



## **M1 SIP Trunk Configuration with Beronet Telephone Appliance**

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## About M1

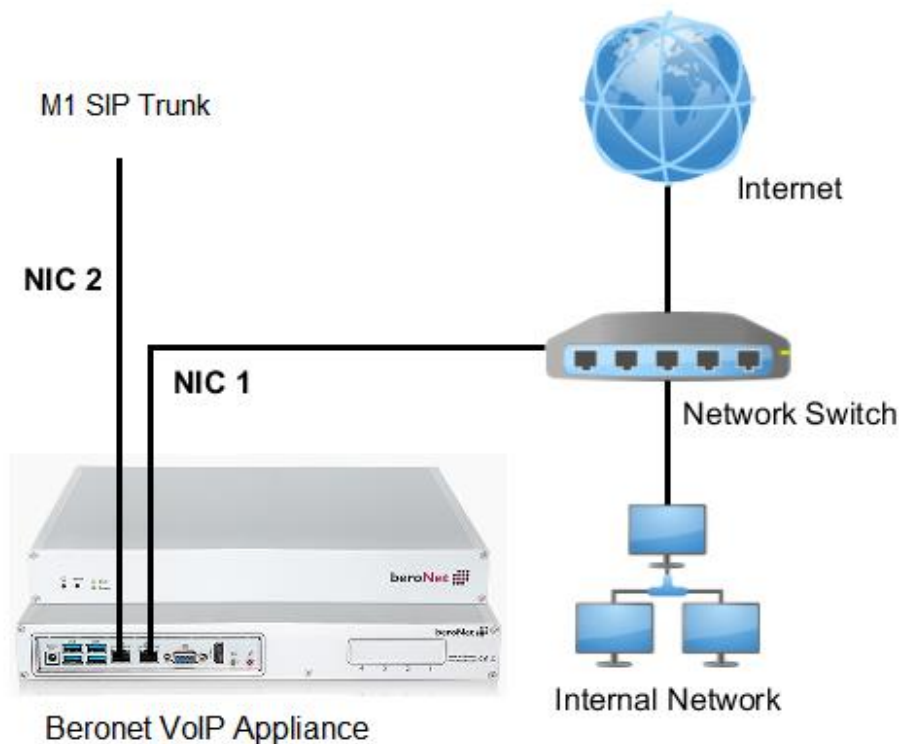
M1 is one of the three major full-service communications providers in Singapore. It offers a suite of mobile voice-and-data communication services over its 3G/3.5G/4G/LTE-A network, including international-call services to both mobile- and fixed-line customers.

These include SMS, MMS, WAP, GPRS, 3G, 3.5G and 4G and was the first Singapore operator to launch a nationwide 4G LTE and 4G LTE-A network. It also offers prepaid mobile services, such as prepaid data plans, under its M Card brand. It is one of the operators in Singapore to offer a prepaid 4G service. Singapore's 2G networks, including M1's, was turned off in April 2017.

## System Preparation

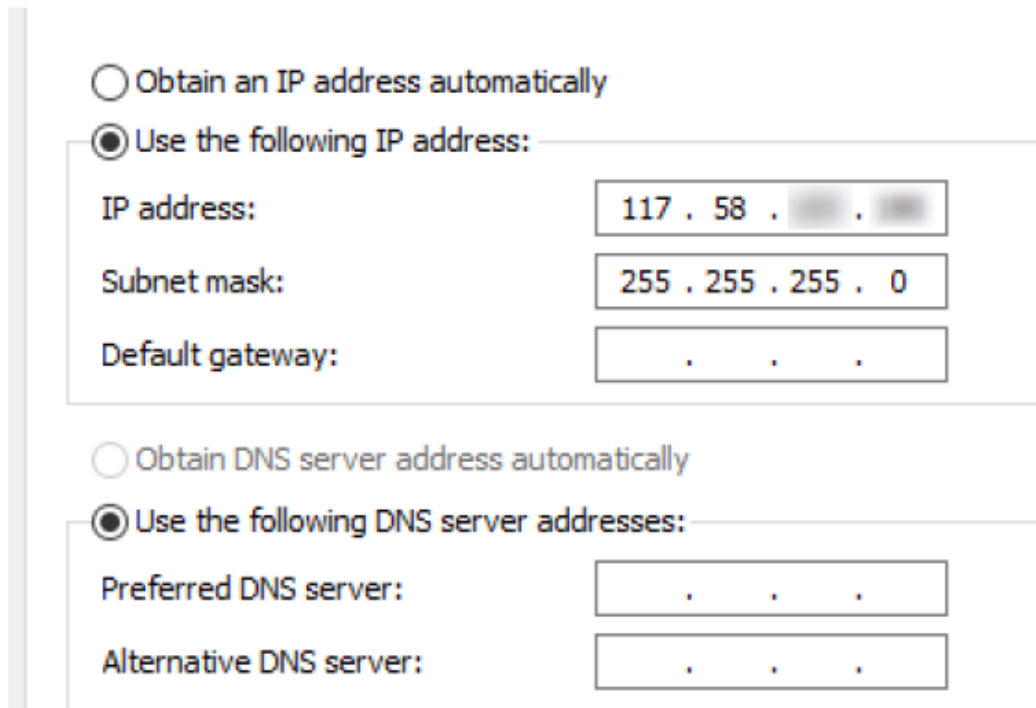
### Network Topology

In order to use a M1 SIP Trunk with 3CX, the 3CX Server needs to have a second network card which will only be used for traffic from and to the M1 servers. The network topology should look similar to the following:



NIC 1 should connect to the main LAN through which the 3CX Server will have access to the internet and will also be the LAN on which Extensions are connected to.

NIC 2 will only cater for traffic from/to the M1 servers.



Obtain an IP address automatically  
 Use the following IP address:

IP address:   
 Subnet mask:   
 Default gateway:

Obtain DNS server address automatically  
 Use the following DNS server addresses:

Preferred DNS server:   
 Alternative DNS server:

**Important!** NIC 2 must not have a Default Gateway configured and the DNS servers must be blank.

## Adding Static Routes

Next you must add a static route so that the traffic from the 3CX server towards the M1 servers is routed out through the correct Gateway. To do this on a Windows OS, you would open a command prompt using the “**Run as Administrator**” option, then run the following command:

```
route -p add 172.16.126.0 mask 255.255.255.0 METRIC 1 [Gateway IP given by Provider]
```

e.g.:

```
route -p add 172.16.126.0 mask 255.255.255.0 METRIC 1 117.58.169.189
```

The Gateway IP that you need to configure should be given to you by M1.

## 3CX Server NIC Setup

For NIC 1, apply the necessary network settings required by your network.

For NIC 2, you must manually configure the M1 WAN IP address that has be given to you and as per the instruction by M1, e.g.:

## 3CX Version

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Some providers gained support and compatibility with 3CX on a specific product version. It is advisable to always run the latest version of 3CX to ensure ongoing compatibility.

Minimum 3CX Version: 3CX Phone System 15.5

## Provider Capabilities

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Below is a short overview of the provider's capabilities and services and whether they're supported or not:

- 1) CLNS (Clip No Screening): No
- 2) Catch All Routing: Yes, static SI per trunk
- 3) Fax in T38: No
- 4) CLIR (Number Suppression): No
- 5) DTMF via RFC 2833: Yes
- 6) Codec Order: G711A,G729, G711U, -
- 7) NAT Support: Yes
- 8) Other: SRTP is not supported, TLS is not supported

## Collecting 3CX Configuration Settings

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In order to configure M1 with 3CX, you should first make sure you have the following information available which must be provided to you by your M1 representative:

- Your DID numbers
- The M1 SIP Server IP address
- Your M1 WAN IP Address

With the above information you can proceed to the next section which explains how you use this information to configure the Trunk in 3CX.

## Configuring the Trunk with 3CX

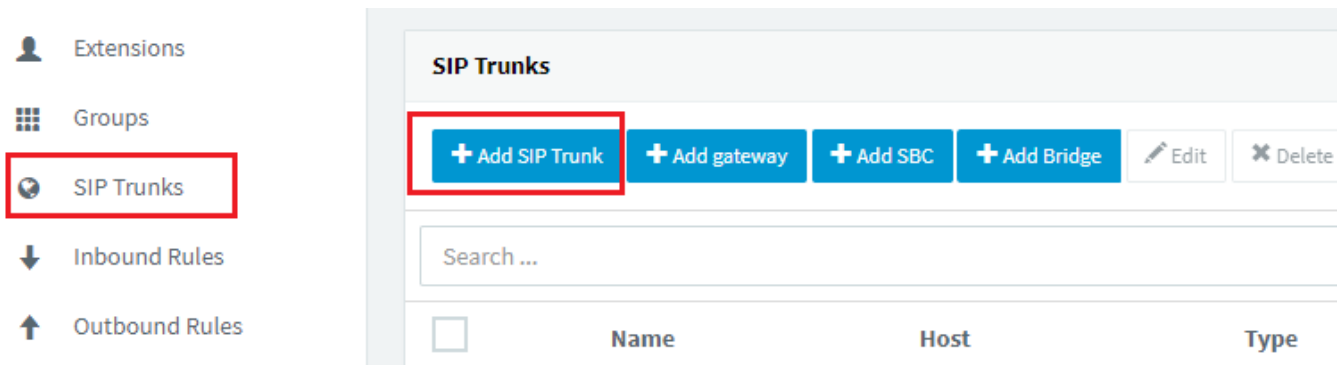
### Adding the Trunk

The general instructions outlining how to add a new SIP Trunk to your 3CX

Additionally, for M1, you must also have a dedicated NIC for the traffic from/to the Provider. The instructions for this are provided in section “System Preparation” and must be done prior to the following.

To add M1 to your 3CX system, open the Management Console and navigate to “SIP Trunks”. Press the “Add SIP Trunk” button and select “Generic” as the country and then “VoIP Trunk Provider” from the following drop-down.

In the “Main Trunk No” field, enter one of your DID numbers, then press OK.



The screenshot shows the 3CX Management Console interface. On the left, a sidebar lists navigation options: Extensions, Groups, SIP Trunks (highlighted with a red box), Inbound Rules, and Outbound Rules. The main panel is titled "SIP Trunks" and contains several action buttons: "+ Add SIP Trunk" (highlighted with a red box), "+ Add gateway", "+ Add SBC", "+ Add Bridge", "Edit", and "Delete". Below the buttons is a search bar labeled "Search ...". At the bottom, a table is visible with columns for "Name", "Host", and "Type".

### Add SIP Trunk/VoIP Provider

Select Country  
Generic

Select Provider in your Country  
Generic VoIP Provider

Main Trunk No  
+6536401XXX

OK Cancel

General | DIDs | Caller ID | Options | Inbound Parameters | Outbound Parameters

#### Trunk Details

Enter name for Trunk  
M1 SIP Trunk

Registrar/Server/Gateway Hostname or IP  
ims.mobileone.net.sg 5060

Outbound Proxy  
172.16.126.41 5060

Number of SIM Calls  
10

#### Authentication

Type of Authentication  
Do not require - IP Based

Authentication ID (aka SIP User ID)

Authentication Password

General   DIDs   Caller ID   **Options**   Inbound Parameters   Outbound Parameters

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**Call options**

- Allow inbound calls
- Allow outbound calls
- Disallow video calls

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**Advanced**

- PBX Delivers Audio
- Supports Re-Invite
- Support Replaces
- Put Public IP in SIP VIA Header
- Alternative Proxy
- SRTP

Re-Register Timeout

180

Select which IP to use in 'Contact' (SIP) and 'Connection'(SDP) fields

Use Default Settings

Transport Protocol

UDP

IP Mode

IPV4

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**Codec Priority**

- G.711 A-law
- G.711 U-law
- G729



M1 SIP Trunk OK Cancel

General DIDs Caller ID Options **Inbound Parameters** Outbound Parameters

**Caller Number/Name Field Mapping:**

Review the SIP header of the INVITE and specify where the following values should be present within the INVITE:

"CalledNum" number that has been dialed (default: To->user)

"CallerName" caller's name (default: From->display name)

"CallerNum" caller's number (default: From->user)

**Call Source Identification**

Configure this option only when the SIP Trunk is IP based (peering), or does not support automatic inbound call detection. If you have multiple trunk toggle this option (on/off) and see what configuration works best for this SIP Trunk

Use both "Call Source Identification" rules and "Caller Number/Name -> CalledNum" field mappings (Note: Disables catch all routing capability)

General DIDs Caller ID Options Inbound Parameters **Outbound Parameters**

**Outbound Parameters**

Assign SIP header fields to 3CX Call Variables. Requires advanced SIP knowledge. Misconfiguration will cause your PBX to malfunction

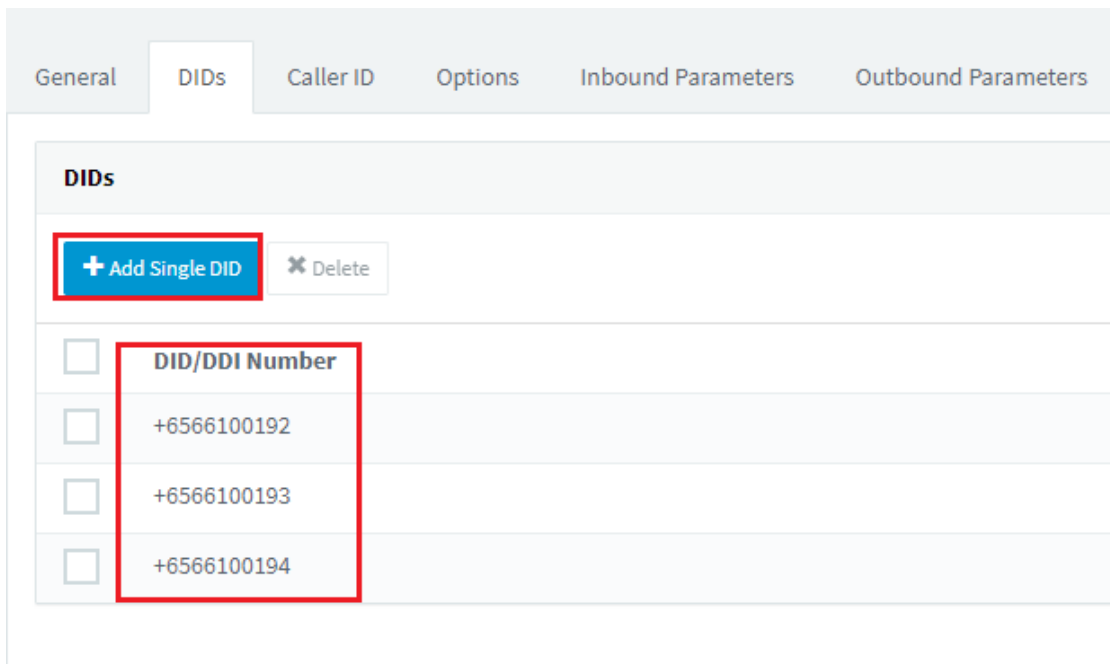
SIP Field	Variable
Request Line URI : User Part	"CalledNum" number that has been dialed (default: To->user)
Request Line URI : Host Part	"GWHostPort" gateway/provider host/port
Contact : User Part	"OutboundCallerId" Outbound caller Id taken from Extension
Contact : Host Part	"ContactUri" usually, content of Contact field
To : Display Name	"CalledName" name that has been dialed (default: To->displa
To : User Part	"CalledNum" number that has been dialed (default: To->user)
To : Host Part	"GWHostPort" gateway/provider host/port
From : Display Name	"OutboundCallerId" Outbound caller Id taken from Extension
From : User Part	"OutboundCallerId" Outbound caller Id taken from Extension
From : Host Part	"GWHostPort" gateway/provider host/port
User Agent : Text String	Leave default value
Remote Party ID - Called Party : Display Name	Leave default value
Remote Party ID - Called Party : User Part	Leave default value

Once you have done the above, press OK and your Trunk will now be configured.

## Adding Additional DIDs

To associate all other DIDs/Numbers you have on your M1 Trunk, what you need to do is go to the Management Console → SIP Trunks, double-click on your M1 Trunk and go to the “DIDs” tab.

Here you should already see 1 entry that is the Main Trunk number you have set. Add all other DIDs/Numbers you have to the list in the National Number format.



General **DIDs** Caller ID Options Inbound Parameters Outbound Parameters

**DIDs**

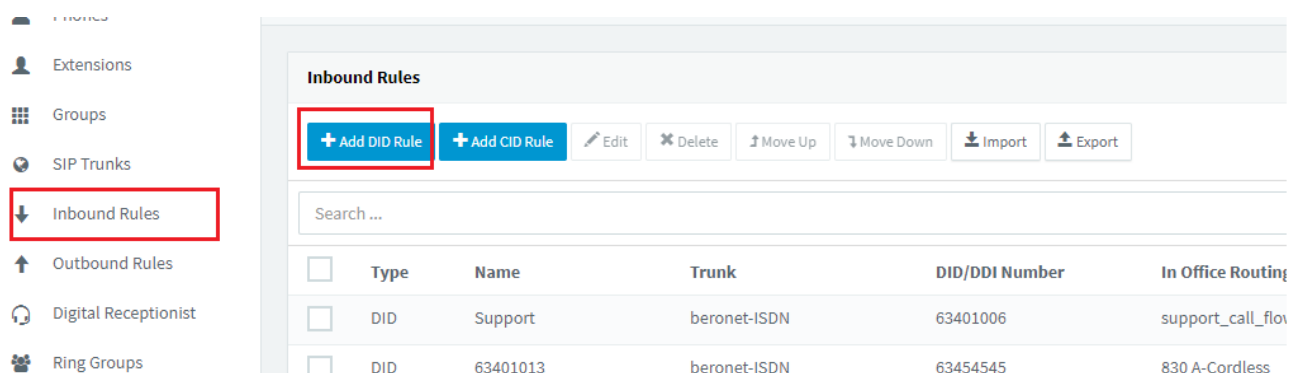
+ Add Single DID Delete

<input type="checkbox"/>	DID/DDI Number
<input type="checkbox"/>	+6566100192
<input type="checkbox"/>	+6566100193
<input type="checkbox"/>	+6566100194

## Creating Inbound Rules

Now that you have associated all your DIDs/Number with your SIP Trunk in 3CX, you can create Inbound Rules to set where calls will be routed when those numbers are called.

Go to Inbound rules and create inbound rule.



Inbound Rules

+ Add DID Rule + Add CID Rule Edit Delete Move Up Move Down Import Export

Search ...

<input type="checkbox"/>	Type	Name	Trunk	DID/DDI Number	In Office Routing
<input type="checkbox"/>	DID	Support	beronet-ISDN	63401006	support_call_flow
<input type="checkbox"/>	DID	63401013	beronet-ISDN	63454545	830 A-Cordless

## Add Inbound Rule

OK

Cancel

### General

Name

Inbound rule name

DID/DDI

+6566100192

### Route calls to

Destination for calls during office hours

Extension

1100 Active

Destination for calls outside office hours

Extension

1100 Active

Set up Specific Office Hours for this trunk

Play holiday prompt when it's a global holiday

# Number Format







## Outbound Caller ID

When making Outbound Calls using your M1 SIP Trunk, you can present any of your DID's as the Outbound Caller ID. Note though that because Clip No Screening is not supported, you cannot present any other number that you don't have associated with your Trunk.

## Outbound Rules

There are no special instructions regarding Outbound Rules that are required, numbers can be dialled either in any valid format.

Go to outbound Rules & Add outbound rule

-  Extensions
-  Groups
-  SIP Trunks
-  **Inbound Rules**
-  Outbound Rules
-  Digital Receptionist

### Outbound Rules

[+ Add](#) [Edit](#) [Delete](#) [Move Up](#) [Move Down](#)

Search ...

<input type="checkbox"/>	Outbound Rule Name	Call from extension(s)	Prefix	Length	Extension C
<input type="checkbox"/>	Test		9	1,2,3,4,5...	

Local

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**General**

Rule Name

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**Apply this rule to these calls**

Calls to numbers starting with prefix

Calls from extension(s)

Calls to Numbers with a length of

Calls from extension group(s)

---

**Make outbound calls on**

Configure up to 5 backup routes for outgoing calls. Each route can be configured differently.

Route		Strip Digits	Prepend
1	<input type="text" value="M1 SIP Trunk"/>	<input type="text" value="0"/>	<input type="text" value="+65"/>
2	<input type="text" value="BLOCK CALLS"/>	<input type="text" value="0"/>	<input type="text"/>

Outbound caller ID should be place in all the extension outbound caller id settings. +656340XXXX..