

## Program VOIP Phone

**Password for a Cisco 7900 series VOIP phone:** `**#` (key in the `**#` sequence using the phone keypad after hitting the Settings key). **Note:** `**#*` and `**#**` are possible variations on some phones. Use the sequence that works below.

**Resetting a Cisco 7900 series VOIP phone:** Hit Settings key, then key in: `**#` to unlock the network configuration menus. Enter option 3 (Network configuration), and then 33 (Erase configuration). Click "Yes" softkey, then click "Save" softkey.

**Statically assign IP addresses on a 7900 series phone:** Hit Settings key, then key in: `**#` to unlock the network configuration menus. Enter option 3 (Network configuration), then 30 (DHCP Enabled), then toggle DHCP Enabled to NO and hit Save softkey. Use options 6 and 7 for the phone's IP address and subnet mask. Use options 8 and 9 for the TFTP server and default router. You will also need to key in the CallManager address.

**Using DHCP instead of static assignments:** Hit Settings key, then key in: `**#` to unlock network configuration menus. Enter option 3 (Network configuration), then 30 (DHCP Enabled), then toggle DHCP Enabled to YES and hit the Save softkey.

**Web access:** You can find information about the phone by using a Web browser to access the phone with the URL: `http://<IP address of phone goes here.>`



## Add a VOIP Phone in CallManager

- In CallManager Administration view, select "Add a New Device" under Device menu.
  - Select Phone and hit Next button.
  - Select phone type of 7940 or 7960 and hit Next button.
  - Enter MAC address of phone (on outside of box for a new phone, also on a label on the back of the phone, also shown in Network Configuration menu using the phone keypad.)
  - Fill in remaining parameters; insert phone in database; reset the phone from the CallManager menu.
- VOIP phone will go through an initialization sequence while resetting, which is visible on the phone.

## Configure VLANs for VOIP Phone

Put voice traffic on VLAN 10, and data traffic on VLAN 20. Set up one switch as the VTP server, and the others as clients. Use domain TBU, password cisco, and reset revision number to 0. Create VLAN 20 and assign gateway interfaces to VLAN 20. Domain/password are case sensitive!

**Reset the revision to 0.**

**On each client, do:**  
`vtp domain temp`  
`vtp domain TBU`  
`vtp mode client`

**On server, do:**

`vtp mode server`  
`vtp domain TBU`  
`vtp password cisco`  
`vlan 20`  
`name Phones`

**On each switch port connecting to a gateway, do:**

`int fa0/9`  
`switchport access vlan 20`  
`switchport mode access`

Configure VLANs for VOIP: plug a VOIP phone into a 35xx port. Verify power over Ethernet:

`3550-pwr01# show power inline`

Configure VLANs:

`3550-pwr01(config-int)# switchport access vlan 10`  
`3550-pwr01(config-int)# switchport mode access`  
`3550-pwr01(config-int)# switchport voice vlan 20`  
`3550-pwr01(config-int)# spanning-tree portfast`  
`3550-pwr01(config-int)# spanning-tree bpduguard enable`

Voice traffic travels on VLAN 20; data traffic travels on vlan 10. Portfast bypasses a lengthy (up to 50 sec) port initialization which eliminates switching loops. Bpduguard protects against a switch being plugged into a port (and possibly causing switching loops) where portfast has been enabled.

Verify voice VLAN operation:

`3550-pwr01# show interface fa0/1 switchport`

## Private Line Automatic Ringdown (PLAR)

Set up a private line automatic ringdown (PLAR) connection between two FXS ports on the same gateway. Pick up one phone and the other rings. There is no need to dial any digits.

Port 1/0/0 has extension 7777. Port 1/0/1 has extension 4000.

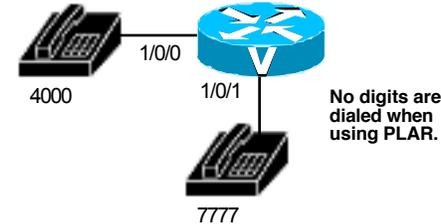
**NOTE:** These are example settings, but may not be the exact step-by-step CLI sequence, which will vary depending on the situation.

`voice-port 1/0/0`  
`connection plar 7777`

`voice-port 1/0/1`  
`connection plar 4000`

`dial-peer voice 1 pots`  
`destination-pattern 4000`  
`port 1/0/0`

`dial-peer voice 2 pots`  
`destination-pattern 7777`  
`port 1/0/1`



The following command verifies the configuration:

`show dial-peer voice`

## Dialing Between Phones

Modify the PLAR configuration above so that the caller dials 7777 or 4000 to reach the other phone.

`voice-port 1/0/0`  
`no connection plar 7777`

`voice-port 1/0/1`  
`no connection plar 4000`



A dialtone will be produced when either phone is picked up. Dial your own extension and you will hear a busy signal. Dial the other extension and the phone will ring.

## VOIP Between Gateways

Set up a VOIP connection over an IP network using H.323 protocol. The two local phones on 192.168.0.21 are 7777 and 4000. The two distant phones on 192.168.0.24 are 7778 and 4001.

**NOTE:** If you use different dial-peer identifiers than in the examples above, start by removing any previous dial-peers and connections with "no dial-peer voice 1 voip" and "no connection plar 7777" etc.

**Router 192.168.0.21**

`dial-peer voice 1 pots`  
`destination-pattern 4000`  
`port 1/0/0`

`dial-peer voice 2 pots`  
`destination-pattern 7777`  
`port 1/0/1`

`dial-peer voice 3 voip`  
`destination-pattern 4001`  
`session target ipv4:192.168.0.24`

`dial-peer voice 4 voip`  
`destination-pattern 7778`  
`session target ipv4:192.168.0.24`

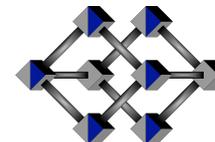
**Router 192.168.0.24**

`dial-peer voice 1 pots`  
`destination-pattern 4001`  
`port 1/0/0`

`dial-peer voice 2 pots`  
`destination-pattern 7778`  
`port 1/0/1`

`dial-peer voice 3 voip`  
`destination-pattern 4000`  
`session target ipv4:192.168.0.21`

`dial-peer voice 4 voip`  
`destination-pattern 7777`  
`session target ipv4:192.168.0.21`



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## Configure Signaling and Caller ID on an FXS Port

You can set the signaling and caller-id features on an FXS port:

```
voice-port 1/0/0
signal loopStart
station-id name Alex Bell
station-id number 9105553456
caller-id enable
```

The caller-id enable line enables caller-id on incoming calls.

## Hunt Groups

Hunt groups: the preference parameter determines which port is tried first. (Note: preference 0 should be the highest priority but cannot be configured in this way, so we start with the highest priority at 1.)

In this example, there are two FXS ports and two FXO ports. When an FXS user dials 9, then the lower numbered FXO port is preferred. If that line is busy then it hunts to the other FXO port. Objective: the preferred line is the internal (non toll) network while the other is the toll network, which is used for overflow or backup when the internal network is down.

Ports 1/1/0 and 1/1/1 should be connected to a wall jack or a phone line on a private branch exchange (PBX), or another gateway.

```
dial-peer voice 1 pots
destination-pattern 4000
port 1/0/0
```

```
dial-peer voice 2 pots
destination-pattern 4001
port 1/0/1
```

```
dial-peer voice 3 pots
preference 1
destination-pattern 9
port 1/1/0
```

```
dial-peer voice 4 pots
preference 2
destination-pattern 9
port 1/1/1
```

## Number Expansion

Using digit translation, we can change the local phones on router 192.168.0.21 to 6000 and 6777, and then use num-exp to map between 6000/6777 and 4000/7777. In this way, distant users can still use the old numbers 4000 and 7777. In practice, you can use num-exp to move a cluster of phones from one numbering plan to another, such as when moving from one public switched telephone network (PSTN) to another, while keeping all of the other dial code tables in your internal network unchanged while you slowly migrate your internal network to the new PSTN.

**Router 192.168.0.21 (starting with the setup from "VOIP Between Gateways")**

```
dial-peer voice 1 pots
destination-pattern 6777
port 1/0/0
```

```
dial-peer voice 2 pots
destination-pattern 6000
port 1/0/1
```

```
num-exp 7777 6777 The syntax is num-exp incoming-pattern final-pattern. Ex:
num-exp 4000 6000 4000 is the incoming pattern, which is translated to 6000.
```

Now test it out: from 192.168.0.24, dial 4000 or 7777, and the 6000 or 6777 phones will ring, respectively. Note that the num-exp expansion is made *before* pattern matching is applied, and that it cannot be cascaded. num-exp is applied in a single pass.

## Configure Netmeeting as an H.323 Client

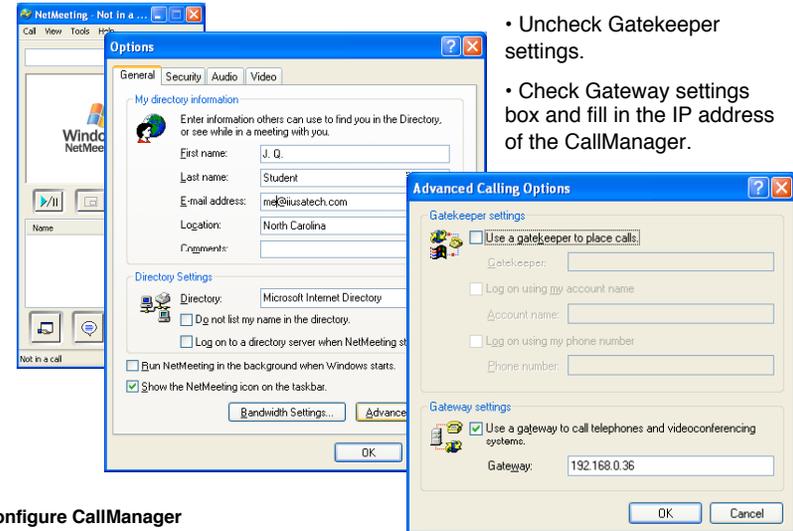
Configure NetMeeting in Windows:

Start NetMeeting: START --> Run... --> conf

Tools --> Options --> Advanced Calling --> Gateway

Place IP address of CallManager in Gateway field.

Bandwidth Settings --> Local Area Network



Configure CallManager

Log into the CallManager.

Device --> Add a New Device. Use Device Type of "Gateway" and click NEXT.

On next screen, choose H.323 Gateway for the Gateway type, and H.225 for the protocol. Click NEXT.

On next screen, place the IP address of the NetMeeting client in the Device Name field. Fill in the other parameters according to need. Click INSERT.

Create either a route pattern, or a pattern-->list-->group that points to the new Gateway device.

Test it out: from a CallManager phone call the route pattern, and the NetMeeting application will ring. From within NetMeeting, place a call to a CallManager phone.

## Server Configuration

CallManager Administration screen: <http://192.168.20.6/CCMAdmin>

CallManager User screen: <http://192.168.20.6/CCMUser>

- Enable DHCP Server in Control panel-->Administrative Tools-->Services.
  - Assign a static IP address to the server machine (otherwise it will not give out DHCP addresses. The server cannot obtain its own address via DHCP.)
  - Go to Control panel-->Administrative Tools-->DHCP and Add a server with a scope of addresses.
  - Add option 150 via Set Predefined Options... to point to the TFTP server on the same machine.
- Settings for option 150 are:
- Name: TFTP Server IP Address
  - Data Type: IP Address
  - Code: 150
  - Description: TFTP Server IP Address for CallManager Devices

- Click OK then Edit Array to enter the TFTP address

Note that it can take several minutes for the DHCP server to start giving out addresses. Remember that the VOIP phones need to have DHCP enabled.

## Inter-Cluster Trunk

- In CallManager Administration view, select “Add a New Device” under Device menu.
- Select Trunk and hit Next button.
- Select Trunk type of Inter-Cluster Trunk (non-Gatekeeper controlled) and hit Next button.
- Enter name and description, and IP address of remote CallManager (**not** the local CallManager IP address.)
- Fill in remaining parameters and insert trunk into database.

The trunk will then be available as a device for Route Plan configuration. NOTE: There can be only one Inter-Cluster trunk to the same destination. You will need to create trunks in both directions for calls to go through.

- In CallManager Administration view, create a new Route Group and point it to the trunk (there can only be one Route group per trunk.)
- Create a route list for the new trunk route group. There can be any number of route lists that point to the same route group.
- Create a route pattern for the new route list. There can be any number of route patterns that point to the same route list.
- Test the trunk by dialing a route pattern.



## MeetMe Conferencing

An ad hoc conference can be started with an analog hook flash, or using the Conf softkey on a Cisco VOIP phone.

In a MeetMe conference, the controller starts the conference at a specified number, and users dial into that number.

- 1) Set up a range of conference numbers in CallManager.
- 2) Conference controller goes offhook on a Cisco VOIP phone, hits More... softkey, then selects MeetMe softkey.
- 3) Controller dials a MeetMe number in the specified range.
- 4) Other users then dial the MeetMe number directly. They should not use the MeetMe softkey.

Note that users can be located anywhere but that the controller must be on a VOIP phone registered with the CallManager hosting the MeetMe number range.

## CallManager Attendant Configuration

1. In CallManager, go to Application-->Install Plugins-->Cisco Callmanager Attendant Console and install the application.
2. From Cisco CallManager Administration, choose User --> **Add a New User**.
3. In the First Name and Last Name fields, enter **ac**.
4. In the User ID field, enter **ac**.
5. In the User Password field and the Confirm Password field, enter **12345**.
6. Enter a PIN and telephone number.
7. Check the **Enable CTI Application Use** check box and click **Insert**.
8. Associate the devices by performing the procedure in **Associating Devices**. Phone lines for associated devices become available for the attendants.
9. Check the **Enable Call Park Retrieval Allowed** check box for ac user.
10. Add yourself as a user in: Service-->Cisco CM Attendant Console-->**Cisco CM Attendant Console User**.
11. Start up the Cisco CallManager Attendant Console.

## Wildcards

Set up four gateways as exchanges 621-XXXX, 622-XXXX, 623-XXXX, and 624-XXXX. We will use 7-digit dialing. See farther down for wildcard rules.

### For the 621 exchange

```
dial-peer voice 1 pots
destination-pattern 6211000
voice-port 1/0/0
```

```
dial-peer voice 2 voip
destination-pattern 622....
Use IP address for 622 gateway
session target ipv4:192.168.0.20
```

```
dial-peer voice 3 voip
destination-pattern 623....
Use IP address for 623 gateway
session target ipv4:192.168.0.21
```

```
dial-peer voice 4 voip
destination-pattern 624....
Use IP address for 624 gateway
session target ipv4:192.168.0.22
```

### For the 622 exchange

```
dial-peer voice 1 pots
destination-pattern 6221000
voice-port 1/0/0
```

```
dial-peer voice 2 voip
destination-pattern 623....
Use IP address for 621 gateway
session target ipv4:192.168.0.19
```

```
dial-peer voice 3 voip
destination-pattern 624....
Use IP address for 623 gateway
session target ipv4:192.168.0.21
```

```
dial-peer voice 4 voip
destination-pattern 621....
Use IP address for 624 gateway
session target ipv4:192.168.0.22
```

### For the 623 exchange

```
dial-peer voice 1 pots
destination-pattern 6231000
voice-port 1/0/0
```

```
dial-peer voice 2 voip
destination-pattern 621....
Use IP address for 621 gateway
session target ipv4:192.168.0.19
```

```
dial-peer voice 3 voip
destination-pattern 622....
Use IP address for 622 gateway
session target ipv4:192.168.0.20
```

```
dial-peer voice 4 voip
destination-pattern 624....
Use IP address for 624 gateway
session target ipv4:192.168.0.22
```

### For the 624 exchange

```
dial-peer voice 1 pots
destination-pattern 6241000
voice-port 1/0/0
```

```
dial-peer voice 2 voip
destination-pattern 621....
Use IP address for 621 gateway
session target ipv4:192.168.0.19
```

```
dial-peer voice 3 voip
destination-pattern 622....
Use IP address for 622 gateway
session target ipv4:192.168.0.20
```

```
dial-peer voice 4 voip
destination-pattern 623....
Use IP address for 623 gateway
session target ipv4:192.168.0.21
```

Test by dialing the 7 digit extensions from any gateway: 621-1000, 622-1000, 623-1000, 624-1000. Note that the CallManager matches the most specific pattern, and so if there are destination-patterns for 6221234 and 622...., then 6221234 matches first.

## Wildcard Rules

1... matches 1000 through 1999 so only one dial-peer voip is needed for the entire number range. 200[0-5] matches 2000-2005. 30+ matches 30, 300, 3000. Ref: page VC-121 of Cisco VOIP manual.

Symbol	Description
.	Indicates a single-digit placeholder. For example, 555 . . . . matches any dialed string beginning with 555, plus at least four additional digits.
[ ]	Indicates a range of digits. A consecutive range is indicated with a hyphen (-); for example, [5-7]. A nonconsecutive range is indicated with a comma (,); for example, [5,8]. Hyphens and commas can be used in combination: [5-7,9]. <b>Note:</b> Only single-digit ranges are supported. For example, [98-102] is invalid.
( )	Indicates a pattern; for example, 408(555). It is used in conjunction with the symbol ?, %, or +.
?	Indicates that the preceding digit occurred zero or one time. Enter <b>ctrl-v</b> before entering ? from your keyboard.
%	Indicates that the preceding digit occurred zero or more times. This functions the same as the "*" used in regular expressions.
+	Indicates that the preceding digit occurred one or more times.
T	Indicates interdigit timeout. The router pauses to collect additional dialed digits.

## Configure MGCP on Gateway

Configure VG200 gateway to be controlled by CallManager via MGCP. Start by removing all dial peers. You cannot have other manually configured dial peers with MGCP.

**hostname VG200-A** This case-sensitive name must agree with the Domain Name used for the CallManager setup in the next section "Adding a Gateway in CallManager".

**ip host MCS7835 192.168.20.6** If DNS isn't running, the CallManager name must be manually resolved. In this case, the CallManager name is MCS7835.

```
mgcp Enable MGCP protocol
mgcp call-agent 192.168.20.6 Identify the CallManager
ccm-manager config server 192.168.20.6 TFTP server address
ccm-manager config Enable download of config files
mgcp dtmf-relay voip codec all mode out-of-band Codec type and DTMF function
ccm-manager mgcp Enable support for CCM within MGCP
```

**ip routing** Enable IP routing. Cisco documentation says this step is needed for VG200's but they seem to work fine from an initialized state without it.

**ip route 0.0.0.0 0.0.0.0 192.168.20.1** Without a default route, the CallManager may get phones to ring but the gateways won't be able to talk to each other if they are on different subnets.

Beware that these commands may vary depending on the IOS version and feature set being used. NOTE: Skip ccm-manager config above if it is unavailable in the IOS.

### FXS ports

```
dial-peer voice 1 pots
application MGCPAPP
port 1/0/0
```

```
dial-peer voice 2 pots
application MGCPAPP
port 1/0/1
```

### FXO ports

```
dial-peer voice 3 pots
application MGCPAPP
port 1/1/0
```

```
dial-peer voice 4 pots
application MGCPAPP
port 1/1/1
```

**Both ports 1/0/0 and 1/0/1 are affected by no shut**

```
voice-port 1/0/0
no shut
```

**Both ports 1/1/0 and 1/1/1 are affected by no shut**

```
voice-port 1/1/0
no shut
```

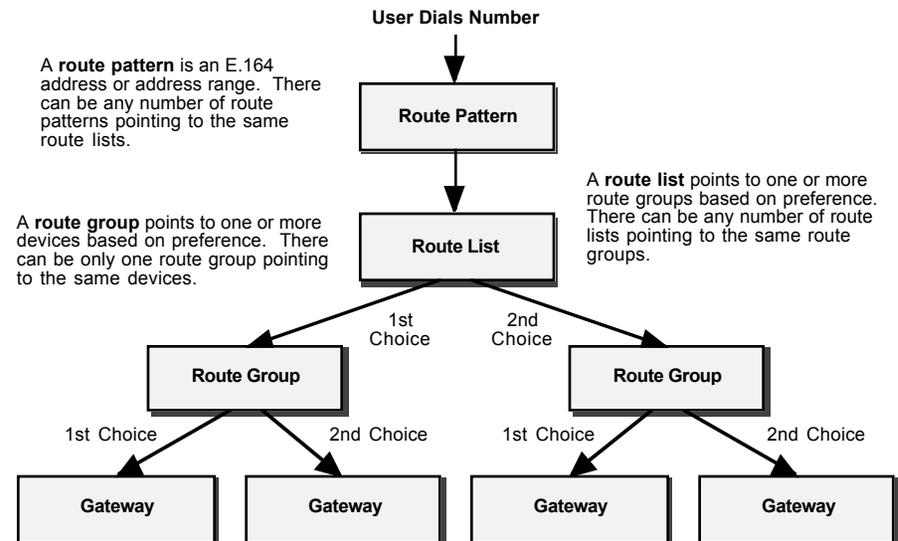
```
copy run start
```

## Adding a Gateway in CallManager

- In CallManager Administration view, select "Add a New Device" under Device menu.
- Select Gateway and hit Next button.
- Select Gateway type of VG200 or your router (such as 26xx) and hit Next button.
- Enter Domain Name which must be exactly the same as the gateway hostname. Description is filled in automatically.
- Fill in remaining parameters and insert gateway into database. Add phones to the gateway from the CallManager menu.
- Gateway must be reset from the CallManager for the changes to take effect.

## Route Pattern/List/Group Creation for FXS/FXO Ports

- In CallManager Administration view, create a new Route Group and point it to a gateway FXS port (there can only be one Route group per FXS port, or group of ports.)
- Create a route list for the new route group. There can be any number of route lists that point to the same route group.
- Create a route pattern for the new route list. There can be any number of route patterns that point to the same route list.
- Test the gateway phones by dialing your route pattern.
- For FXO ports, do the same setup as for the FXS ports above, but make the route pattern agree with the calling space on the attached PBX. For example, if the attached PBX has extensions 8765100 - 8765109, then use route pattern 876510X to the FXO port.
- Inbound calls from the PBX to the FXO port will need the attendant number configured.



## MGCP and H.323 Tips

- If you manually configure both MGCP and H.323 dial peers, be sure to configure the MGCP ones first if you are using the **ccm-manager config** auto-config feature.
- When changing the MGCP configuration, it is a good idea to reinitialize MGCP by issuing **no mgcp** and then **mgcp**.
- Use **show ccm-manager** to verify the gateway is registered with the correct Callmanager.
- Bind (source) all media and control traffic to a loopback interface when multiple paths exist:

```
interface Loopback99
ip address 10.1.49.99 255.255.255.0
mgcp bind control source-interface Loopback99
```
- "Hairpinning" between VOIP dial peers can be done on newer IOS's with:

```
voice service voip
allow-connections h323 to h323
```

The goal is to connect two VOIP dial peers, with no local POTS lines.