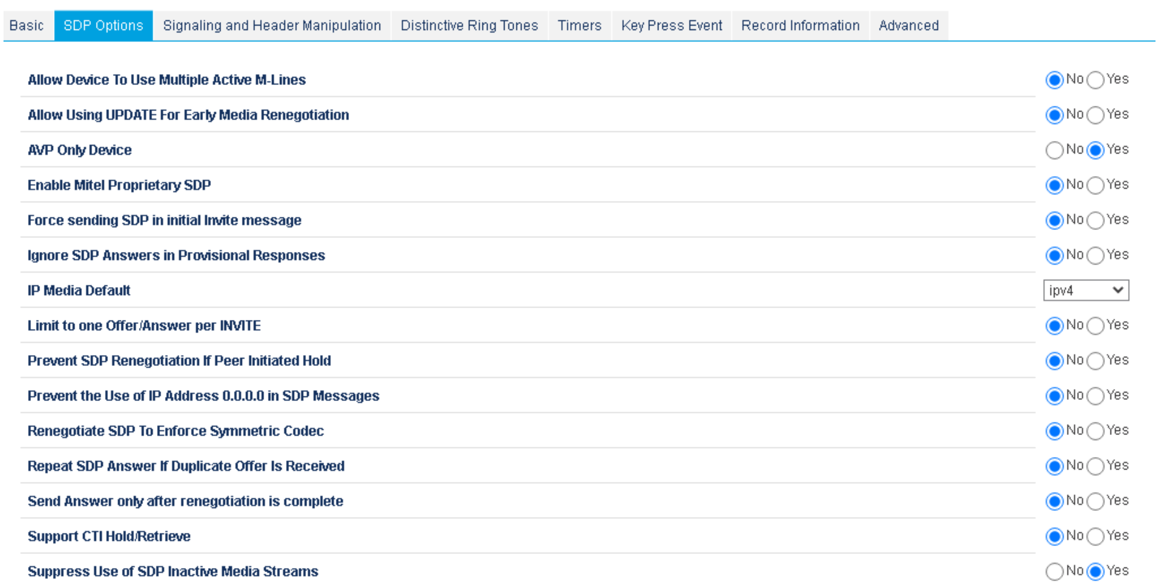


Navigate to the System Properties → System Feature Settings → SIP Device Capabilities page. If a SIP Device Capabilities has been defined with the necessary settings, it can be used instead of defining a new one for the Valcom device. The minimum settings are shown below.
On the Basic page, confirm *TLS Only* is set for *No*, and *Replace System based with Device based In-Call Features* is set to *Yes*.


Basic | SDP Options | Signaling and Header Manipulation | Distinctive Ring Tones | Timers | Key Press Event | Record Information | Advanced

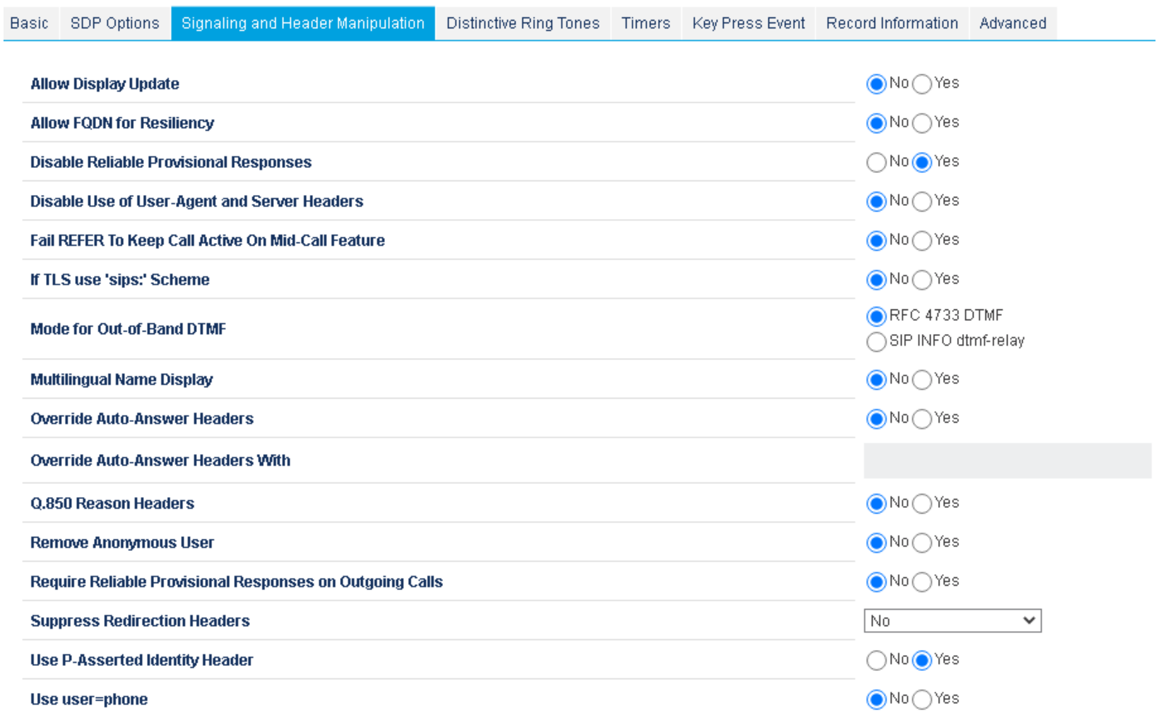
SIP Device Capabilities Number	5
Comment	Valcom
Call Routing and Administration Options	
Outbound Proxy Server	
Replace System based with Device based In-Call Features	Yes
Allow MWI Notifications without Subscription	No
Enable Digit Collection In Busy Or Alerting State	No
TLS Only	No

On the SDP Options page, the default values as shown below are sufficient.


Basic | SDP Options | Signaling and Header Manipulation | Distinctive Ring Tones | Timers | Key Press Event | Record Information | Advanced

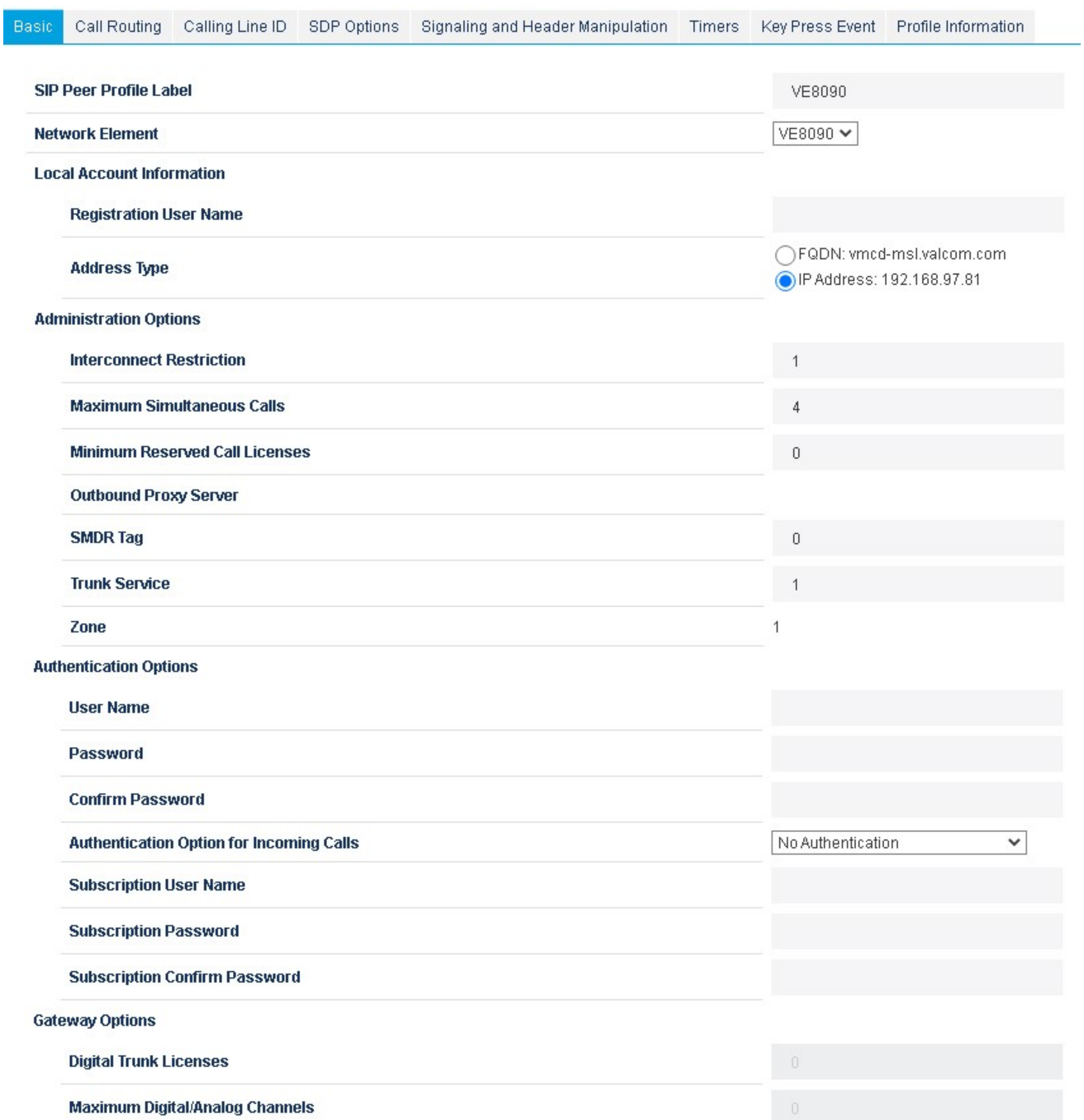
Allow Device To Use Multiple Active M-Lines	No
Allow Using UPDATE For Early Media Renegotiation	No
AVP Only Device	Yes
Enable Mitel Proprietary SDP	No
Force sending SDP in initial Invite message	No
Ignore SDP Answers in Provisional Responses	No
IP Media Default	ipv4
Limit to one Offer/Answer per INVITE	No
Prevent SDP Renegotiation If Peer Initiated Hold	No
Prevent the Use of IP Address 0.0.0.0 in SDP Messages	No
Renegotiate SDP To Enforce Symmetric Codec	No
Repeat SDP Answer If Duplicate Offer Is Received	No
Send Answer only after renegotiation is complete	No
Support CTI Hold/Retrieve	No
Suppress Use of SDP Inactive Media Streams	Yes

On the Signaling and Header Manipulation page, confirm *Disable Reliable Provisional Responses* is set to *Yes*.

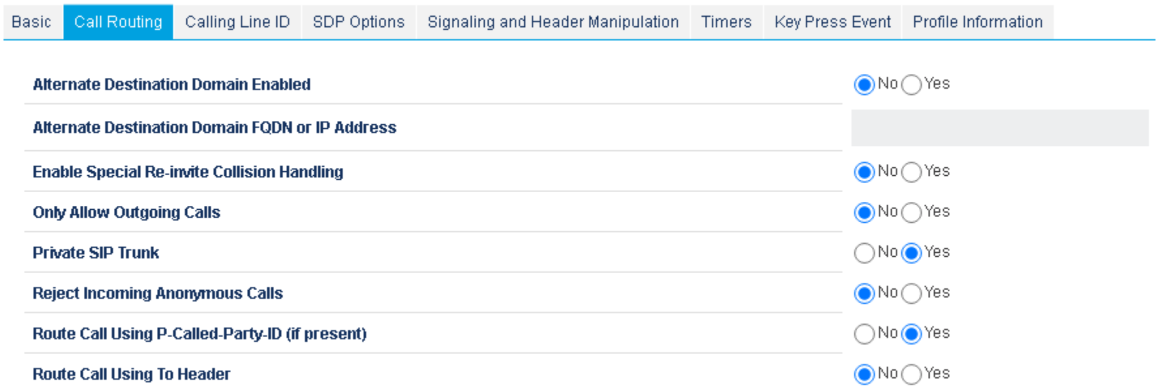


Programming on the remaining Tabs in the SIP Device Capabilities can be left at the default values.

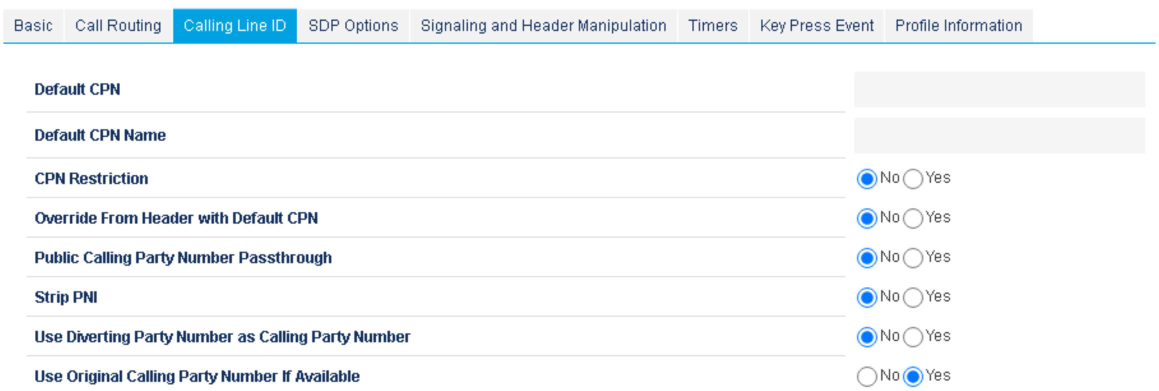
Navigate to the Trunks → SIP → SIP Peer Profile page, and create a new SIP Peer Profile. Enter a descriptive name for the *SIP Peer Profile Label* and select the *Network Element* created previously for the Valcom device. Set the *Maximum Simultaneous Calls* to 4. Other items in the Administration Options should be reviewed for any site-specific settings, such as Zone. The *Authentication Options* and *Gateway Options* can be left blank.



On the Call Routing page, set *Private SIP Trunk* to *Yes*.



On the Calling Line ID page, set *Use Original Calling Party Number if Available* to *Yes*.



The SDP Options page can be left at default values.

On the Signaling and Header Manipulation page, set *Disable Reliable Provisional Responses* to *Yes*, set *Require Reliable Provisional Responses on Outgoing Calls* to *No*.

Basic | Call Routing | Calling Line ID | SDP Options | **Signaling and Header Manipulation** | Timers | Key Press Event | Profile Information

Trunk Group Label	
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	
E.164: Enable sending '+'	No
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
Include Diversion Header for EHDU	No
Mode for Out-of-Band DTMF	RFC 4733 DTMF (options: RFC 4733 DTMF, SIP INFO dtmf-relay)
Multilingual Name Display	No
Only use SDP to decide 180 or 183	Yes
Prefer From Header for Caller ID	No
Q.850 Reason Headers	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-Early-Media 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

Navigate to the Call Routing → Automatic Route Selection (ARS) → ARS Routes page. Select an available Route Number and click Change. For *Routing Medium*, select *SIP Trunk*. For *SIP Peer Profile*, select the profile created previously.

ARS Routes

Route Number	3
Routing Medium	SIP Trunk
Trunk Group Number	
SIP Peer Profile	VE8090
PBX Number / Cluster Element ID	
COR Group Number	1
Digit Modification Number	1
Digits Before Outpulsing	
Route Type	Non-verified Account
Compression	Off

Navigate to the Call Routing → Automatic Route Selection (ARS) → ARS Digits Dialed page. Leave the number of records to add at 1. In the *Digits Dialed* field, enter the digits that will be dialed by a phone user to call the VE8090. In this example, 60000 will be used as the phone number to call. For *Termination Type* select *Route* and for *Termination Number* select the *Route Number* that was used in the previous step.

Add Range Programming - *ARS Digits Dialed* Help

This form allows you to add one or more records.

1. Enter the number of records to add: 1

2. Define the Add Range Programming Pattern:

Field Name	Value to Add	Increment by
Digits Dialed	60000	
Number of Digits to Follow	0	-
Termination Type	Route	-
Termination Number	3	

Programming the Valcom device for SIP communication is done using the VIP-102B IP Solutions Setup Tool. The information contained in this guide is limited to configuration of the “SIP” tab in the VIP-102B Setup Tool for the Valcom VIP device. More information on Valcom VIP device configuration, such as IP address assignment, relay activation, etc., may be found in the VIP-102B Reference Manual. This document may be downloaded from our website at www.valcom.com
The example below is for programming the VE8090 SIP Controller in Trunk mode.

For *SIP Mode* select *Trunk*, and for *Transport* select the same protocol that was set in the SIP Peer Transport on the Network Element page. For this example, *Accept: TCP+UDP, Originate: TCP* would be the correct choice.

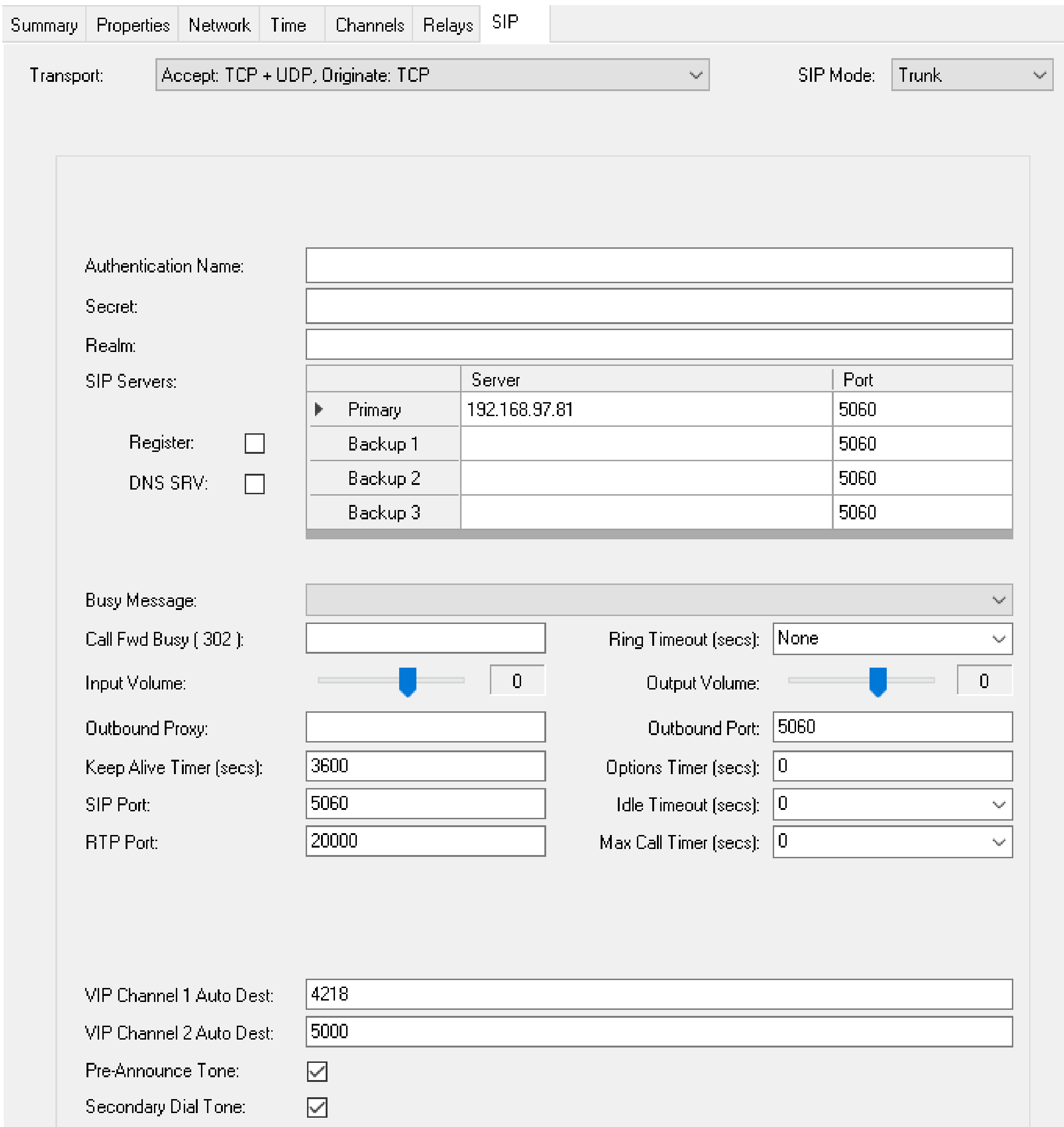
Enter the IP address of the Mitel SIP PBX for the Primary SIP Server. All of the Port values may be left at default, unless they were changed in the Mitel configuration previously.

The *VIP Channel 1 Auto Dest* and *VIP Channel 2 Auto Dest* fields contain the phone numbers of the Mitel phones that should be called when a Valcom IP speaker places an intercom call to either the Channel 1 or Channel 2 dial code. In this example, intercom calls from an IP Speaker to the VE8090 Channel 1 will be forwarded to the SIP phone number 4218.

The *Secondary Dial Tone* check box controls whether the VE8090 will prompt for user input by playing Dial Tone to the caller, or whether the VE8090 will attempt to route the call based on the SIP phone number sent to the VE8090. When the Secondary Dial Tone box is checked, the VE8090 will answer an incoming call from SIP and play dial tone to the caller. The caller will then key in the Valcom paging group or device dial code they wish to contact. This is typically referred to as “two-stage dialing”.

If the Secondary Dial Tone box is not checked, the VE8090 will examine the phone number sent to it in the SIP INVITE message and try to match it to a Valcom paging group or device dial code. With this method, the Route Pattern defined in the Mitel ARS Dialed Digits page will be defined as one or more leading digits followed by a set number of additional digits. In our example, instead of the dialed digits being “60000” followed by no additional digits it could be defined as “60” followed by 3 digits. Calling to 60000 or 60999 or any other 5-digit number starting with 60 would be routed to the VE8090. The VE8090 will use its Dial Code Length setting (defined under the System menu in the VIP-102B tool) to determine how many digits of the SIP number to use. The default Dial Code Length is 3, so for this example a call to 60000 will try to match 000 to a Valcom dial code. A call to 60999 will try to match 999 to a Valcom dial code.

For this example, the Secondary Dial Tone box is ticked and the VE8090 will play dial tone to the caller.


Summary
Properties
Network
Time
Channels
Relays
SIP
Transport:
Accept: TCP + UDP, Originate: TCP
SIP Mode:
Trunk
Authentication Name:
Secret:
Realm:
SIP Servers:
Server
Port
Primary
192.168.97.81
5060
Backup 1
5060
Backup 2
5060
Backup 3
5060
Register:
DNS SRV:
Busy Message:
Call Fwd Busy (302):
Ring Timeout (secs):
None
Input Volume:
0
Output Volume:
0
Outbound Proxy:
Outbound Port:
5060
Keep Alive Timer (secs):
3600
Options Timer (secs):
0
SIP Port:
5060
Idle Timeout (secs):
0
RTP Port:
20000
Max Call Timer (secs):
0
VIP Channel 1 Auto Dest:
4218
VIP Channel 2 Auto Dest:
5000
Pre-Announce Tone:
Secondary Dial Tone: