

Expanding Possibilities...

BUILDING YOUR OWN FREE HOME PHONE SYSTEM WITH MIKROTIK

By

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OVERVIEW



1. Hardware & Software Used In This Presentation.
2. What Is MetaROUTER?
3. How To Install MetaROUTER In Mikrotik RouterBOARD?
4. How To Install Asterisk 1.8 With GUI?
5. Basic Asterisk Server Configuration:
 - a. SIP Extension Configuration.
 - b. Register Your PC / Android Mobile Phone With Asterisk.
 - c. Register Analog Telephone Adapter (ATA) With Asterisk.
6. What Is Next?
 - a. Send/Receive Calls Using Your Asterisk Server While You Are Anywhere Across The Globe!
 - b. Connecting Two Asterisk Servers Together Via SIP Trunk.
 - c. Does Your Mikrotik Need To Have A Static IP Address?
 - d. Can We Use A Broadband USB Modem For Internet Connection?
 - e. Why Not Integrate Your Asterisk Server With Your Existing Business Telecommunication Systems?!
 - f. Can We Make Outbound Calls To PSTN Using Our Asterisk Server?

1. HARDWARE & SOFTWARE USED IN THIS PRESENTATION

RB951Ui-2HnD

Architecture	MIPSBE
CPU	AR9344
CPU core count	1
CPU nominal frequency	600 MHz
Operating System	RouterOS (version = 6.43.2)
Size of RAM	128 MB
Storage size	128 MB
Storage type	NAND

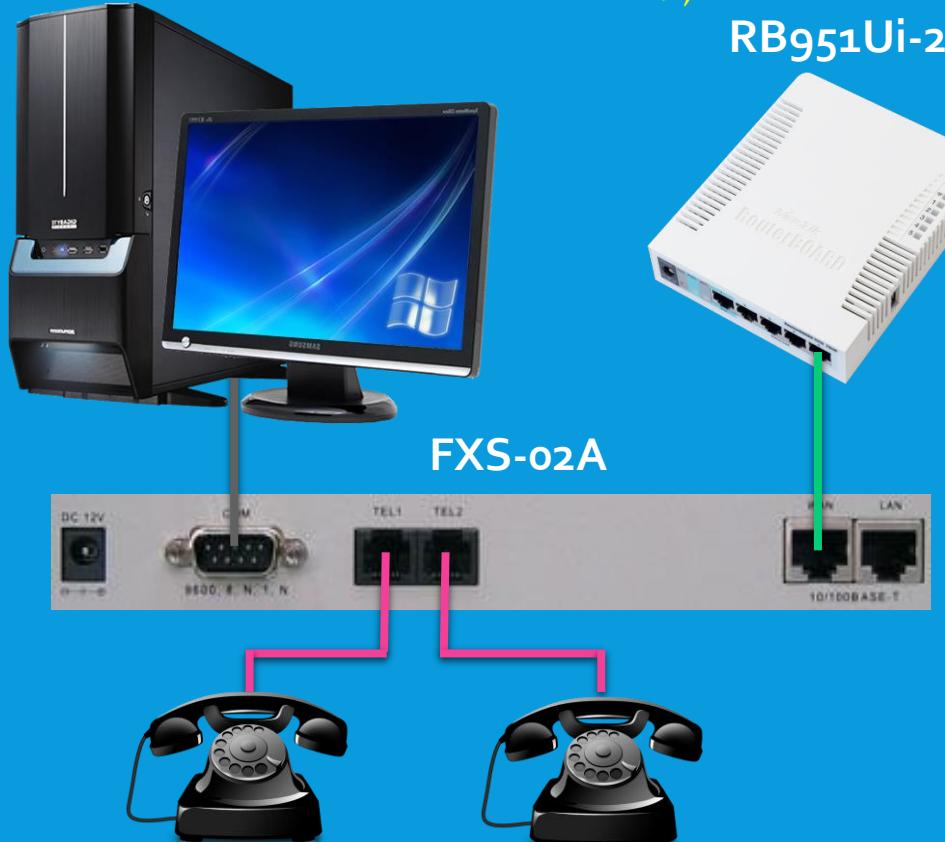


1. HARDWARE & SOFTWARE USED IN THIS PRESENTATION (CONTINUED...)

VoIP Gateways with 2 FXS ports (SIP)

PC

(for initial setup of FXS VoIP Gateway)



1. HARDWARE & SOFTWARE USED IN THIS PRESENTATION (CONTINUED...)

Free VoIP SIP Softphone Application for PC

X-Lite



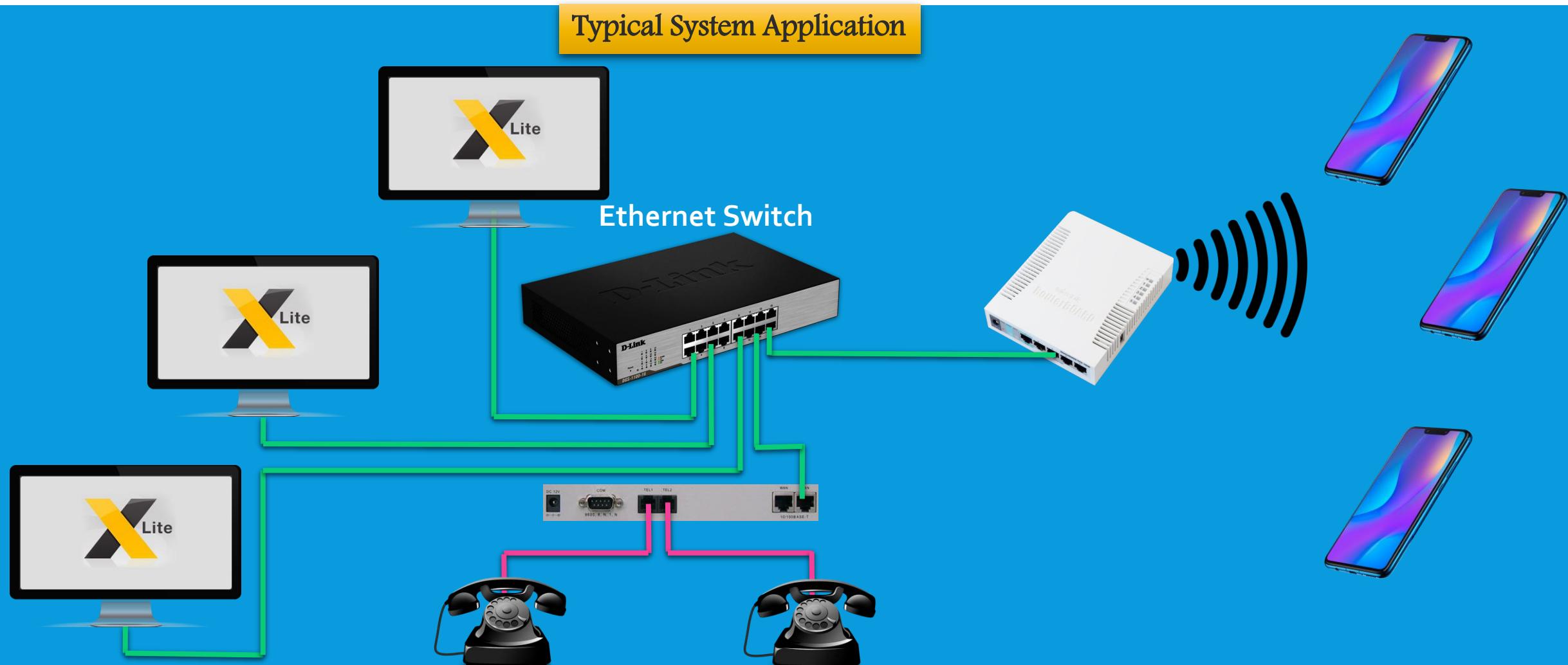
Download Link: <https://www.counterpath.com/XLiteForWindows>

Free VoIP SIP Softphone Application for Android



Download Link: <https://play.google.com/store/apps/details?id=com.csipsimple>

1. HARDWARE & SOFTWARE USED IN THIS PRESENTATION (CONTINUED...)



2. WHAT IS METAROUTER?

- MetaROUTER is a way to have logical routers running on your existing RouterBOARD.
 - Since v3.21 support for MetaROUTER on mipsbe platform,
 - Since v3.26 support for MetaROUTER on PPC (RB1000).
- Virtual environment allows user to partition system into different administrative domains.
- Able to run either RouterOS or OpenWRT patched Linux.
- Each RouterOS instance requires at least 16MB Ram, 32MB Ram recommended.
- Commonly deployed for customer administered router (RouterOS) or running specific simple task without need of dedicated server (Squid proxy, **Asterisk PBX**, Apache webserver).
- Currently MetaROUTER can be used on:
 - **RB400, RB700 series** except models with SPI flash, **RB900 series** except models with SPI flash, **RB2011** boards.
 - Listed PPC boards: **RB1000, RB1100, RB1100AH** and **RB800**.

2. WHAT IS METAROUTER?

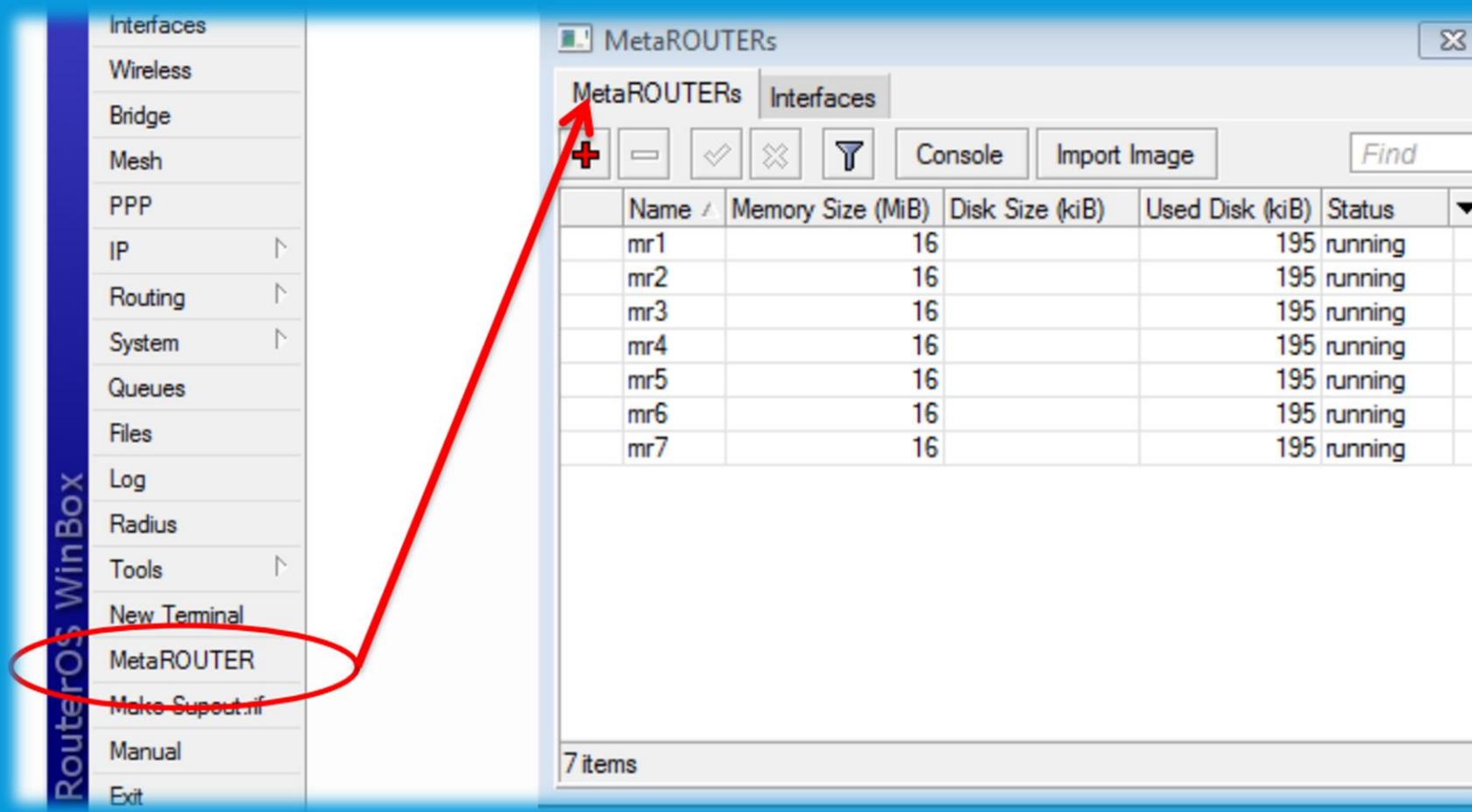
(CONTINUED...)

➤ **MetaROUTER Limitations & Faults:**

- Only 8 instances per RouterBOARD.
- No CF or microSD devices can be used for running images.
- No ability to export running virtual image back into a file.
- OpenWRT on MetaROUTER won't properly shutdown when RouterOS reboots.
- Limited by available Ram (256MB 450G).
- No ability to monitor running states with Dude Server.
- Host Router on occasion reboots with watchdog timeout error (V3.28)

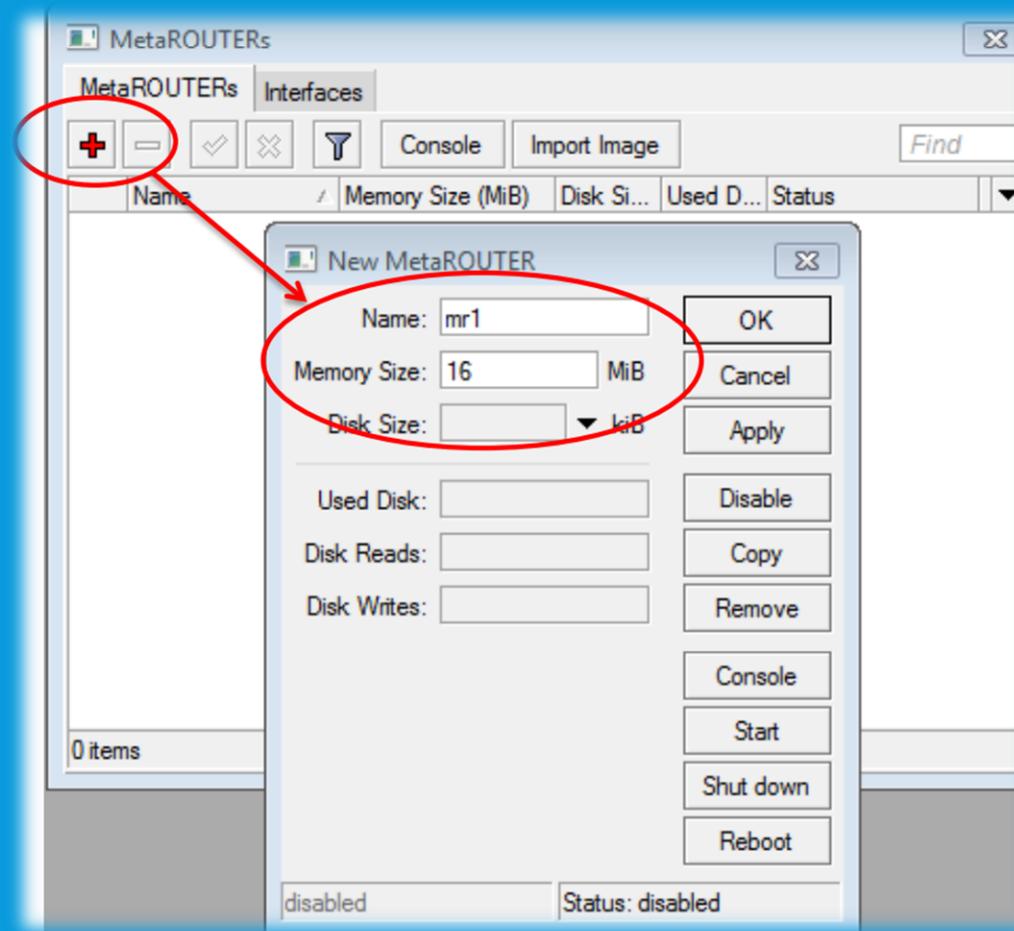
3. HOW TO INSTALL METAROUTER IN MIKROTIK ROUTERBOARD?

➤ The MetaROUTER Winbox Interface



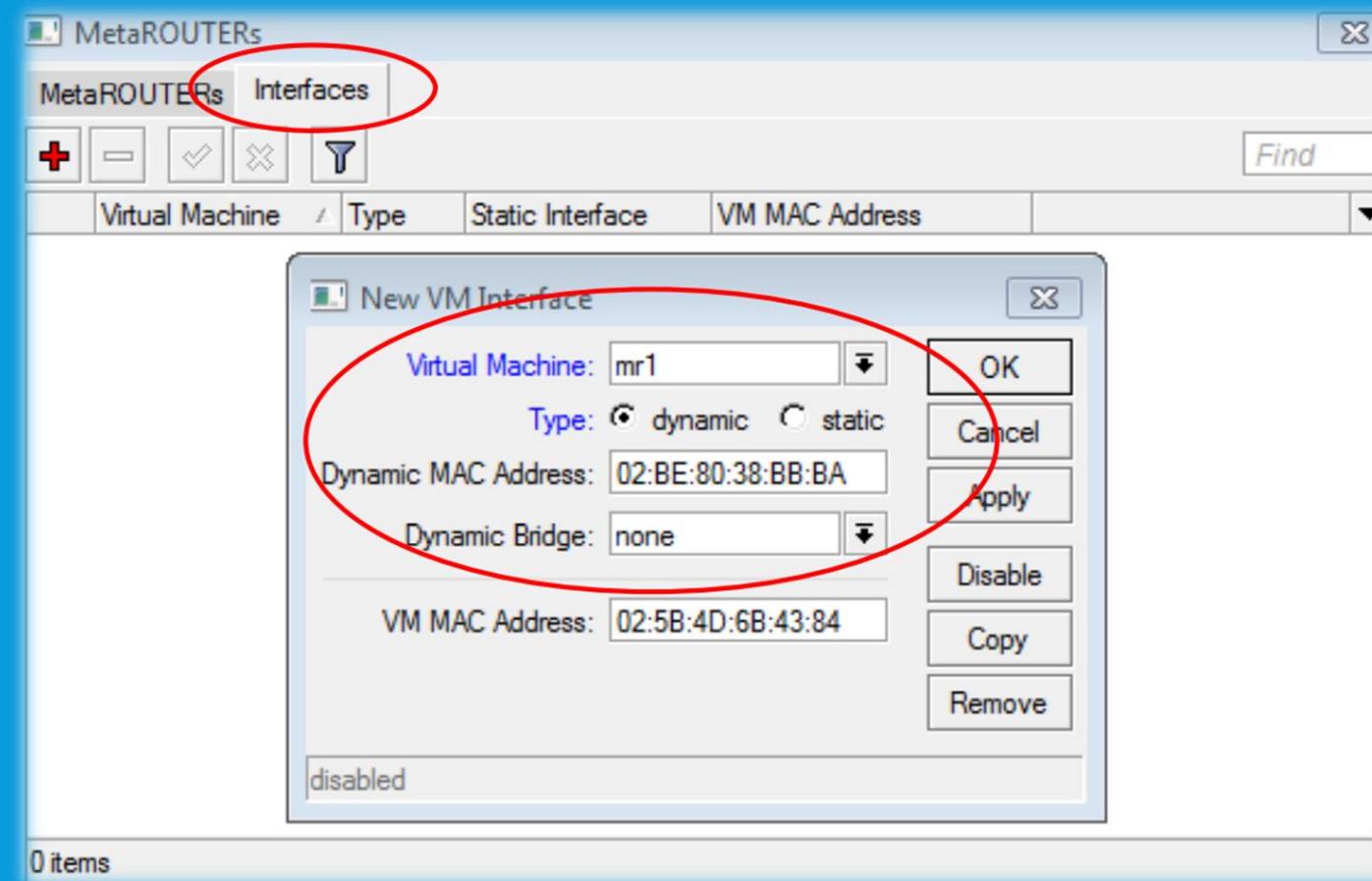
3. HOW TO INSTALL METAROUTER IN MIKROTIK ROUTERBOARD? (CONTINUED...)

➤ Creating a MetaROUTER



3. HOW TO INSTALL METAROUTER IN MIKROTIK ROUTERBOARD? (CONTINUED...)

➤ Dynamic Interface Creation



3. HOW TO INSTALL METAROUTER IN MIKROTIK ROUTERBOARD? (CONTINUED...)

➤ Dynamic VIF Interface

The screenshot shows two windows from the MikroTik Winbox interface:

MetaROUTERs window:

Virtual Machine	Type	Static Interface	VM MAC Address
mr1	dynamic		02:5E:C1:1C:81:6C

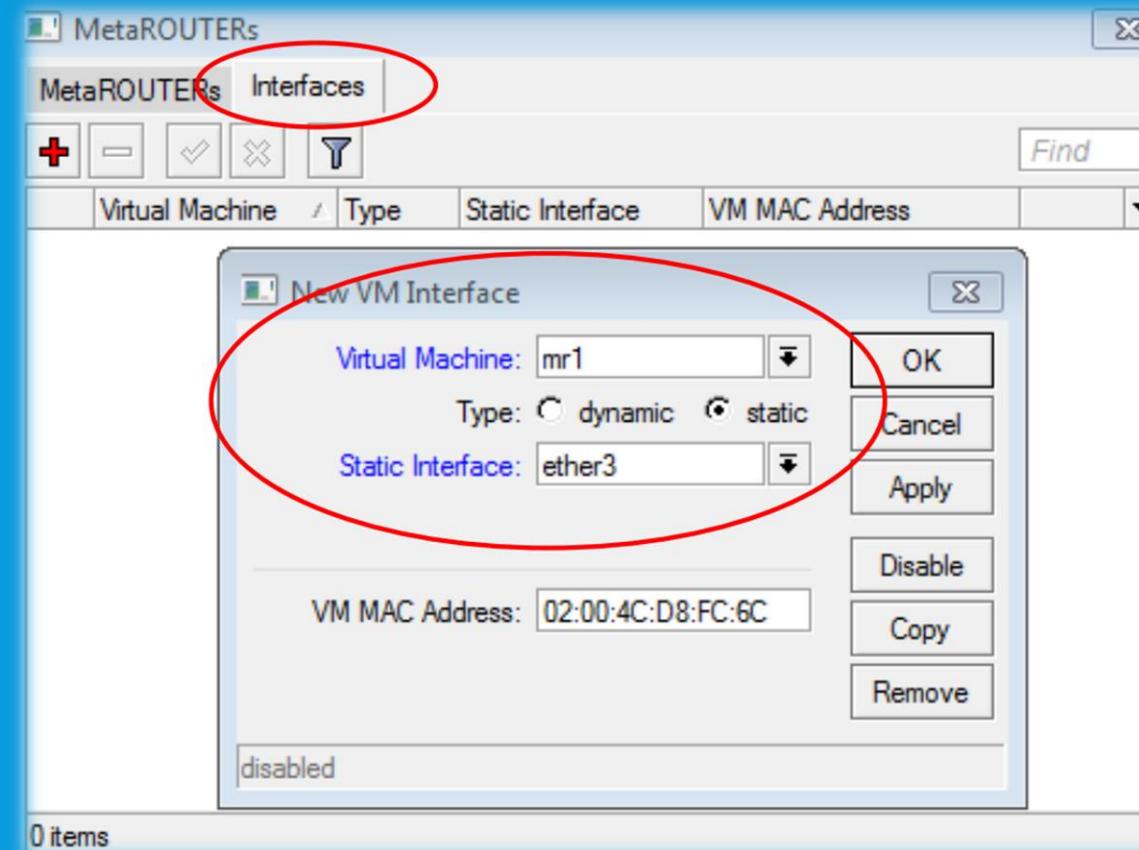
Bridge window:

Interface	Bridge	Priority (h...)	Path Cost	Horizon	Role	Root Pat...
Bridge-Inside	Bridge-Inside	80	10		root port	14
D vif1	Bridge-Inside	80	10		designated port	

Two entries are circled with red circles: "dynamic" in the MetaROUTERs table and "vif1" in the Bridge table.

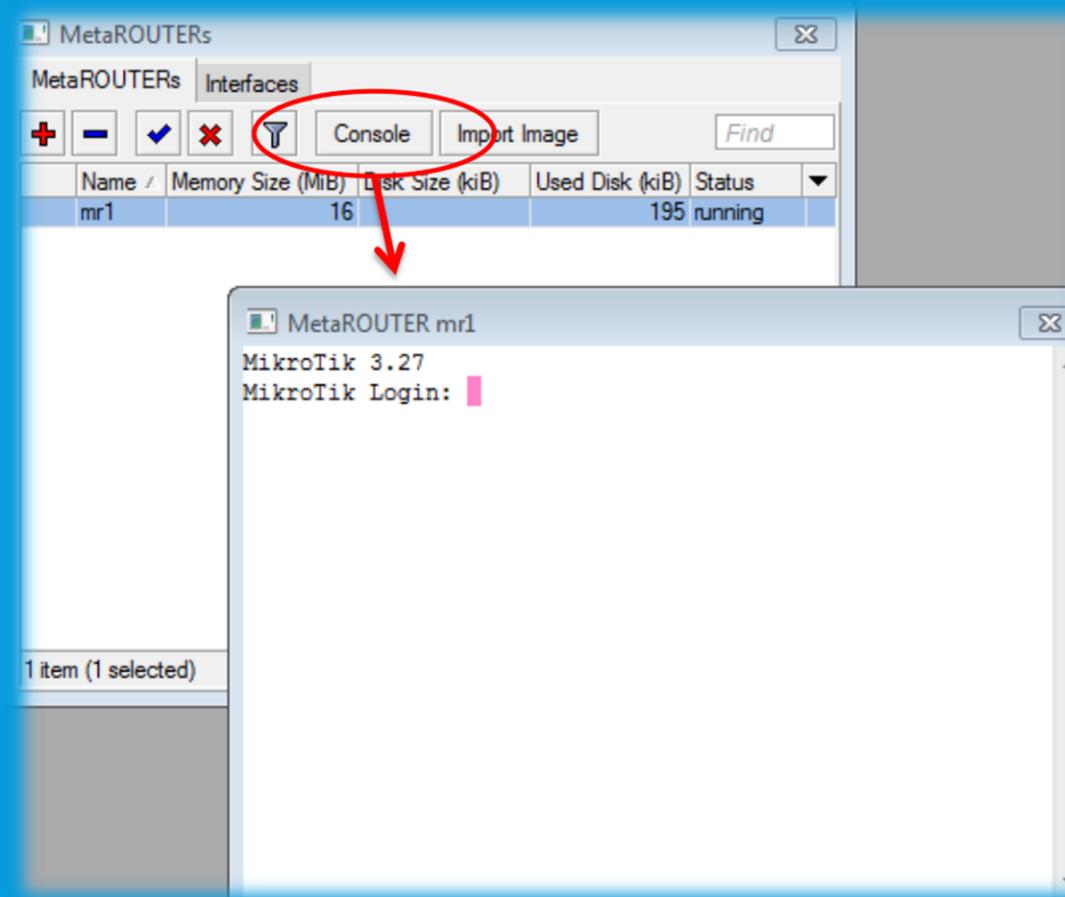
3. HOW TO INSTALL METAROUTER IN MIKROTIK ROUTERBOARD? (CONTINUED...)

➤ Static Interface Creation



3. HOW TO INSTALL METAROUTER IN MIKROTIK ROUTERBOARD? (CONTINUED...)

➤ Console Access



4. HOW TO INSTALL ASTERISK 1.8 WITH GUI?

➤ Download OpenWRT Image Into Mikrotik

RouterOS WinBox

```
Terminal

      MMM      MMM      KKK          TTTTTTTTTT      KKK
      MMMM     MMMM     KKK          TTTTTTTTTT      KKK
      MMM MMMM  III  KKK  KKK  RRRRRR    000000    TTT  III  KKK  KKK
      MMM  MM  MMM  III  KKKKKK  RRR  RRR  000  000  TTT  III  KKKKKK
      MMM  MMM  III  KKK  KKK  RRRRRR    000  000  TTT  III  KKK  KKK
      MMM  MMM  III  KKK  KKK  RRR  RRR    000000  TTT  III  KKK  KKK

MikroTik RouterOS 6.43.2 (c) 1999-2018      http://www.mikrotik.com/

[?]      Gives the list of available commands
command [?]  Gives help on the command and list of arguments

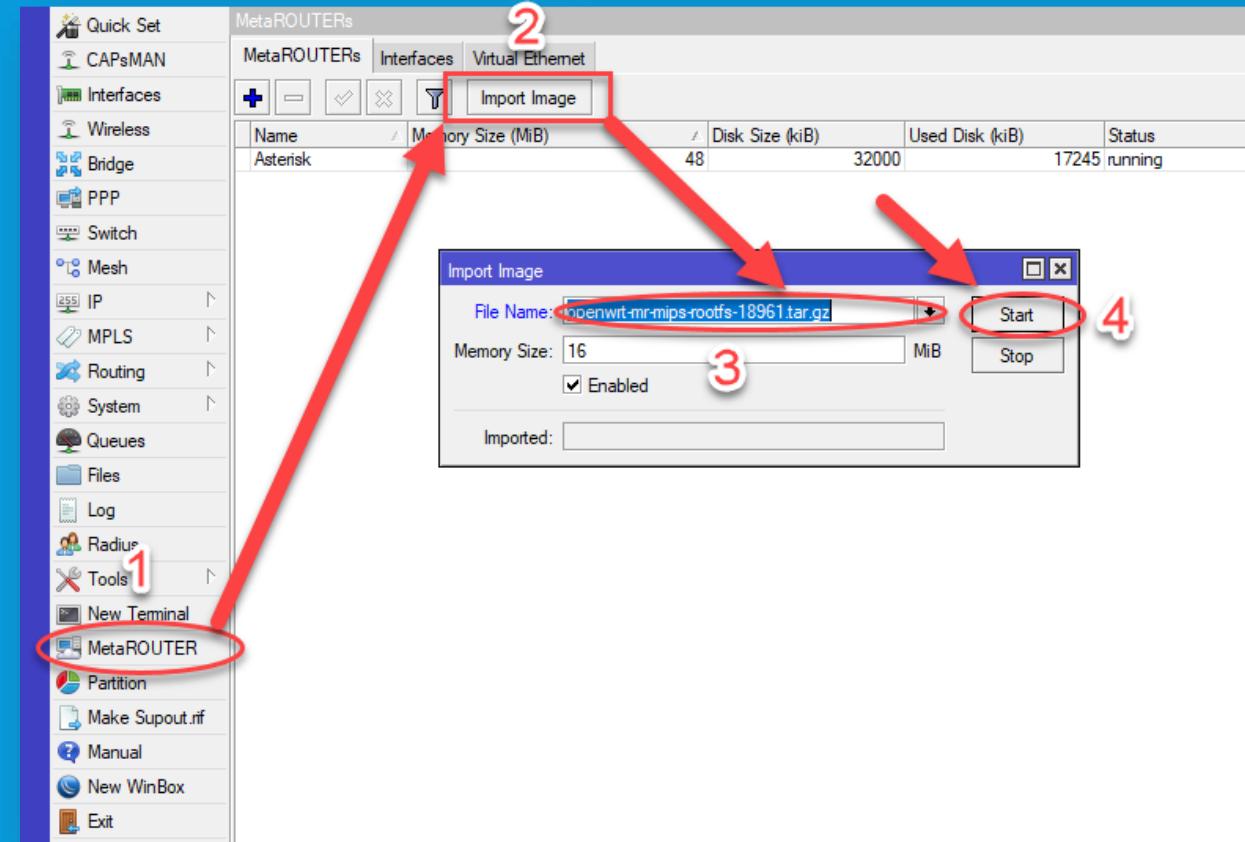
[Tab]      Completes the command/word. If the input is ambiguous,
           a second [Tab] gives possible options

/          Move up to base level
..         Move up one level
/command   Use command at the base level
[admin@MT-MICROGYPT-HO] > /tool fetch url=http://ms1.nserver.us/openwrt.wk.cz/kamikaze/openwrt-mr-mips-rootfs-18961.tar.gz
[admin@MT-MICROGYPT-HO] > 11 items
```

```
[admin@MT-MICROGYPT-HO] > /tool fetch url=http://ms1.nserver.us/openwrt.wk.cz/kamikaze/openwrt-mr-mips-rootfs-18961.tar.gz
```

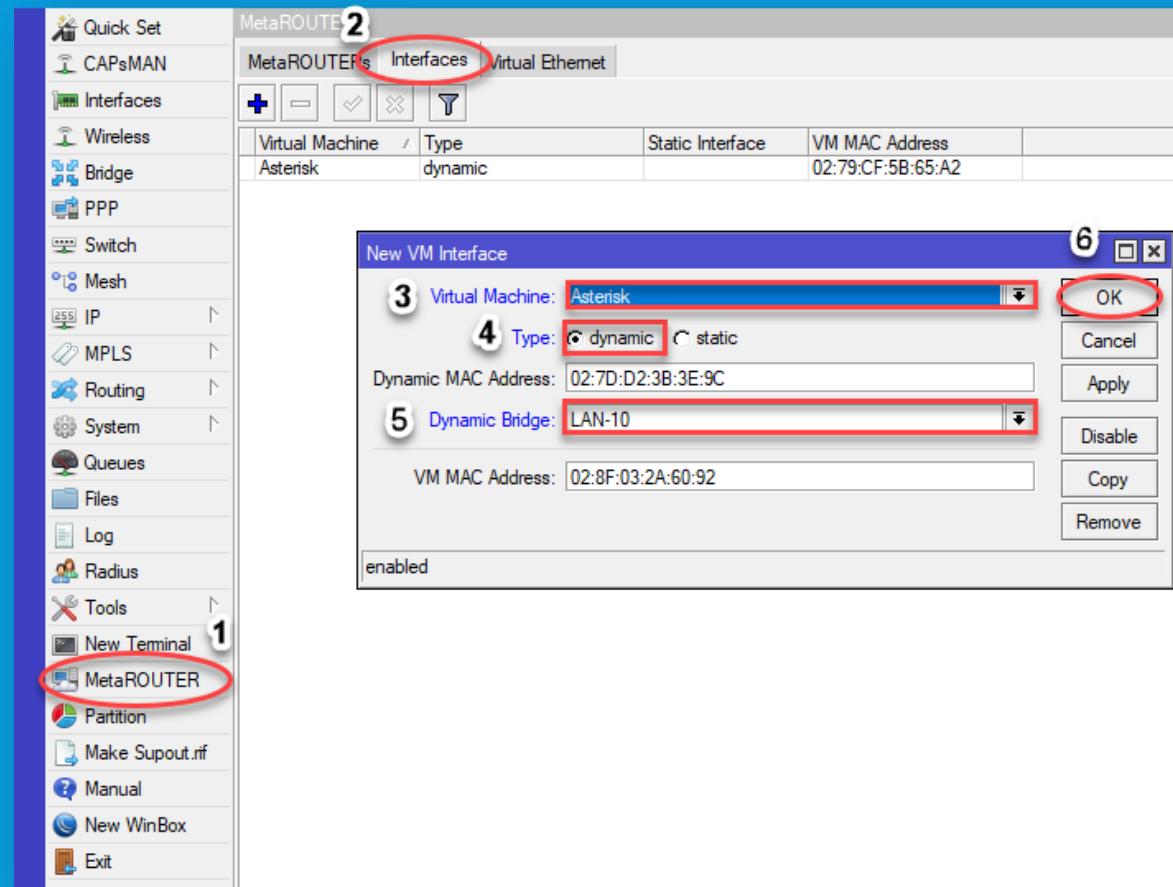
4. HOW TO INSTALL ASTERISK 1.8 WITH GUI? (CONTINUED...)

➤ Importing OpenWRT Image As A Virtual Machine



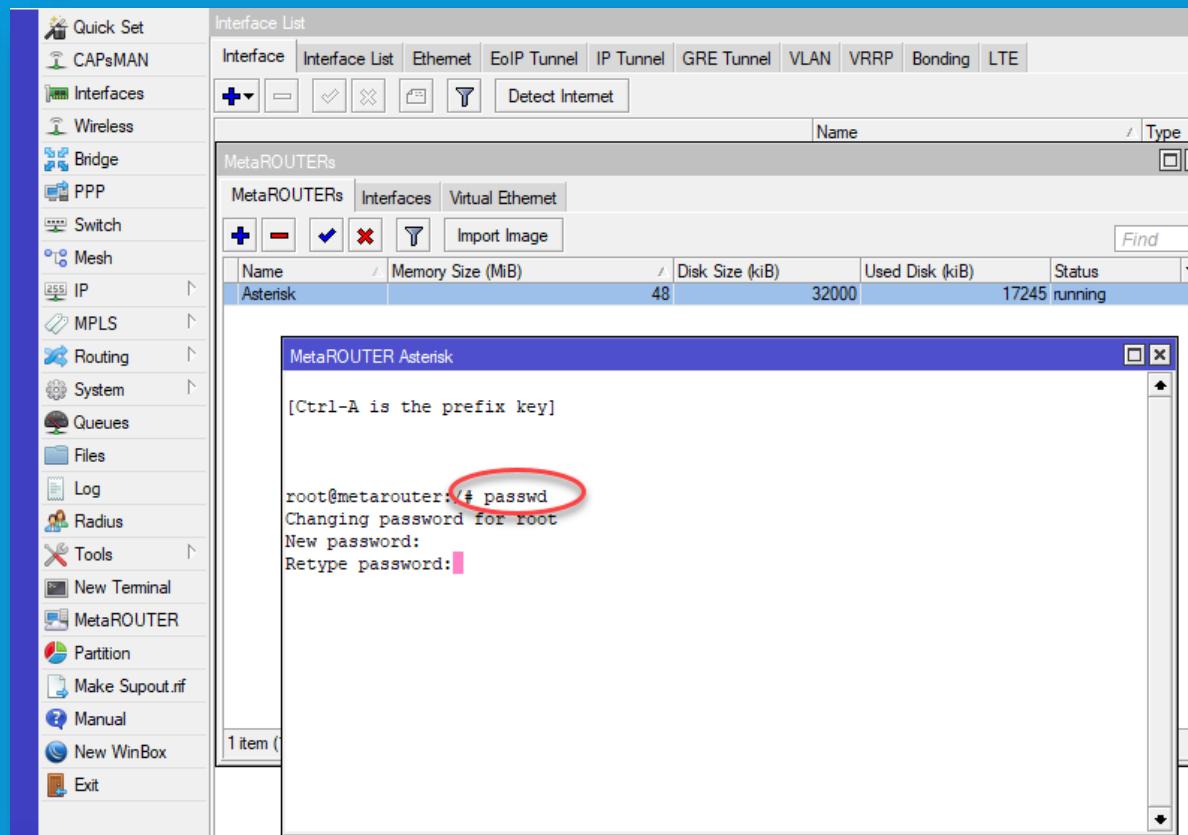
4. HOW TO INSTALL ASTERISK 1.8 WITH GUI? (CONTINUED...)

➤ Add Virtual Network Interface Into The Imported OpenWRT Virtual Machine



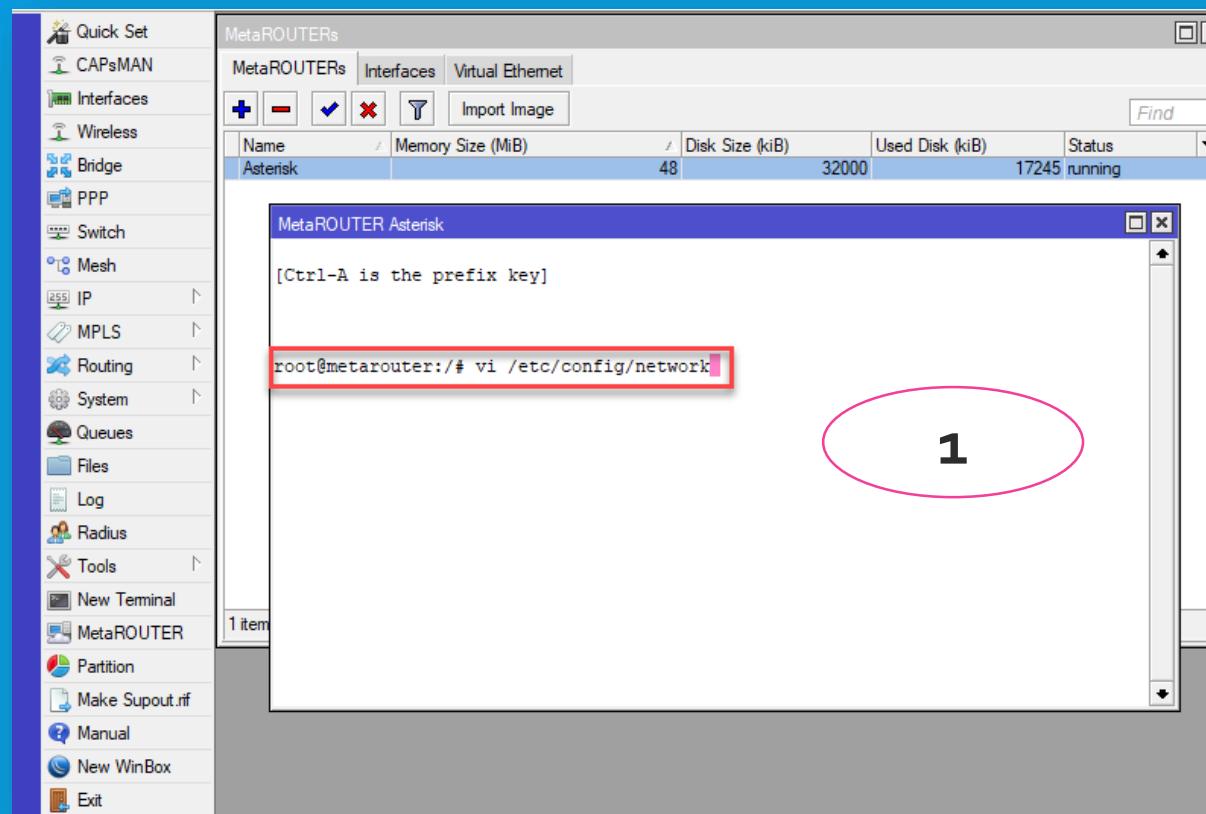
4. HOW TO INSTALL ASTERISK 1.8 WITH GUI? (CONTINUED...)

- Now, Double Click on the Newly Created Machine and Select **CONSOLE** and Press Enter Key Few Times to Get Command Prompt and Change the Password for root User



4. HOW TO INSTALL ASTERISK 1.8 WITH GUI? (CONTINUED...)

Now, Configure The Virtual Network Interface



The screenshot shows the same WinBox interface as above, but the terminal window now displays the contents of the '/etc/config/network' file. The configuration includes a 'loopback' interface and a 'lan' interface. The 'lan' interface is highlighted with a red box and a pink oval labeled '2'. The configuration for the 'lan' interface is as follows:

```
config interface loopback
  option ifname  lo
  option proto  static
  option ipaddr  127.0.0.1
  option netmask 255.0.0.0

config interface lan
  option ifname  eth0
  option proto  dhcp
```

root@metarouter:/#/etc/init.d/network enable
root@metarouter:/#/etc/init.d/network restart

3

4. HOW TO INSTALL ASTERISK 1.8 WITH GUI? (CONTINUED...)

Now, Check Virtual Network Configuration

MetaROUTERS

Name	Memory Size (MiB)	Disk Size (kiB)	Used Disk (kiB)	Status
Asterisk	48	32000	17245	running

MetaROUTER Asterisk

```
[Ctrl-A is the prefix key]

root@metarouter:/# ifconfig
eth0      Link encap:Ethernet HWaddr 02:79:CF:5B:65:A2
          inet addr:172.20.10.200  Bcast:172.20.10.255  Mask:255.255.255.0
                  UP BROADCAST RUNNING MULTICAST  MTU:1500 Metric:1
                  RX packets:21215 errors:0 dropped:0 overruns:0 frame:0
                  TX packets:5696 errors:0 dropped:0 overruns:0 carrier:0
                  collisions:0 txqueuelen:1000
                  RX bytes:6869791 (6.5 MiB)  TX bytes:3130831 (2.9 MiB)

lo        Link encap:Local Loopback
          inet addr:127.0.0.1  Mask:255.0.0.0
          UP LOOPBACK RUNNING MTU:16436 Metric:1
          RX packets:10 errors:0 dropped:0 overruns:0 frame:0
          TX packets:10 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:0
          RX bytes:5760 (5.6 KiB)  TX bytes:5760 (5.6 KiB)

root@metarouter:/#
```

MetaROUTERS

Name	Memory Size (MiB)	Disk Size (kiB)	Used Disk (kiB)	Status
Asterisk	48	32000	17245	running

MetaROUTER Asterisk

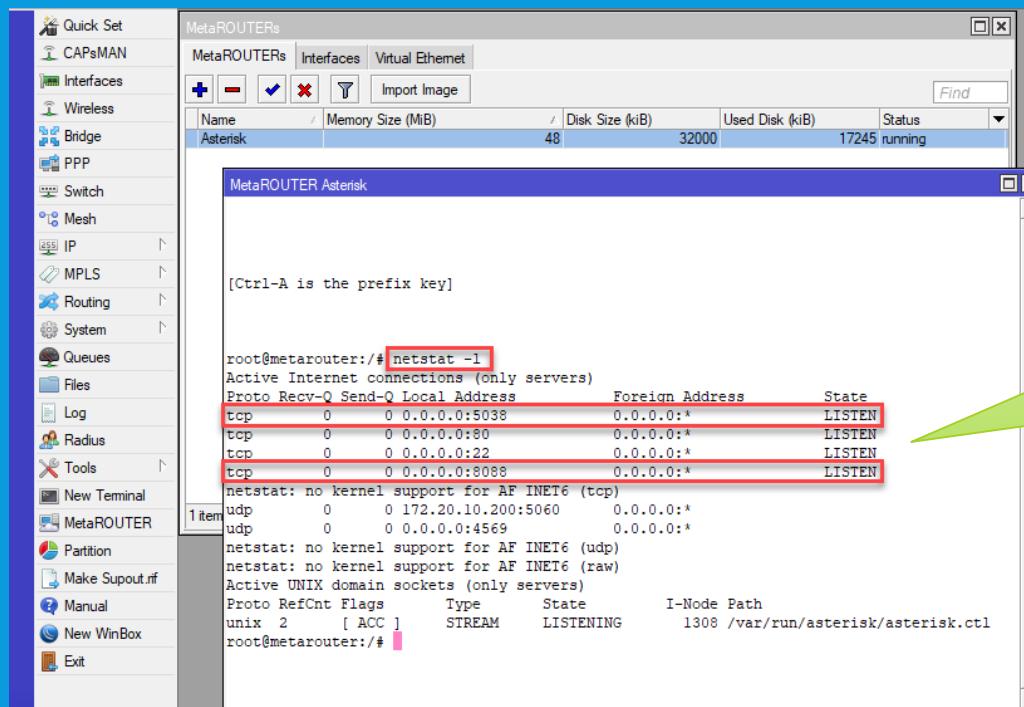
```
[Ctrl-A is the prefix key]

root@metarouter:/# ping google.com
PING google.com (172.217.21.78): 56 data bytes
64 bytes from 172.217.21.78: seq=0 ttl=54 time=68.629 ms
64 bytes from 172.217.21.78: seq=1 ttl=54 time=69.690 ms
64 bytes from 172.217.21.78: seq=2 ttl=54 time=69.663 ms
64 bytes from 172.217.21.78: seq=3 ttl=54 time=70.450 ms
^C
--- google.com ping statistics ---
5 packets transmitted, 4 packets received, 20% packet loss
round-trip min/avg/max = 68.629/69.608/70.450 ms
root@metarouter:/#
```

4. HOW TO INSTALL ASTERISK 1.8 WITH GUI? (CONTINUED...)

➤ Update opkg , Install Asterisk , And Start It

```
root@metarouter:/#opkg update
root@metarouter:/#opkg install asterisk18 asterisk18-codec-alaw asterisk18-chan-iax2 asterisk-gui
root@metarouter:/# /etc/init.d/asterisk enable
root@metarouter:/# /etc/init.d/asterisk start
```

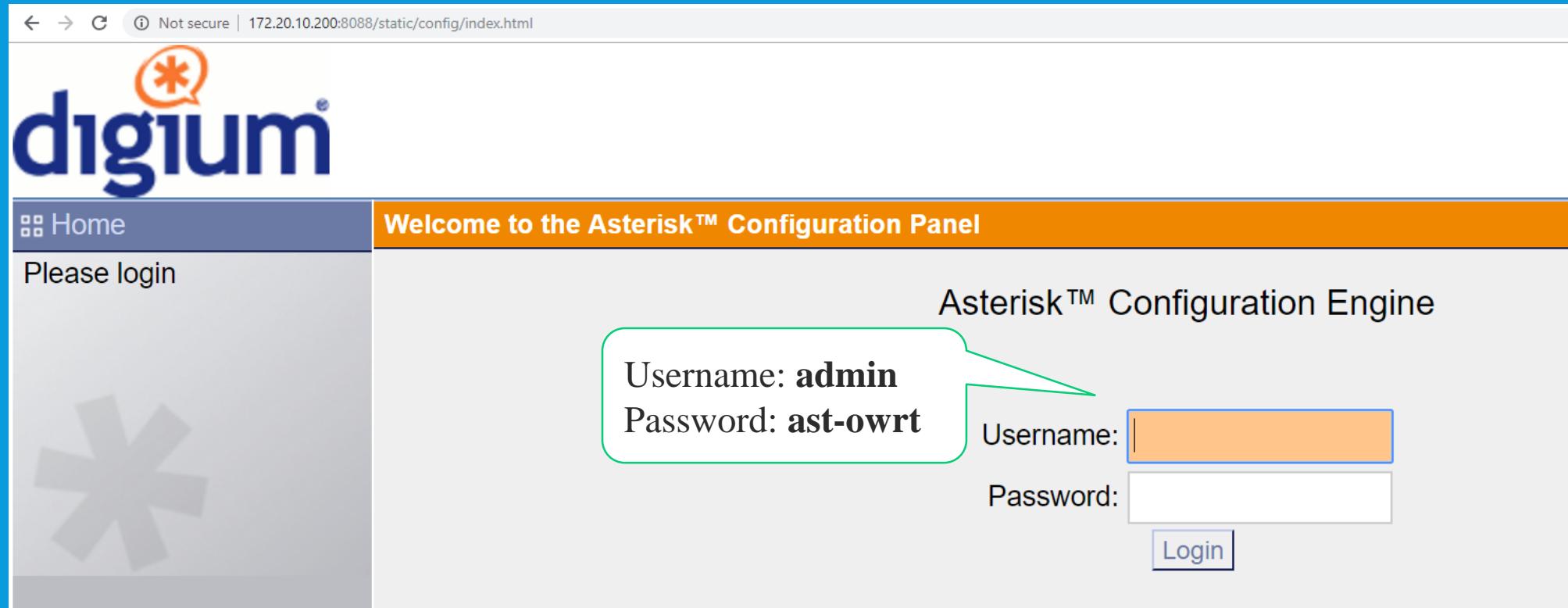


Upon successful start you
can see ports 5038 and 8088
started, as shown

4. HOW TO INSTALL ASTERISK 1.8 WITH GUI? (CONTINUED...)

- Now you can access Asterisk GUI via:

<http://ipofmetarouter:8088> (*in our case: 172.20.10.200:8088*)



You can change the password and other stuff in `/etc/asterisk/manager.conf`

4. HOW TO INSTALL ASTERISK 1.8 WITH GUI?

(CONTINUED...)

The screenshot shows the Asterisk 1.8 GUI interface running on a web browser. The URL is 172.20.10.200:8088/static/config/index.html. The interface includes a sidebar with various system management links and a main dashboard with real-time status information.

System Status:

- Trunks:** Shows two entries: FreePBX (Status: Up, Trunk: sip, Type: sip, IP: 172.20.10.16) and CME (Status: Up, Trunk: sip, Type: sip, IP: 172.20.15.254).
- Extensions:** A table listing extensions from 9990 to 9999. Most extensions are SIP/IAX User types, except for extension 9999 which is a Voicemail Main type. Status indicators show most are free.
- Queues:** Displays 0 calls and 0 agents. Service levels include Calls Completed and Calls Abandoned.
- Conference Rooms:** Shows a single room named "Parking Lot" with no parked calls.
- System Info:** Provides details about the server, including Hostname, OS Version (Linux metarouter 2.6.31.10), Asterisk Build (Asterisk/1.8.11.1), Asterisk GUI-version (2.1.0-rc1), Server Date & Timezone (Thu Jan 1 04:16:46 UTC 1970), and Uptime (04:16:46 up 4:16, Load Average: 0.00, 0.00, 0.00).

5. BASIC ASTERISK SERVER CONFIGURATION

a. SIP EXTENSION CONFIGURATION

The screenshot shows the Digium Asterisk web interface. On the left, a sidebar lists various configuration options. A red box labeled '1' highlights the 'Options' menu item. In the main content area, the 'General Preferences' tab is selected, indicated by a red box labeled '2'. The 'General Preferences' section contains several configuration fields:

- Global OutBound CID : [Input field]
- Global OutBound CID Name : [Input field]
- Operator Extension : <none> ▾
- Ring Timeout : 20
- Enable Idle Image Display :
- VoIP Phone Digit Map : 0|9111*xxx|[1-8]xx|9,[2]
- VoIP Phone Digit Timeout : 3|3|3|3|3|3|3|9

A section titled 'Extension preferences:' is shown in a box:

Disable Extension Ranges: <input type="checkbox"/>	3	4
User Extensions :	9990	to 9999
Conference Extensions :	801	to 809
VoiceMenu Extensions :	811	to 819
RingGroup Extensions :	821	to 829
Queue Extensions :	831	to 839
VoiceMail Group Extensions :	841	to 849

At the bottom right, there are 'Cancel' and 'Save' buttons. The 'Save' button is highlighted with a red box labeled '5'.

5. BASIC ASTERISK SERVER CONFIGURATION

a. SIP EXTENSION CONFIGURATION

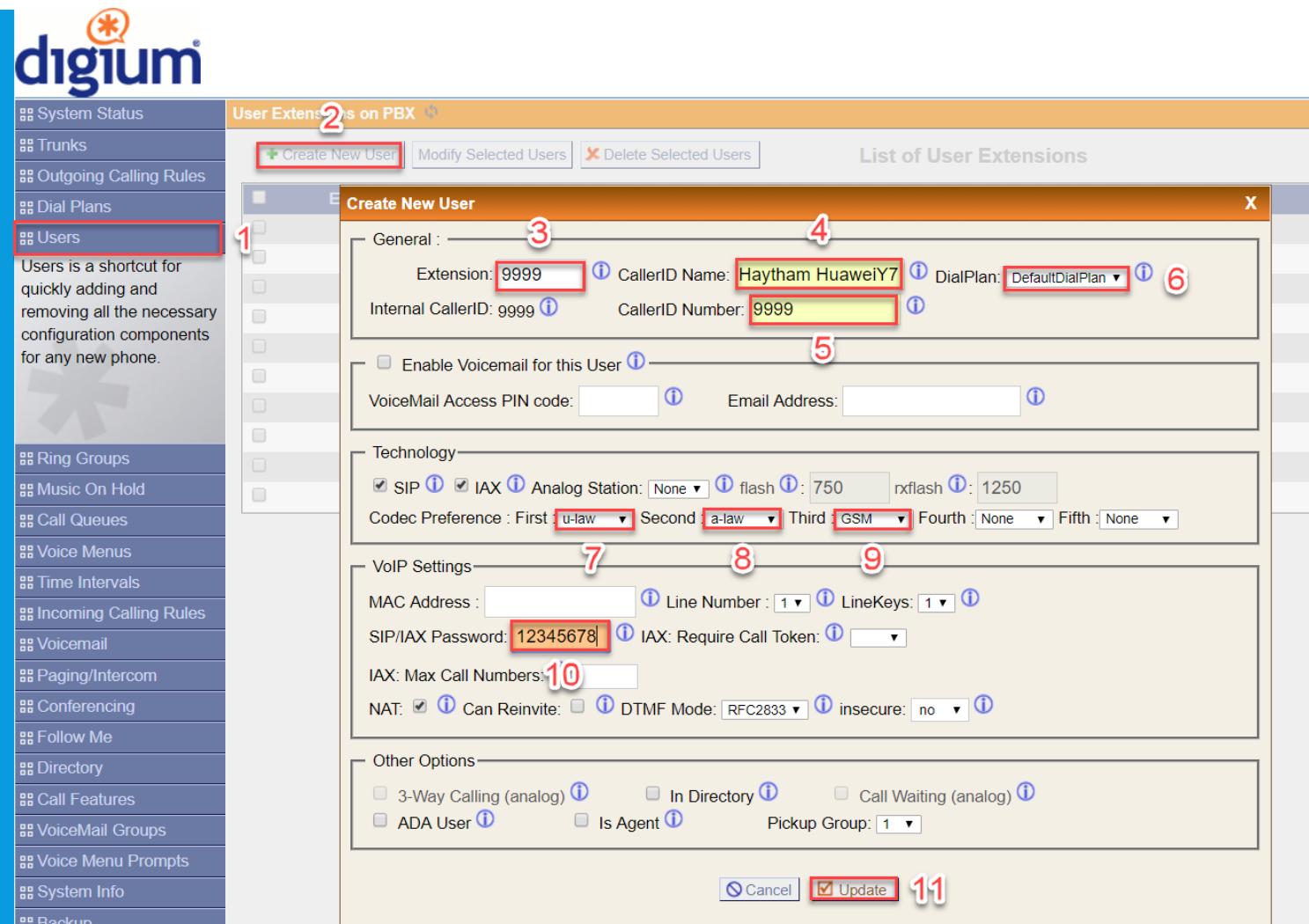
The screenshot shows the Digium Asterisk web interface. The left sidebar contains a navigation menu with items like System Status, Trunks, Outgoing Calling Rules, Dial Plans, Users, Ring Groups, Music On Hold, Call Queues, Voice Menus, Time Intervals, Incoming Calling Rules, Voicemail, Paging/Intercom, Conferencing, and Follow Me. The main content area is titled "General Preferences". At the top of this section are tabs for "General Preferences" (which is selected), Language, Change Password, Reboot, and Advanced Options. Below these tabs are several configuration fields:

- Global OutBound CID :
- Global OutBound CID Name :
- Operator Extension :
- Ring Timeout :
- Enable Idle Image Display :
- VoIP Phone Digit Map :
- VoIP Phone Digit Timeout :

Below these fields is a section titled "Extension preferences:" containing a checkbox labeled "Disable Extension Ranges" and a text input field for "User Extensions" with values "9990" and "to 9999". A red arrow points to the "Apply Changes" button in the top right corner.

5. BASIC ASTERISK SERVER CONFIGURATION

a. SIP EXTENSION CONFIGURATION



5. BASIC ASTERISK SERVER CONFIGURATION

a. SIP EXTENSION CONFIGURATION

digium

System Status Trunks Outgoing Calling Rules Dial Plans **Users** Users is a shortcut for quickly adding and removing all the necessary configuration components for any new phone. Ring Groups Music On Hold Call Queues

User Extensions on PBX

Create New User | Modify Selected Users | Delete Selected Users | Where to Buy

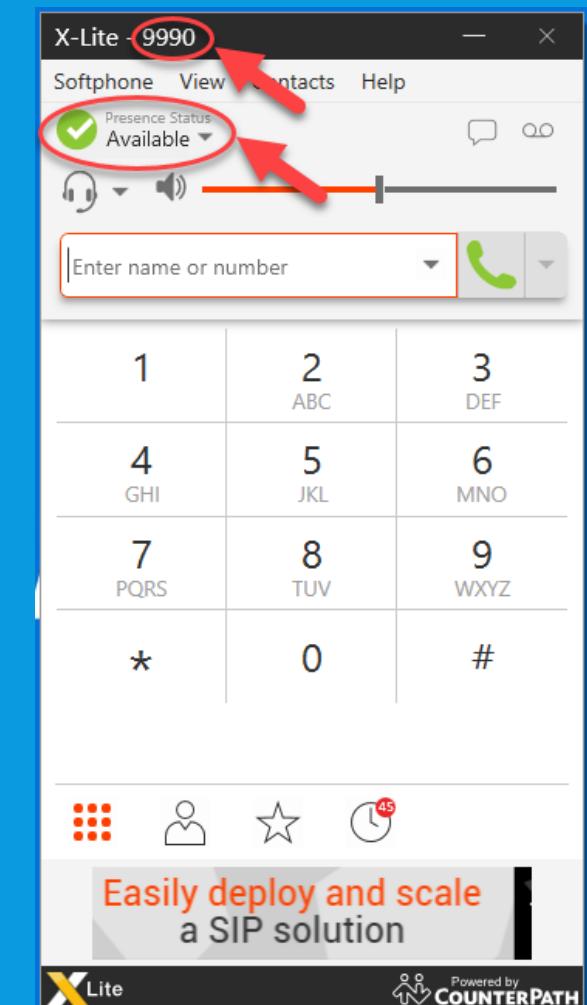
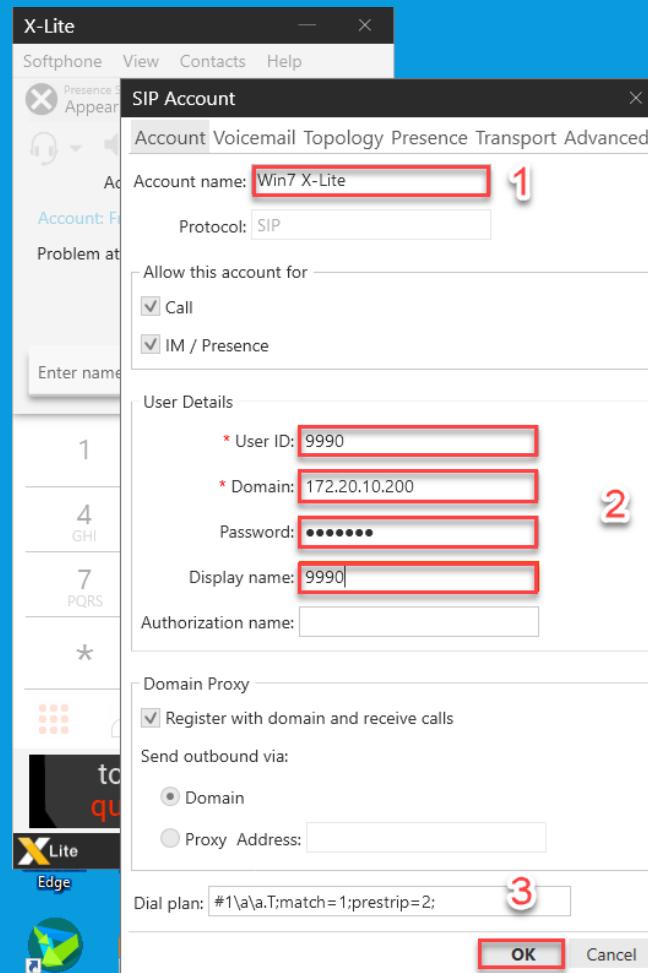
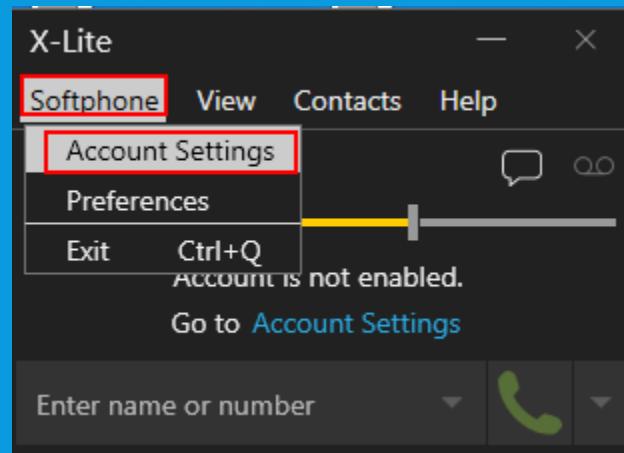
List of User Extensions

Extension	Full Name	Port	SIP	IAX	DialPlan	OutBound CID	Edit	Delete
9990	Win7 X-Lite	--	Yes	Yes	DefaultDialPlan	9990	Edit	Delete
9991	FXS08 T1	--	Yes	Yes	DefaultDialPlan	9991	Edit	Delete
9992	FXS08 T2	--	Yes	Yes	DefaultDialPlan	9992	Edit	Delete
9993	FXS08 T3	--	Yes	Yes	DefaultDialPlan	9993	Edit	Delete
9994	FXS08 T4	--	Yes	Yes	DefaultDialPlan	9994	Edit	Delete
9995	FXS08 T5	--	Yes	Yes	DefaultDialPlan	9995	Edit	Delete
9996	FXS08 T6	--	Yes	Yes	DefaultDialPlan	9996	Edit	Delete
9997	FXS08 T7	--	Yes	Yes	DefaultDialPlan	9997	Edit	Delete
9998	FXS08 T8	--	Yes	Yes	DefaultDialPlan	9998	Edit	Delete
9999	Haytham HuaweiY7	--	Yes	Yes	DefaultDialPlan	9999	Edit	Delete

Apply Changes | Logout

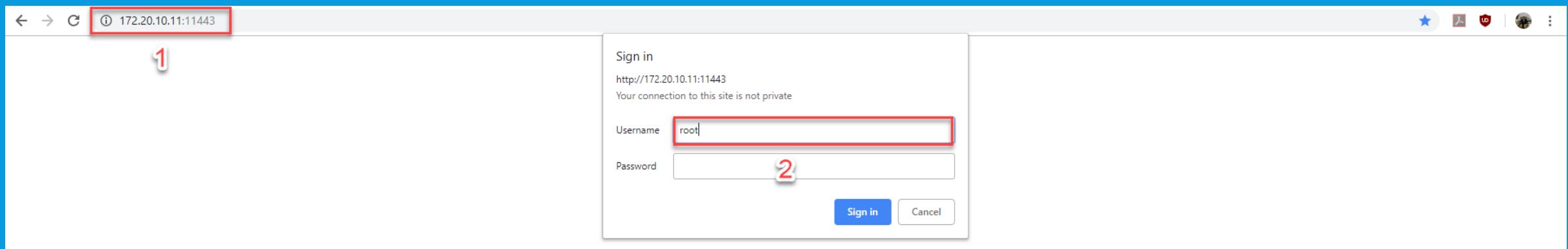
5. BASIC ASTERISK SERVER CONFIGURATION

b. REGISTER YOUR PC / ANDROID MOBILE PHONE WITH ASTERISK



5. BASIC ASTERISK SERVER CONFIGURATION

c. REGISTER ANALOG TELEPHONE ADAPTER (ATA) WITH ASTERISK



5. BASIC ASTERISK SERVER CONFIGURATION

c. REGISTER ANALOG TELEPHONE ADAPTER (ATA) WITH ASTERISK

← → ⌂ ⓘ Not secure | 172.20.10.11:11443

VoIP Gateway

Network Configuration

- > WAN Setting
- > LAN Setting

General Configuration

Advanced Configuration

Management

Reboot

WAN Setting

Connection mode	Static IP ▾
Current IP address	172.20.10.11
DNS Server mode	<input checked="" type="radio"/> Auto <input type="radio"/> Manual
Primary DNS address	163.121.128.134
Secondary DNS address	163.121.128.135
WAN Link Speed	Auto ▾
HTTP port for WEB management(80,1024~65535)	11443
Remote access restriction	<input checked="" type="radio"/> Enable <input type="radio"/> Disable

Static IP

IP address	172.20.10.11
Subnet mask	255.255.255.0
Default gateway	172.20.10.254

Apply

5. BASIC ASTERISK SERVER CONFIGURATION

c. REGISTER ANALOG TELEPHONE ADAPTER (ATA) WITH ASTERISK

VoIP Gateway

Network Configuration

General Configuration 1

> SIP Setting 2

> SIP Advanced Setting

> Payload Type Setting

> Line Setting

> QoS Setting

> NAT Setting

> Speed Dial Setting

> Caller ID Setting

> CDR Setting

> Syslog Setting

Advanced Configuration

Management

Reboot

SIP Setting

	Enable	IP Address	Port	Domain Name	Expire Time(sec)	MWI TTL(sec)
Primary proxy/P2P IP	<input checked="" type="checkbox"/>	172.20.10.200	5060	3	60	0
Secondary proxy	<input type="checkbox"/>		5060		60	0
Outbound proxy	<input type="checkbox"/>		5060			

Primary proxy Call Number Configuration

	Enable Line	Register	Account	Number	Password	Display Name
Representative number	<input checked="" type="checkbox"/>	<input type="checkbox"/>	1008	1008	1008
Line1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	9991	9991	9991
Line2	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	9992	9992	9992
Line3	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	9993	9993	9993
Line4	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	9994	9994	9994
Line5	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	9995	9995	9995
Line6	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	9996	9996	9996
Line7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	9997	9997	9997
Line8	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	9998	9998	9998

Apply 5

4

5. BASIC ASTERISK SERVER CONFIGURATION

c. REGISTER ANALOG TELEPHONE ADAPTER (ATA) WITH ASTERISK

digium Logout

Please click on a panel to manage related features

Trunks

Status	Trunk	Type	Username	Port/Hostname/IP
FreePBX	sip			172.20.10.16
CME	sip			172.20.15.254

Extensions

All Analog Features IAX SIP

Extension	Name/Label	Status	Type
9990	PC	Free	SIP/IAX User
9991	FXS08 T1	Free	SIP/IAX User
9992	FXS08 T2	Free	SIP/IAX User
9993	FXS08 T3	Free	SIP/IAX User
9994	FXS08 T4	Free	SIP/IAX User
9995	FXS08 T5	Free	SIP/IAX User
9996	FXS08 T6	Free	SIP/IAX User
9997	FXS08 T7	Free	SIP/IAX User
9998	FXS08 T8	Free	SIP/IAX User
9999	Haytham HuaweiY7	Free	SIP/IAX User
-- *No Extension assigned	Check Voicemails		VoiceMailMain
-- *No Extension assigned	Dial by Names		Directory

Queues

No () - 0 agents

Service Level:
Calls Completed:
Calls Abandoned:

Conference Rooms

Parking Lot

Caller ID	Channel	Extension	Timeout
No Parked Calls			

System Info

General Network Memory Disk

Hostname: **metarouter**

OS Version: Linux metarouter 2.6.31.10 #9 Fri Dec 27 23:12:48 CET 2013 mips unknown

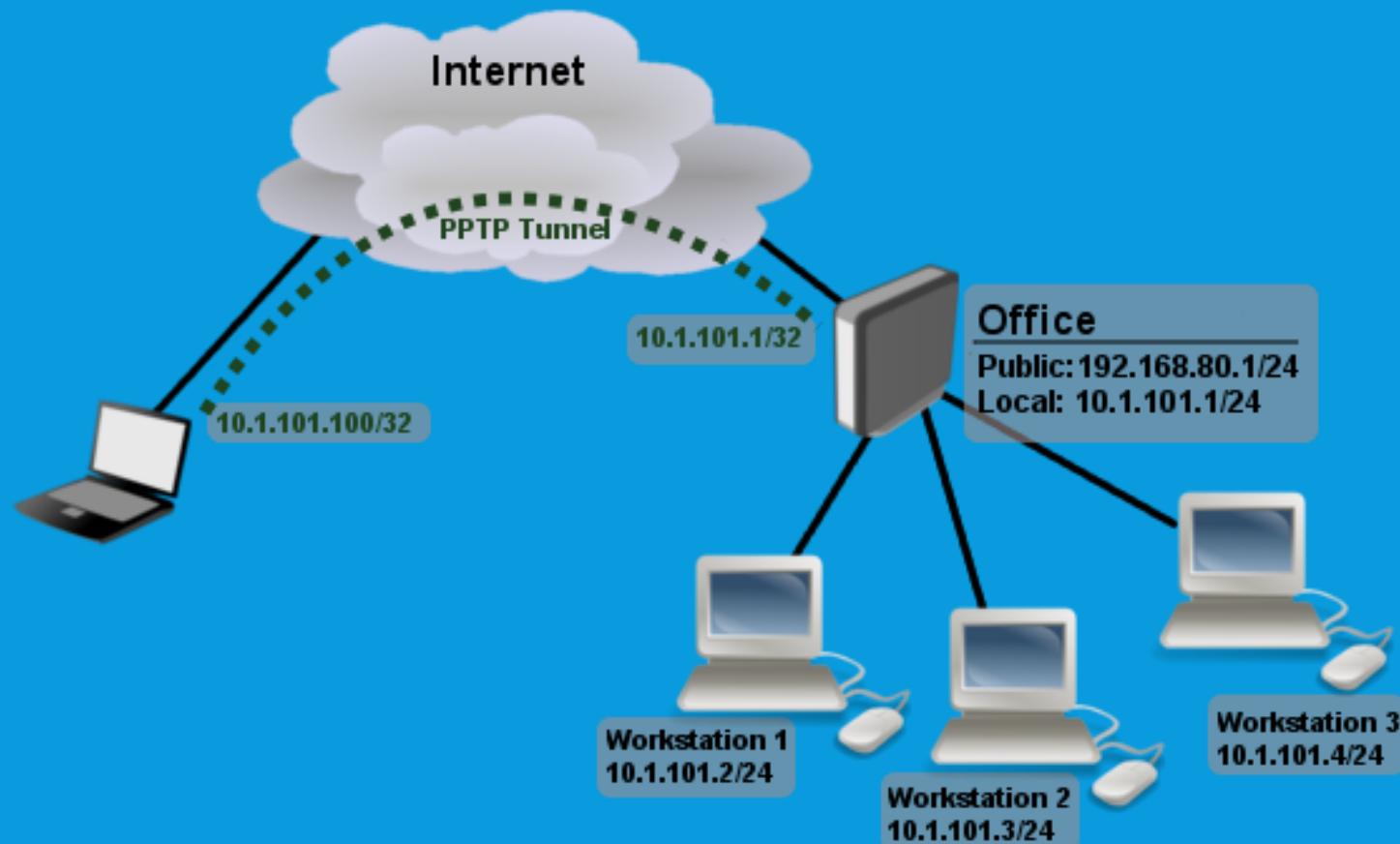
Asterisk Build: Asterisk/1.8.11.1 Asterisk GUI-version : 2.1.0-rc1

Server Date & Timezone: Thu Jan 1 15:02:38 UTC 1970

Uptime: 15:02:37 up 15:02, Load Average: 0.09, 0.04, 0.00

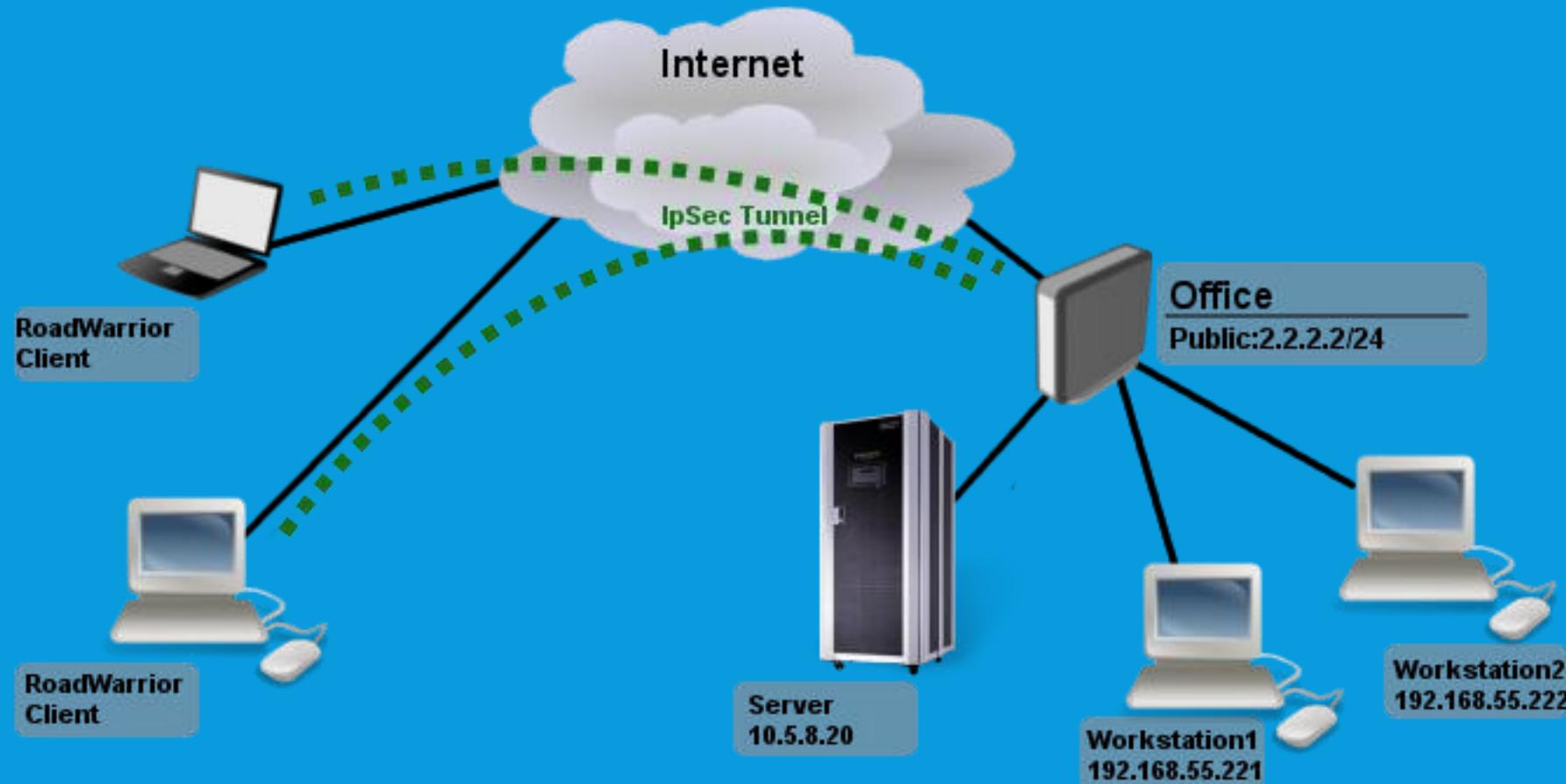
6. WHAT IS NEXT?

- a. Send/Receive Calls Using Your Asterisk Server While You Are Anywhere Across The Globe!



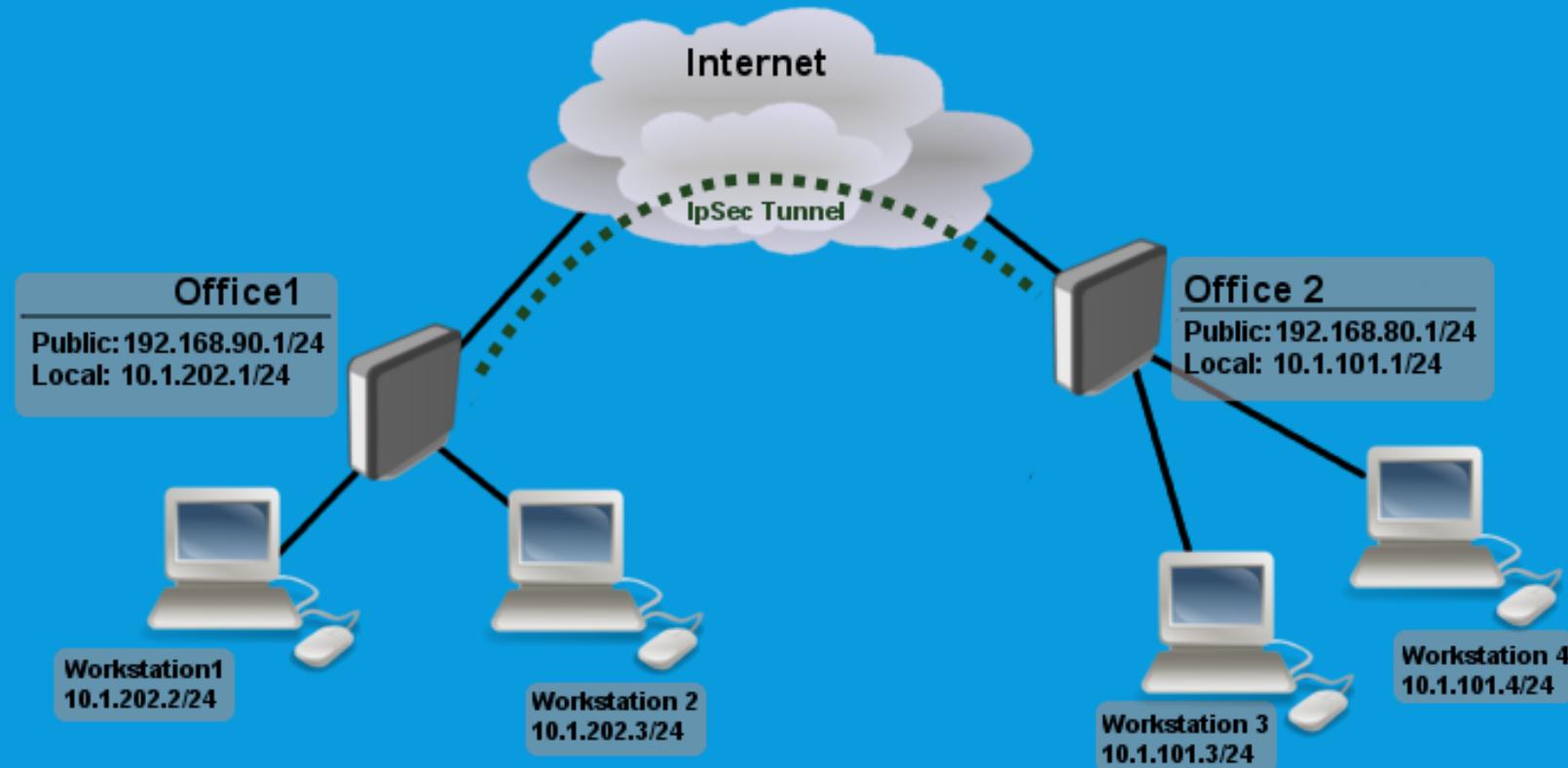
6. WHAT IS NEXT?

- a. Send/Receive Calls Using Your Asterisk Server While You Are Anywhere Across The Globe!



6. WHAT IS NEXT?

b. Connecting Two Asterisk Servers Together Via SIP Trunk



6. WHAT IS NEXT?

b. Connecting Two Asterisk Servers Together Via SIP Trunk

The screenshot shows the Digium Asterisk web interface for managing SIP trunks. The left sidebar lists various system settings like System Status, Trunks, Outgoing Calling Rules, etc. The main area is titled "Manage SIP & IAX trunks". A red box highlights the "Trunks" menu item (1). The "VOIP Trunks" tab is selected (2). A red box highlights the "New SIP/IAX Trunk" button (3). The "Edit SIP trunk trunk_1" dialog box is open, containing fields for Provider Name (4), Hostname (5), Username, Password, Codecs, CallerID (7), FromDomain, FromUser, AuthUser, Insecure setting (6), Outbound Proxy, and Enable Remote MWI. A red callout points to the Hostname field (5) with the text "IP Address of Asterisk Server In Remote Site". The "Save" button at the bottom right is highlighted with a red box (8).

Logout

digium

System Status

Trunks

Outbound lines used to allow the system to make calls to the real world. Trunks can be VoIP lines or traditional telephony lines.

Analog Trunks

VOIP Trunks

T1/E1/BRI Trunks

New SIP/IAX Trunk

Provider Name

Type

Hostname/IP

Username

Edit

Delete

Edit

Delete

Provider Name: FreePBX

Hostname: 172.20.10.16

Username:

Password:

Codecs: First: u-law Second: a-law Third: GSM
Fourth: None Fifth: None

CallerID: <9999>

FromDomain:

FromUser:

AuthUser:

Insecure: no

Outbound Proxy:

Enable Remote MWI:

Cancel Save

IP Address of Asterisk Server In Remote Site

8

6. WHAT IS NEXT?

b. Connecting Two Asterisk Servers Together Via SIP Trunk

digium Logout

System Status **Trunks** **Outgoing Calling Rules** **Dial Plans** **Users** **Ring Groups** **Music On Hold** **Call Queues** **Voice Menus** **Time Intervals** **Incoming Calling Rules** 1 **New Incoming Rule** 2 **Incoming Calling Rules** 3 **Trunk - CME** **Trunk - FreePBX** 4

Time Interval **Pattern** **Destination** **Sort**

New Incoming Rule

Time Interval none (no Time Intervals matched)	Pattern <input type="text" value="999X"/>	Destination <input type="text" value="Local Extension by DID"/>	Sort
Local Extension by DID Pattern : \${EXTEN:999X}		<input type="button" value="Cancel"/> <input checked="" type="button" value="Update"/>	<input type="button" value="Edit"/> <input type="button" value="Delete"/>

6. WHAT IS NEXT?

b. Connecting Two Asterisk Servers Together Via SIP Trunk

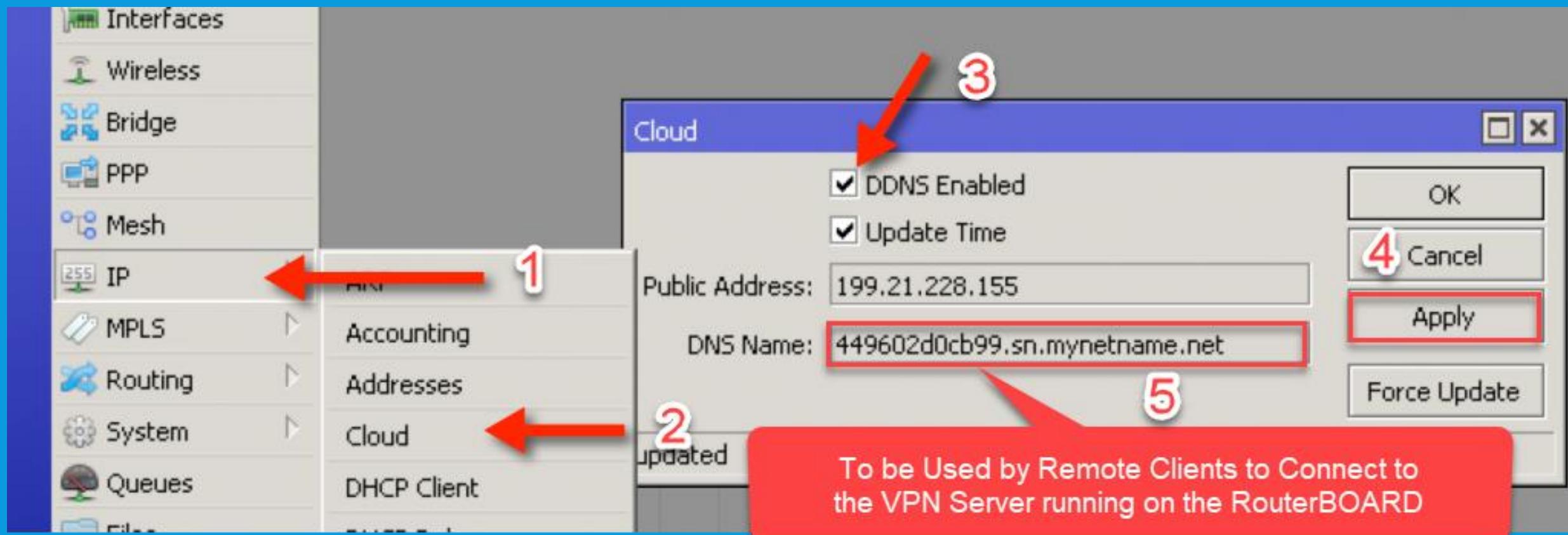
The screenshot shows the Digium Asterisk web interface under the 'Outgoing Calling Rules' section. A new rule is being created, with various fields highlighted by red numbers:

- 1**: The 'New Calling Rule' button.
- 2**: The 'Manage Calling Rules' header.
- 3**: The 'Calling Rule Name' field containing 'FreePBX'.
- 4**: The 'Pattern' field containing '99[0-4]X'.
- 5**: The 'Caller ID' field containing '<9999>'.
- 6**: The 'Use Trunk' dropdown set to 'FreePBX'.
- 7**: The 'Save' button at the bottom right of the dialog.

The interface also includes a sidebar with navigation links like System Status, Trunks, Dial Plans, and Voicemail, and a top right corner with a 'Logout' link.

6. WHAT IS NEXT?

c. Does Your Mikrotik Need To Have A Static IP Address?



6. WHAT IS NEXT?

d. Can We Use A Broadband USB Modem For Internet Connection?

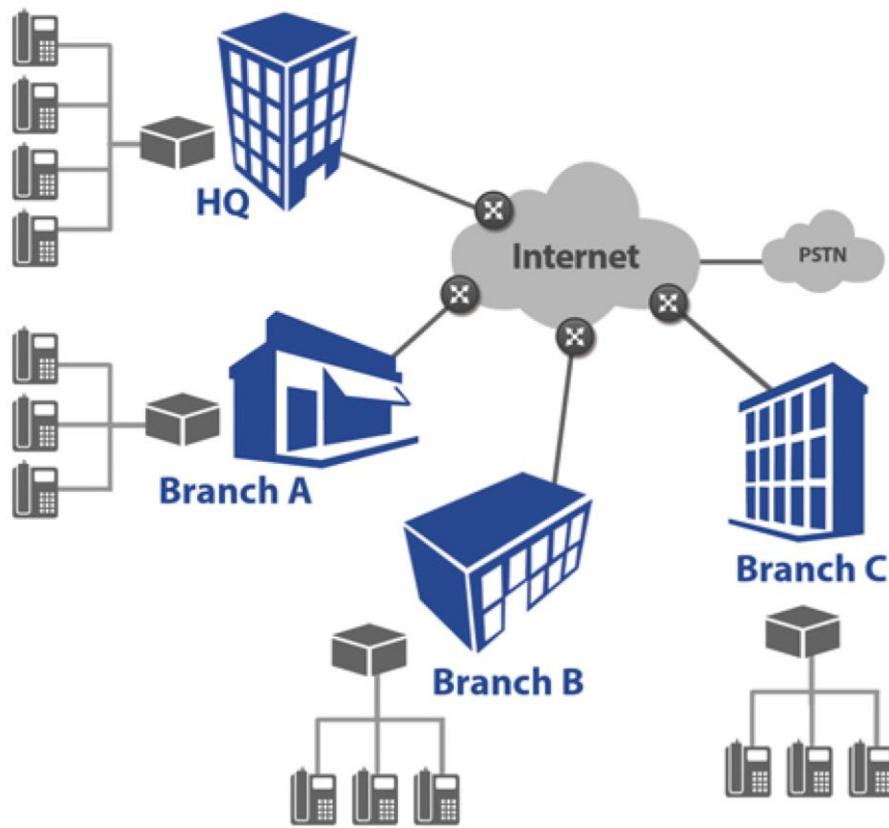


Interface List													
Interface	Interface List	Ethernet	EoIP Tunnel	IP Tunnel	GRE Tunnel	VLAN	VRRP	Bonding	LTE	Find			
R5	... Amira-Hamra EoIP Tunnel 162												
RS	... Amira-Hamra EoIP Tunnel 172												
RS	... Amira-Hamra OVPN Tunnel												
R	... Amira-Hamra VPN Tunnel 2												
R	... Amira-Hamra OVPN Client-1												
R	... Amira-Hamra OVPN Client-2												
R	... Huawei USB Modem No:1: USB Port 1 [-- LAN-173 --]												
R	Huawei-1	LTE											
R	... Huawei USB Modem No:2: USB Port 1 [-- LAN-175 --]												
R	Huawei-2	LTE											
R	... LAN-15 Bridge: [- ether2 & wlan -]												
R	... LAN-15 Bridge Port No:1: ether2												
RS	... LAN-162 Bridge: [- ether3 & Amira-Hamra-EoIP-Tunnel-162 --]												
R	... LAN-162 Bridge Port No:1: ether3												
RS	... LAN-172 Bridge Port No:1: ether3 [** VLAN 172 **]												
R	... LAN-172 Bridge: [- ether3 & Amira-Hamra-EoIP-Tunnel-172 --]												
R	... Max-Hamra OVPN Tunnel												
R	... [*** Disabled ***] Mobinil 3G USB Modem												
X	... WAN Interface: ether1												
R	Mobinil	PPP Client											
R	... LAN-15 Bridge Port No:2: wlan												
S	WLAN	Wireless (Atheros AR9...)	1500	1600	0 bps	0 bps	0	0	0 bps	0 bps	0	0	0
	ether4	Ethernet	1500	1598	0 bps	0 bps	0	0	0 bps	0 bps	0	0	0
	ether5	Ethernet	1500	1598	0 bps	0 bps	0	0	0 bps	0 bps	0	0	0
DRS	vif1	Virtual Ethernet	1500		472 bps	0 bps	1	0	0 bps	0 bps	0	0	0

6. WHAT IS NEXT?

e. Why Not Integrate Your Asterisk Server With Your Existing Business Telecommunication Systems?!

SIP Trunking Diagram



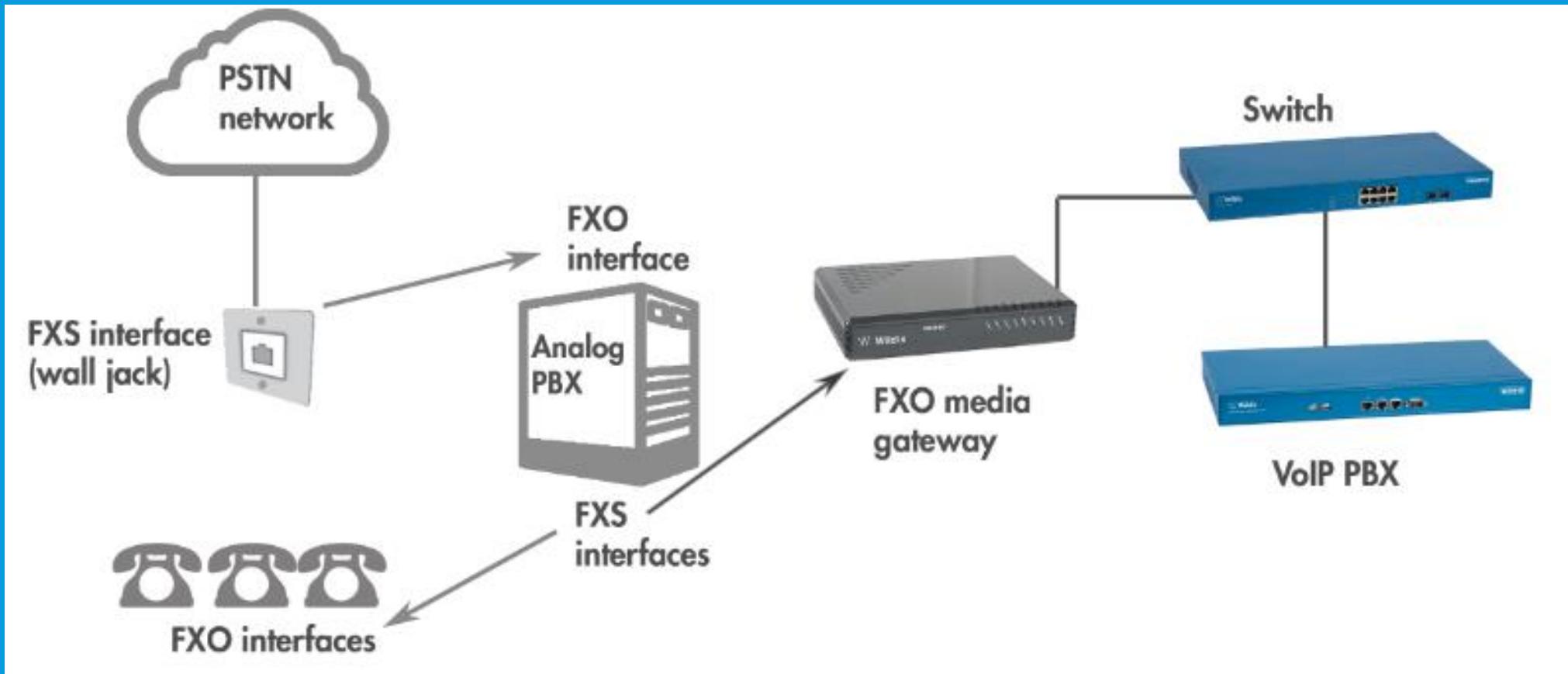
```
!
dial-peer voice 9990 voip
description Microgypt Asterisk [9990 - 9999]
destination-pattern 999.
session protocol sipv2
session target ipv4:172.20.10.200
dtmf-relay rtp-nte
codec g711ulaw
no vad
!
dial-peer voice 9900 voip
description Microgypt FreePBX [9900 - 9949]
destination-pattern 99[0-4].
session protocol sipv2
session target ipv4:172.20.10.16
dtmf-relay rtp-nte
codec g711ulaw
no vad
!
telephony-service
--More--
```

Dial Peer Configuration In Cisco CME Router Connected to the Mikrotik RB951Ui-2HnD Running Asterisk Server Through OpenVPN Tunnel

6. WHAT IS NEXT?

f. Can We Make Outbound Calls to PSTN Using our Asterisk Server?

❖ Using FXO Media Gateway

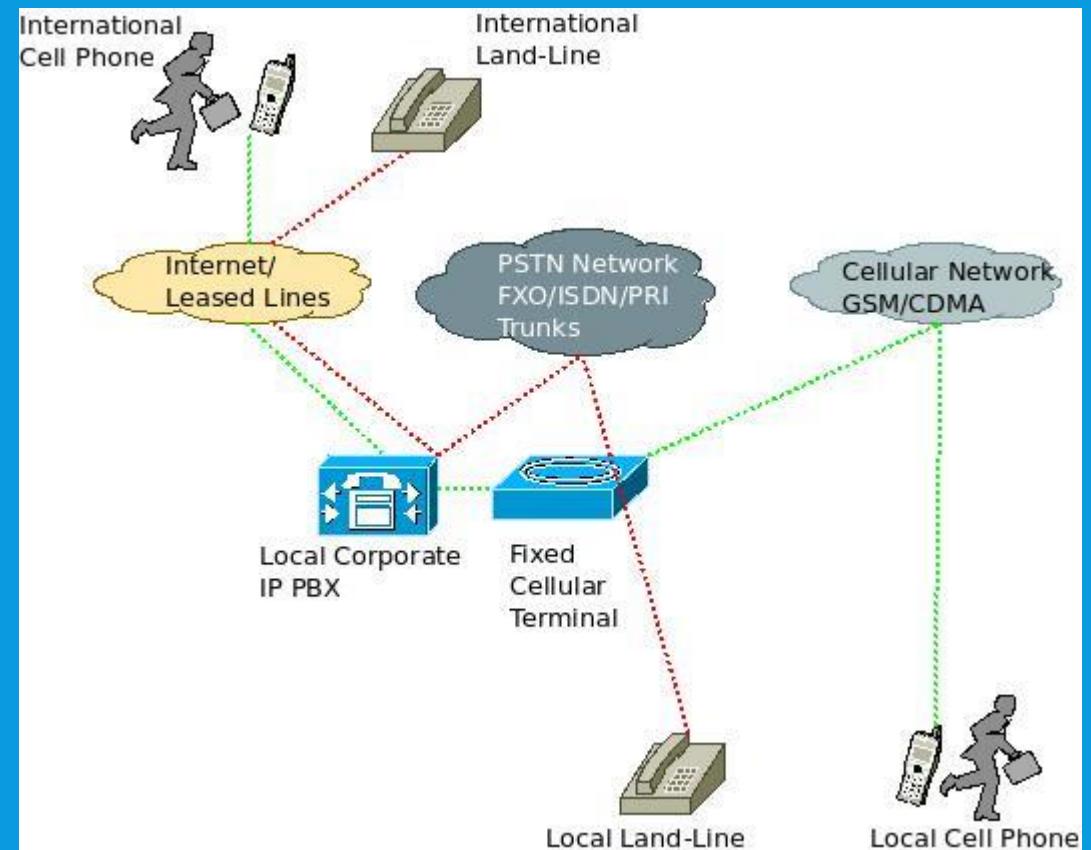


6. WHAT IS NEXT?

f. Can We Make Outbound Calls to PSTN Using our Asterisk Server?

❖ Using DISA

DISA (Direct Inward System Access) allows someone calling in from outside the telephone switch (PBX) to obtain an “internal” system dial tone and dial calls as if from one of the extensions attached to the telephone switch.



Thanks for your Attention



If later, feel free to contact me:



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