



Voice over IP Configurations

This chapter provides an overview of Voice over IP (VoIP) operations on the Cisco uBR925 cable access router. It also describes how to configure the Cisco uBR925 cable access router for basic VoIP operation in both bridging and routing modes.



Note

The information in this chapter applies only to the Cisco uBR925 cable access router.

This chapter contains the following sections:

- [Overview, page 4-1](#)
- [H.323v2 Static Bridging Configuration, page 4-7](#)
- [H.323v2 Static Routing Configuration, page 4-10](#)
- [H.323v2 Dynamic Mapping Configuration, page 4-12](#)
- [SGCP Configuration, page 4-17](#)
- [MGCP Configuration, page 4-19](#)



Note

The configurations shown in this chapter can be combined with most of the data-only configurations shown in [Chapter 3, “Advanced Data-Only Configurations.”](#) All voice configurations assume that the CMTS and associated servers, gateways, and gatekeepers have been configured accordingly.

Overview

When using a Cisco IOS image that contains voice support, the Cisco uBR925 cable access router supports Voice over IP (VoIP), which transmits voice and fax calls over a TCP/IP network such as the Internet. Depending on the services purchased from the cable service provider, subscribers can place and receive calls without using the local telco exchange carrier.

The Cisco uBR925 cable access router has two voice ports that support two simultaneous voice and fax calls from each subscriber site, but multiple telephones and fax devices can be connected to each of the two VoIP telephone lines (provided that the 5 Ringer Equivalence Number (REN) limit for each telephone line is not exceeded). Telephones at each subscriber site must support touch-tone dialing; rotary dialing is not supported. Special telephone features such as call waiting, forwarding, and conferencing are supported only when using Cisco IOS images that support those features.

**Note**

Fax devices—standard Group III and computer-based Group III machines up to 14,400 baud—are supported in Cisco IOS Release 12.0(5)T and higher images that support VoIP.

Introduction

The Cisco uBR925 cable access router uses IP packets to transmit and receive digitized voice over an IP network. Voice traffic is supported in both the DOCSIS-bridging and routing modes.

**Note**

When the router is acting in DOCSIS-bridging mode, a voice call originating from the router's Ethernet interface cannot terminate on another device attached to that same Ethernet interface; it must terminate on a device that is reached through the cable interface. The router must be operating in routing mode to allow calls to both originate and terminate on the Ethernet interface.

Voice signals are packetized and transported in compliance with the following protocols:

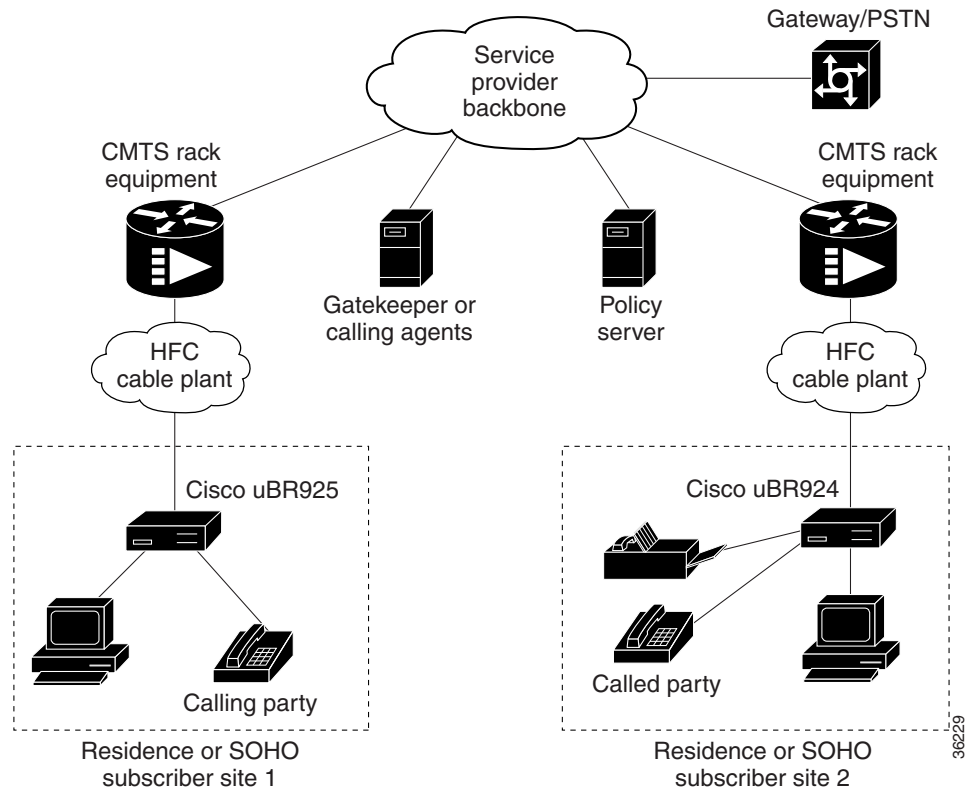
- H.323v2—Second version of an International Telecommunications Union (ITU) standard that specifies call signaling and control protocols for an IP data network.
- Simple Gateway Control Protocol (SGCP) Version 1.1—A signaling protocol under review by the Internet Engineering Task Force (IETF).
- Media Gateway Control Protocol (MGCP) Version 0.1—A proposed IETF voice control protocol intended to eventually supersede the existing SGCP 1.1 protocol.

**Note**

The MGCP 0.1 and SGCP 1.1 protocols have been merged on the Cisco uBR925 cable access router so that the router can respond efficiently to either protocol. The MGCP and SGCP protocols cannot be used if the H.323v2 protocol is used.

Figure 4-1 illustrates a typical broadband cable system that supports VoIP transmission. In this situation, a Cisco uBR925 cable access router is acting as the residential gateway for subscriber site 1, while a Cisco uBR924 router is acting as the residential gateway for subscriber site 2.

Figure 4-1 Simplified VoIP Network



The CMTS at the headend routes IP telephony calls from the point of origination to the destination, transmitting them along with other traffic (both voice and data). To route voice calls across the local IP network to a destination on the Internet or the public switched telephone network (PSTN), the Cisco uBR925 cable access router and CMTS deploy IP telephony as a local-loop bypass service. One of the following routing methods is then used, depending on the protocol being used:

- If using H.323v2, the Cisco uBR925 cable access router acts as the H.323v2 gateway that forwards the voice packets to the CMTS, which then sends them to a telephony gatekeeper. The gatekeeper transmits the packets to their ultimate destination.
- If using SGCP or MGCP, the Cisco uBR925 cable access router acts as the residential gateway that forwards the voice packets to the CMTS, which then connects to the external call agent (SGCP or MGCP) or media gateway controller (MGCP). The call agent or controller determines how to transmit the call across the network to the trunking gateway that will be its ultimate destination.

The gateway at the destination typically interconnects the IP network to the public switched telephone network (PSTN) so that calls can be made to any phone, not just those that are part of the IP telephony network.

Voice calls are digitized, encoded, compressed, and packetized in an originating gateway, and then, decompressed, decoded, and reassembled in the destination gateway. A server maintains subscriber profiles and policy information. See the Cisco service provider voice documentation set if you have Cisco gatekeeper, gateway, or other applicable products.

**Caution**

In certain countries, the provisioning of voice telephony over the Internet or use of these products may be prohibited and/or subject to laws, regulations or licenses, including requirements applicable to the use of the products under telecommunications and other laws and regulations; customer must comply with all such applicable laws in the country where the customer intends to use the product.

Voice Handling

With IP telephony, telephone calls can be delivered at rates as low as 8 kbps in a packet format using compression algorithms. Depending on the software release used, the Cisco uBR925 cable access router supports the following algorithms:

- G.711 A-Law—64,000 bps PCM uncompressed encoding, using the A-Law standard used in most of the world except for North America and a few other countries.
- G.711 Mu-Law—64,000 bps PCM uncompressed encoding, using the Mu-Law standard used in North America and a few other countries.
- G.723—5,300 bps CELP and 6300 bps ML-MLQ compression, as defined by H.324 family of standards.
- G.726—Adaptive Differential PCM (ADPCM) coding at 32, 24, and 16 kbps.
- G.728—16,000 bps low-delay variation of code excited linear prediction (CELP) voice compression that provides a good compromise between voice quality and signal compression.
- G.729—8,000 bps compressed CS-ACELP encoding (default for telephone calls).
- G.729 Annex B—8,000 bps compressed CS-ACELP encoding using the Annex B format.

**Caution**

Because voice is delay-sensitive, a well-engineered network is critical. Fine-tuning your network to adequately support VoIP typically involves a series of protocols and features geared to support QoS.

To achieve acceptable voice quality and reduce network bandwidth usage, several voice processing techniques are used. Digital Signal Processors (DSPs) provide the stream-to-packet and packet-to-stream conversion, as well as voice processing capabilities. Typical voice processing services include echo cancellation, voice compression, Voice Activity Detection (VAD) or silence compression, and Dual Tone Multi-Frequency (DTMF) tone detection and generation.

Quality of Service Support

Data traffic typically is sent only on a “best effort” basis, and if a packet is lost or delayed, it can be easily retransmitted without significantly affecting the connection. Such delays and losses are unacceptable, however, for real-time traffic such as voice calls.

For this reason, the CMTS and Cisco uBR925 router assign separate service identifiers (SIDs) for the voice and data traffic flows. Each SID has a separate class of service (CoS) that determines how its traffic flow is handled, allowing voice traffic to have a higher priority than the data traffic.

The CMTS and router can use different traffic shaping mechanisms to ensure that the higher priority voice traffic always has the bandwidth it needs. This allows voice calls (and other real-time traffic) to share the same channel as data traffic, without the quality of the voice calls being degraded by bursty data transmissions.

**Note**

Separate CoS flows are available only when the router is connected to a CMTS that supports multiple classes of service per router. In addition, the router's configuration file must enable multiple classes of service.

The DOCSIS 1.0 specification does not support multiple CoS flows, so this flow technique is not available when the Cisco uBR925 cable access router interoperates with a DOCSIS 1.0 CMTS. In this situation, voice and data traffic are both transmitted on a "best effort" basis. This may cause poorer voice quality and lower data throughput when calls are being made from the router's telephone ports.

The Cisco uBR925 cable access router supports the following service classes:

- The first CoS in the router's configuration file is configured as the "Tiered Best Effort Type Class" and is the default CoS for data traffic. The class has no minimum upstream rate specified for the channel.

This service class is assigned to the primary SID for the router. In addition to being used for data traffic, the router uses this SID for all MAC message exchanges with the CMTS, as well as for SNMP management traffic.

All traffic using this SID is transmitted on a "best effort" basis, but data traffic within this class can be prioritized into eight different priority levels; although all data traffic still has lower priority than the voice traffic, this allows certain data traffic (such as MAC messages) to be given higher priority than other data traffic. The CMTS system administrator defines the traffic priority levels and must include the traffic priority fields in the configuration file downloaded to the Cisco uBR925.

- The second and third CoS are for the first and second voice ports, respectively, which are assigned to the secondary SIDs used for the voice ports. If using a Cisco IOS image that supports dynamic multi-SID assignment, these secondary SIDs are automatically created when a call is placed from one of the voice ports; when the call terminates, the secondary SID associated with it is deleted. If the Cisco IOS image does not support multi-SIDs, static SIDs are created for each of the voice ports during the power-on provisioning process, permanently reserving the bandwidth needed for the voice traffic.

The CMTS system administrator typically configures these secondary classes of service so that they have higher QoS classes for use by higher priority voice traffic. These classes should also have a minimum upstream data rate specified for the channel to guarantee a specific amount of bandwidth for the corresponding traffic flows. When static SIDs are used, that bandwidth is always reserved for voice calls; however, when dynamic multi-SID assignment is used, that bandwidth is reserved only when the voice calls are active.

H.323v2 Protocol

In architectures using the VoIP H.323v2 protocol stack, the session application manages two call legs for each call: a telephony leg managed by the voice telephony service provider and the VoIP leg managed by the cable system operator—the VoIP service provider. Use of the H.323v2 protocol typically requires a dial plan and mapper at the headend or other server location to map IP addresses to telephone numbers.

When both legs of the call have been set up, the session application creates a conference between them. The opposite leg's transmit routine for voice packets is given to each provider. The CMTS router passes data to the gateway and gatekeeper. The H.323v2 protocol stack provides signaling through H.225 and feature negotiation through H.245.

**Note**

For more information on using H.323v2, see the document *H.323 Version 2 Support*, available on Cisco.com and the Documentation CD-ROM.

To make and receive H.323 calls, the Cisco uBR925 cable access router must be configured for the following:

- The IP address of the gateway for the destination dialed—The Cisco uBR925 cable access router can configure these IP addresses statically via the command-line interface (CLI) using **voip dial peer group** commands. When using Cisco gatekeeper products running Cisco IOS Release 12.0(5)T or higher images, the Cisco uBR925 cable access router can obtain these addresses dynamically from the gatekeeper using the Registration, Admission, and Status (RAS) protocol.
- The telephone numbers of the attached devices—The Cisco uBR925 cable access router can configure these IP addresses statically via the CLI **pots port** commands. When using Cisco Network Registrar (CNR) version 3.0 or higher with the **relay.tcl** and **setrouter.tcl** scripts, and Cisco gatekeeper products running Cisco IOS Release 12.0(5)T or higher images, you can obtain these addresses dynamically from CNR. The telephone numbers of attached devices are then sent in DHCP response messages. When the Cisco uBR925 processes the DHCP response, it automatically creates the **pots dial peer** for each port, creates the **voip dial peer** for the RAS target, and starts the H.323v2 RAS gateway support.

**Tip**

To configure H.323 voice calls for use with IPsec encryption over a Virtual Private Network (VPN), use the **h323-gateway voip bind srcaddr** command so that VoIP traffic is sent using the IP address of the Ethernet address, instead of the IP address used for the cable interface.

**Note**

To support voice configurations on Cisco gatekeeper products using RAS, Cisco IOS Release 12.0(5)T or higher images with gatekeeper support are required. The headend must have IP multicast enabled. The cable interface must be designated as the default for RAS to discover the gatekeeper. The gatekeeper then resolves all dialed destinations sent to the RAS protocol.

SGCP and MGCP Protocol Stack

The Cisco uBR925 router supports both the SGCP 1.1 and MGCP 0.1 voice signaling protocols. The MGCP protocol is intended to eventually supersede the SGCP protocol. Both MGCP and SGCP are signaling protocols that interact with a remote call agent (CA) to provide call setup and teardown for VoIP calls.

Using the call agent, SGCP and MGCP communicate with the voice gateways, dynamically resolving and routing calls. This creates a distributed system that enhances performance, reliability, and scalability while still appearing as a single VoIP gateway to external clients.

The remote call agent also provides the signaling and feature negotiation that would otherwise be provided by the Cisco uBR925 cable access router when using the H.323v2 protocol. Similarly, the call agent also provides the mapping of IP addresses to telephone numbers, eliminating the dial plan mapper and static configurations that are required on the router when using the H.323v2 protocol.

The SGCP and MGCP protocols implement the gateway functionality using both trunk and residential gateways. The Cisco uBR925 cable access router functions in this mode as a residential gateway with two endpoints.

SGCP and MGCP can preserve Signaling System 7 (SS7) style call control information as well as additional network information such as routing information and authentication, authorization, and accounting (AAA) security information. SGCP and MGCP allow voice calls to originate and terminate on the Internet, as well as allowing one end to terminate on the Internet and the other to terminate on a telephone on the PSTN.

**Note**

The Cisco uBR925 cable access router supports both H.323 and SGCP/MGCP call control, but only one method can be active at a time.

H.323v2 Static Bridging Configuration

When the Cisco uBR925 cable access router is running in DOCSIS-bridging mode and using a Cisco IOS image with voice support, it can route voice calls using an H.323v2 static dialing map. This requires the following minimum configuration:

- Create a local dial peer for each voice port that will receive incoming calls. This requires configuring each voice port on the router with the phone numbers for the devices attached to those voice ports. The Cisco uBR925 cable access router uses these numbers to determine which voice port should receive the call. Typically, the complete phone number or extension is specified for each port; when the Cisco uBR925 router receives an incoming call, all digits in the number are matched and stripped off, and the voice port is connected to the call.

**Note**

The voice ports on the Cisco uBR925 cable access router support only FXS devices.

- Configure a remote dial peer for each possible destination for outgoing calls. This requires specifying the phone number(s) for the destination devices. Use the following guidelines for what numbers to enter:
 - For a single telephony device, such as a one-line phone or fax machine, enter the complete phone number or extension.
 - To direct a group of numbers to a specific destination—such as the extensions used on a remote PBX—enter a pattern matching the prefix used for those lines; asterisks can be used to match any number of digits and a period matches a single digit. For example, “572*” matches any phone numbers starting with 572 while “572.” matches the numbers 5720–5729.

You must also specify the IP address for the destination host that will deliver the call to the telephony device (or if the destination device is an IP telephone, the IP address for that telephone). You can optionally specify an IP precedence level for the type of service (ToS) bits in the IP header to signify that these voice packets should be given higher priority in transit across the IP network.

If not being done by the CoS, you can also specify which coding/decoding (CODEC) algorithm should be used.

These functions are done using the **dial-peer** command, as shown in the following table:

| | Command | Purpose |
|---------|---|--|
| Step 1 | To configure incoming calls on voice port V1: uBR925(config)# dial-peer voice <i>id-number</i> pots | Specify a unique <i>id-number</i> for this incoming dial-peer and enter dial-peer configuration mode. |
| Step 2 | uBR925(config-dial-peer)# destination-pattern <i>digits</i> | Specify the telephone number(s) associated with this voice port. |
| Step 3 | uBR925(config-dial-peer)# port 0 | Specify that voice port V1 is attached to this telephony equipment. |
| Step 4 | uBR925(config-dial-peer)# dtmf-relay [cisco-rtp] [h245-signal] [h245-alphanumeric] | Optionally configure the dial peer to support out of band signaling of DTMF tones. |
| Step 5 | uBR925(config-dial-peer)# exit | Exit dial-peer configuration mode. |
| Step 6 | To configure incoming calls on voice port V2: uBR925(config)# dial-peer voice <i>id-number</i> pots | Specify a unique <i>id-number</i> for this incoming dial peer and enter dial-peer configuration mode. |
| Step 7 | uBR925(config-dial-peer)# destination-pattern <i>digits</i> | Specify the telephone number(s) associated with this voice port. |
| Step 8 | uBR925(config-dial-peer)# port 1 | Specify that voice port V2 is attached to this telephony equipment. |
| Step 9 | uBR925(config-dial-peer)# dtmf-relay [cisco-rtp] [h245-signal] [h245-alphanumeric] | Optionally configure the dial peer to support out-of-band signaling of DTMF tones. |
| Step 10 | uBR925(config-dial-peer)# exit | Exit dial-peer configuration mode. |
| Step 11 | Repeat for each possible outgoing destination: uBR925(config)# dial-peer voice <i>id-number</i> voip | Specify a unique <i>id-number</i> for this outgoing dial peer and enter dial-peer configuration mode. |
| Step 12 | uBR925(config-dial-peer)# destination-pattern <i>digits</i> | Specify the telephone number(s) associated with this dial peer. |
| Step 13 | uBR925(config-dial-peer)# session target [ipv4:ipaddress dns:hostname] | Specify the destination IP address or hostname for this dial peer. This could be the IP address or hostname for either an IP telephone or another router or host providing voice services. |
| Step 14 | uBR925(config-dial-peer)# ip precedence <i>number</i> | (Optional) Specify an IP packet precedence level (1-5) for packets carrying calls to this dial peer (1–5, where 5 is the highest precedence for normal IP flows). |
| Step 15 | uBR925(config-dial-peer)# code [g711alaw g711ulaw g726r16 g726r24 g726r32 g729br8 g729r8] | (Optional) Specify the codec algorithm to be used for these calls. The default is g711r8 (8Kbps compression; A-Law and Mu-Law are 64Kbps compression). |
| Step 16 | uBR925(config-dial-peer)# dtmf-relay [cisco-rtp] [h245-signal] [h245-alphanumeric] | Optionally configure the dial peer to support out of band signaling of DTMF tones. |
| Step 17 | uBR925(config-dial-peer)# exit | Exit dial-peer configuration mode. |
| Step 18 | uBR925# copy running-config startup-config Building configuration... | Save the configuration to nonvolatile memory so that it will not be lost in the event of a reset, power cycle, or power outage. |
| Step 19 | uBR925# show startup-config | Display the configuration file that was just created. |

**Note**

The ID numbers assigned using the **dial-peer voice** command must be unique but they are local to the Cisco uBR925 router. These numbers are used only when configuring each particular dial peer and have no meaning when dialing numbers or routing calls.

The following example shows a Cisco uBR925 cable access router set up to support bridging and a static H.323 dial map with the following characteristics:

- Voice port V1 is connected to a telephony device that receives calls for the number 4123.
- Voice port V2 is connected to a telephony device that receives calls for the number 4124.
- Outgoing calls to the numbers 6000—6999 are routed to the dial peer at IP address 10.1.71.65.
- Outgoing calls to the numbers 7000—7999 are routed to the dial peer at IP address 10.1.71.75. These calls are sent with an IP ToS precedence of “5” and using the G.711 Mu-law codec algorithm.

The commands that set up the H.323v2 dial map are shown in bold:

```

version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname uBR925
!
clock timezone - 3
ip subnet-zero
no ip routing
!
!
voice-port 0
input gain -3
!
voice-port 1
input gain -3
!
dial-peer voice 1 pots
destination-pattern 4123
port 0
!
dial-peer voice 2 pots
destination-pattern 4124
port 1
!
dial-peer voice 1001 voip
destination-pattern 6...
session target ipv4:10.1.71.65
dtmf-relay cisco-rtp h245-signal h245-alphanumeric
!
dial-peer voice 1002 voip
destination-pattern 7...
ip precedence 5
codec g711ulaw
session target ipv4:10.1.71.75
dtmf-relay cisco-rtp h245-signal h245-alphanumeric
!
!
interface Ethernet0
no ip directed-broadcast
no ip route-cache
bridge-group 59

```

```

bridge-group 59 spanning-disabled
!
interface cable-modem0
 ip address docsis
 no ip directed-broadcast
 no ip route-cache
 cable-modem downstream saved channel 537000000 26
 bridge-group 59
 bridge-group 59 spanning-disabled
!
interface USB0
 ip address negotiated
 no ip route-cache
 arp timeout 0
 bridge-group 59
 bridge-group 59 spanning-disabled
!
!
 ip classless
 no ip http server
 no service finger
!
!
 line con 0
   exec-timeout 0 0
   transport input none
 line vty 0 4
   login
end

```

H.323v2 Static Routing Configuration

When the Cisco uBR925 cable access router is operating in routing mode, the configuration of an H.323v2 static dial map uses the same commands as those given in the “[H.323v2 Static Bridging Configuration](#)” section on page 4-7. The only difference is that calls can terminate and originate on the Ethernet interface, which is not possible in DOCSIS-bridging mode.



Tip

To configure H.323 voice calls for use with IPsec encryption over a Virtual Private Network (VPN), use the **h323-gateway voip bind srcaddr** command so that VoIP traffic is sent using the IP address of the Ethernet address, instead of the IP address used for the cable interface.

The following sample configuration shows a Cisco uBR925 cable access router set up for a static H.323v2 dial map with the following characteristics:

- Local dial peer 1 specifies that voice port V1 is connected to a telephone or fax machine with the number 6101.
- Local dial peer 2 specifies that voice port V2 is connected to a telephone or fax machine with the number 6102.
- Remote dial peer 101 specifies that calls to numbers 6200–6299 should be routed to IP address 10.1.71.62.
- Remote dial peers 102 and 103 specify that calls to numbers 6101 and 6102 should be routed to IP address 24.1.61.5, which is the IP address for the Cisco uBR925 cable access router’s Ethernet interface. This allows the router to complete calls between voice ports V1 and V2.

The commands related to the dial map are in bold.

```
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
!
hostname uBR925
!
!
!
class-map class-default
  match any
!
!
!
clock timezone - 3
ip subnet-zero
!
!
!
voice-port 0
!
voice-port 1
!
dial-peer voice 1 pots
  destination-pattern 6101
  port 0
!
dial-peer voice 2 pots
  destination-pattern 6102
  port 1
!
dial-peer voice 101 voip
  destination-pattern 62..
  session target ipv4:10.1.71.62
  dtmf-relay cisco-rtp
!
dial-peer voice 102 voip
  destination-pattern 6101
  session target ipv4:24.1.61.5
!
dial-peer voice 103 voip
  destination-pattern 6102
  session target ipv4:24.1.61.5
  dtmf-relay cisco-rtp
!
!
interface Ethernet0
  ip address 24.1.61.1 255.255.255.0
  no ip directed-broadcast
  no ip mroute-cache
!
interface cable-modem0
  ip address docsis
  no ip directed-broadcast
  no ip mroute-cache
  no cable-modem compliant bridge
!
interface USB0
  ip address 24.1.62.1 255.255.255.0
  no ip directed-broadcast
  no ip route-cache
  arp timeout 0
!
```

```

!
router rip
  version 2
  network 10.0.0.0
  network 24.0.0.0
  no auto-summary
!
no ip classless
ip route 0.0.0.0 0.0.0.0 10.1.71.1
no ip http server
no service finger
!
!
line con 0
  exec-timeout 0 0
  transport input none
line vty 0 4
  login
!
!
end

```

**Note**

The above configuration assumes that the DHCP server assigns an IP address to the cable interface that is in the class A private network (10.0.0.0).

H.323v2 Dynamic Mapping Configuration

The Cisco uBR925 cable access router supports using the Registration, Admission, and Status (RAS) protocol to allow a remote gatekeeper to translate phone numbers (E.164 addresses) to the IP addresses of specific dial peers. This allows the gatekeeper to maintain a central database of dial peers, so that this information does not have to be entered into static dial maps on every router that is acting as a voice gateway.

**Note**

The Cisco uBR925 cable access router can use H.323v2 dynamic mapping in either DOCSIS-bridging mode or routing mode.

The example shown in this section assumes that Cisco Network Registrar (CNR) version 3.0 or higher is being used as the DHCP server. CNR assigns the E.164 addresses to local voice ports and uses DHCP to define the E.164 addresses-to-port assignments.

The gatekeeper can be a Cisco router, such as the Cisco 3620, with a Cisco IOS image that supports the gatekeeper function. The Cisco uBR925 cable access router acts as the H.323v2 gateway and creates the dial peers, starts H.323 RAS gateway support, and registers the E.164 addresses with the gatekeeper. The gatekeeper resolves the remote peers' IP addresses when the router sends a request using RAS.

**Note**

Support for RAS and H.323v2 in Cisco gatekeeper products is found in Cisco IOS Release 12.0(5)T or higher. Support for multiple classes of service when using Cisco uBR7200 CMTS equipment is found in Cisco 12.0(4)XI or higher.

If you are not using CNR or Cisco gatekeeper products running Cisco IOS Release 12.0(5)T software, use a static dial-map as shown in the previous H.323 configurations ([“H.323v2 Static Bridging Configuration”](#) and [“H.323v2 Static Routing Configuration”](#)).

You must do the following to configure the Cisco uBR925 cable access router for dynamic mapping:

- Configure the local dial peers—This is done in the same way as for a static H.323v2 dial map.
- Configure the remote dial peers—This is done in the same way as for a static H.323v2 dial map, except that instead of specifying a target IP address or host name, you specify **ras** as the target.
- Enable the VoIP gateway function using the **gateway** global configuration command.
- Configure the cable modem interface to be the gateway interface.

These functions are done using the commands shown in the following table:

| | Command | Purpose |
|---------|---|---|
| Step 1 | To configure incoming calls on voice port V1: uBR925(config)# dial-peer voice <i>id-number</i> pots | Specify a unique <i>id-number</i> for this incoming dial peer and enter dial-peer configuration mode. |
| Step 2 | uBR925(config-dial-peer)# destination-pattern <i>digits</i> | Specify the telephone number(s) associated with this voice port. |
| Step 3 | uBR925(config-dial-peer)# port 0 | Specify that voice port V1 is attached to this telephony equipment. |
| Step 4 | uBR925(config-dial-peer)# dtmf-relay [cisco-rtp] [h245-signal] [h245-alphanumeric] | Optionally configure the dial peer to support out of band signaling of DTMF tones. |
| Step 5 | uBR925(config-dial-peer)# exit | Exit dial-peer configuration mode. |
| Step 6 | To configure incoming calls on voice port V2: uBR925(config)# dial-peer voice <i>id-number</i> pots | Specify a unique <i>id-number</i> for this incoming dial peer and enter dial-peer configuration mode. |
| Step 7 | uBR925(config-dial-peer)# destination-pattern <i>digits</i> | Specify the telephone number(s) associated with this voice port. |
| Step 8 | uBR925(config-dial-peer)# port 1 | Specify that voice port V2 is attached to this telephony equipment. |
| Step 9 | uBR925(config-dial-peer)# dtmf-relay [cisco-rtp] [h245-signal] [h245-alphanumeric] | Optionally configure the dial peer to support out of band signaling of DTMF tones. |
| Step 10 | uBR925(config-dial-peer)# exit | Exit dial-peer configuration mode. |
| Step 11 | Repeat for each possible outgoing destination: uBR925(config)# dial-peer voice <i>id-number</i> voip | Specify a unique <i>id-number</i> for this outgoing dial peer and enter dial-peer configuration mode. |
| Step 12 | uBR925(config-dial-peer)# destination-pattern <i>digits</i> | Specify the telephone number(s) associated with this dial peer. |
| Step 13 | uBR925(config-dial-peer)# session target ras | Specify that RAS will be used to resolve the destination for the dial peer. |
| Step 14 | uBR925(config-dial-peer)# ip precedence <i>number</i> | (Optional) Specify an IP packet precedence level (1-5) for packets carrying calls to this dial peer (1-5, where 5 is the highest precedence for normal IP flows). |
| Step 15 | uBR925(config-dial-peer)# code [g711alaw g711ulaw g723ar53 g723ar63 g726r16 g726r24 g726r32 g729br8 g729r8] | (Optional) Specify the codec algorithm to be used for these calls. The default is g711r8 (8Kbps compression; A-Law and Mu-Law are 64Kbps compression). |
| Step 16 | uBR925(config-dial-peer)# dtmf-relay [cisco-rtp] [h245-signal] [h245-alphanumeric] | Optionally configure the dial peer to support out of band signaling of DTMF tones. |

| | Command | Purpose |
|---------|---|---|
| Step 17 | uBR925(config-dial-peer)# exit | Exit dial-peer configuration mode. |
| Step 18 | uBR925(config)# gateway | Enable the VoIP gateway on the Cisco uBR925 router. |
| Step 19 | uBR925(config)# interface cable-modem 0 | Enter interface configuration mode for the cable interface. |
| Step 20 | uBR925(config-if)# (enter appropriate cable interface configuration commands) | Enter whatever commands are needed to configure the cable interface such as IP address, downstream channel, whether DOCSIS-bridging is enabled, and so forth. |
| Step 21 | uBR925(config-if)# h323-gateway voip interface | Specify that the cable interface is the H.323 Gateway VoIP interface. |
| Step 22 | uBR925(config-if)# h323-gateway voip id <i>gatekeeper-id ipaddr IP-address port-number</i> | Identify the RAS gatekeeper by specifying its gatekeeper ID (which must match the ID configured on the gatekeeper), its IP address, and the port number which services gateway requests. |
| Step 23 | uBR925(config-if)# h323-gateway voip h323-id <i>interface-id</i> | Specify the H.323 ID for this interface. This ID is any string that uniquely identifies this gateway to the gatekeeper. Typically, this is the gateway's name and domain (such as " uBR925@cisco.com "). |
| Step 24 | uBR925(config-if)# h323-gateway voip tech-prefix <i>prefix</i> | (Optional) Specify a technology prefix to identify the type of service this gateway can provide. If more than one service is being provided, give this command for each separate technology prefix. (The prefix is defined at the gatekeeper and can be up to 11 characters long, with the pound sign (#) as the last character.) |
| Step 25 | uBR925(config-if)# exit | Exit interface configuration mode. |
| Step 26 | uBR925(config)# interface Ethernet 0 | Enter interface configuration mode for the Ethernet interface. |
| Step 27 | uBR925(config-if)# (enter appropriate Ethernet interface configuration commands) | Enter whatever commands are needed to configure the cable interface such as IP address, downstream channel, and so forth. |
| Step 28 | uBR925(config-if)# h323-gateway voip bind srcaddr <i>ip-address</i> | (Optional) Configure the VoIP traffic so that it is sent using the IP address of the Ethernet interface, to allow VoIP calls to be made over an IPsec VPN connection. Note The <i>ip-address</i> should be the same as the one assigned to the Ethernet interface. |
| Step 29 | uBR925(config-if)# exit | Exit interface configuration mode. |
| Step 30 | uBR925(config)# exit | Exit global configuration mode. |
| Step 31 | uBR925# copy running-config startup-config Building configuration... | Save the configuration to nonvolatile memory so that it will not be lost in the event of a reset, power cycle, or power outage. |
| Step 32 | uBR925# show startup-config | Display the configuration file that was just created. |

**Note**

For additional information on the gateway configuration commands, see the document *Configuring H.323 VoIP Gateway for Cisco Access Platforms*, available on Cisco.com and the Document CD-ROM.

The following configuration shows a Cisco uBR925 cable access router configured for routing mode and using RAS dynamic mapping with the following characteristics:

- The router's V1 voice port is connected to a telephone or fax machine with the number 1000, and the V2 voice port is connected to a telephone or fax machine with the number 1001.
- Four remote dial-peers are configured, with the numbers 1000, 1001, 2000, and 2001. All use the G.711 Mu-Law CODEC, and the RAS protocol is used to resolve their number-address mapping. (The local dial-peer numbers, 1000 and 1001 are included as remote dial peers to allow the router to forward calls between the two local dial peers, as well as between local and remote dial peers; the router must be in routing mode to support this.)
- The cable interface is configured as the gatekeeper interface, using the gatekeeper named **gatekeeper3620** at the IP address **10.1.70.50** and at port **1719**. The router identifies itself as the gateway named **uBR925** with a tech-prefix of **1#**.

**Tip**

To configure H.323 voice calls for use with IPSec encryption over a Virtual Private Network (VPN), use the **h323-gateway voip bind srcaddr** command so that VoIP traffic is sent using the IP address of the Ethernet address, instead of the IP address used for the cable interface.

The commands related to the dial mapping are in bold.

```

version 12.2
service config
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname uBR925
!
clock timezone - 4
ip subnet-zero
ip host-routing
!
voice-port 0
!
voice-port 1
!
dial-peer voice 1 pots
destination-pattern 1000
port 0
!
dial-peer voice 2 pots
destination-pattern 1001
port 1
!
dial-peer voice 10 voip
destination-pattern 1001
codec g711ulaw
session target ras
!
dial-peer voice 20 voip
destination-pattern 1000
codec g711ulaw

```

```

    session target ras
  !
dial-peer voice 30 voip
  destination-pattern 2000
  codec g711ulaw
  session target ras
  !
dial-peer voice 40 voip
  destination-pattern 2001
  codec g711ulaw
  session target ras
  !
gateway
  !
  !
interface Ethernet0
  ip address 24.1.0.1 255.255.0.0
  no ip directed-broadcast
  no ip mroute-cache
  !
interface cable-modem0
  ip address docsis
  no ip directed-broadcast
  no ip mroute-cache
  no keepalive
  no cable-modem compliant bridge
  h323-gateway voip interface
  h323-gateway voip id gatekeeper3620 ipaddr 10.1.70.50 1719
  h323-gateway voip h323-id uBR925
  h323-gateway voip tech-prefix 1#
  !
interface USB0
  ip address 24.1.2.1 255.255.255.0
  no ip directed-broadcast
  no ip route-cache
  arp timeout 0
  !
  !
router rip
  version 2
  network 10.0.0.0
  network 24.0.0.0
  !
ip classless
no ip http server
no service finger
  !
  !
line con 0
  transport input none
line vty 0 4
  !
end

```

**Note**

The above configuration assumes that the DHCP server assigns an IP address to the cable interface that is in the class A private network (10.0.0.0).

SGCP Configuration

The Cisco uBR925 cable access router can use the SGCP protocol for routing voice calls. This transfers the dial mapping to an external call agent, so that the VoIP gateways do not have to be individually configured with the dial mappings.



Note

The Cisco uBR925 cable access router can use SGCP in either DOCSIS-bridging mode or routing mode.

You must do the following to configure the Cisco uBR925 cable access router for a dynamic mapping configuration:

- Enable SGCP operation on the Cisco uBR925 cable access router.
- Specify the SGCP call agent's IP address.
- Configure the local dial peers to be SGCP applications.
- Optionally enable the sending of SNMP traps for SGCP.



Note

No configuration of remote dial peers is needed when using SGCP.

These functions are done using the commands shown in the following table:

| | Command | Purpose |
|----------------|---|---|
| Step 1 | To configure incoming calls on voice port V1: uBR925(config)# dial-peer voice <i>id-number</i> pots | Specify a unique <i>id-number</i> for this incoming dial peer and enter dial-peer configuration mode. |
| Step 2 | uBR925(config)# application SGCPAPP | Specify that this dial peer is handled as an SGCP application. |
| Step 3 | uBR925(config-dial-peer)# destination-pattern <i>digits</i> | Specify the telephone number(s) associated with this voice port. |
| Step 4 | uBR925(config-dial-peer)# port 0 | Specify that voice port V1 is attached to this telephony equipment. |
| Step 5 | uBR925(config-dial-peer)# exit | Exit dial-peer configuration mode. |
| Step 6 | To configure incoming calls on voice port V2: uBR925(config)# dial-peer voice <i>id-number</i> pots | Specify a unique <i>id-number</i> for this incoming dial-peer and enter dial-peer configuration mode. |
| Step 7 | uBR925(config)# application SGCPAPP | Specify that this dial peer is handled as an SGCP application. |
| Step 8 | uBR925(config-dial-peer)# destination-pattern <i>digits</i> | Specify the telephone number(s) associated with this voice port. |
| Step 9 | uBR925(config-dial-peer)# port 1 | Specify that voice port V2 is attached to this telephony equipment. |
| Step 10 | uBR925(config-dial-peer)# exit | Exit dial-peer configuration mode. |
| Step 11 | uBR925(config)# sgcp | Enable SGCP operations on the router. |
| Step 12 | uBR925(config)# sgcp call-agent <i>ip-address</i> [<i>port</i>] | Specify the IP address and optional UDP port number for the SGCP call-agent. If no port number is given, the default of 2427 (the well-known SGCP port number) is used. |

| | Command | Purpose |
|---------|--|---|
| Step 13 | uBR925(config)# snmp-server enable traps xgcp | (Optional) If SNMP management is used for this router, specify that SGCP and related traps be sent to the SNMP manager. |
| Step 14 | uBR925# copy running-config startup-config Building configuration... | Save the configuration to nonvolatile memory so that it will not be lost in the event of a reset, power cycle, or power outage. |
| Step 15 | uBR925# show startup-config | Display the configuration file that was just created. |

The following configuration shows a Cisco uBR925 cable access router configured in DOCSIS-bridging mode that uses SGCP for the routing of its voice calls. The relevant commands are shown in bold.

```

version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname uBR925
!
!
clock timezone - 0 6
ip subnet-zero
no ip routing
ip domain-name cisco.com
ip name-server 4.0.0.32
!
sgcp
sgcp call-agent 10.186.1.36
!
xgcp snmp sgcp
!
!
voice-port 0
!
voice-port 1
!
dial-peer voice 100 pots
application SGCPAPP
destination-pattern 5551212
port 0
!
dial-peer voice 101 pots
application SGCPAPP
destination-pattern 5551213
port 1
!
process-max-time 200
!
interface Ethernet0
no ip directed-broadcast
no ip route-cache
no ip mroute-cache
bridge-group 59
bridge-group 59 spanning-disabled
!
interface cable-modem0
ip address docsis
no ip directed-broadcast
no ip route-cache

```

```
no ip mroute-cache
bridge-group 59
bridge-group 59 spanning-disabled
!
interface USB0
ip address negotiated
no ip directed-broadcast
no ip route-cache
no ip mroute-cache
arp timeout 0
bridge-group 59
bridge-group 59 spanning-disabled
!
!
ip classless
no ip http server
no service finger
!
!
line con 0
transport input none
line vty 0 4
login
!
end
```

MGCP Configuration

The Cisco uBR925 cable access router can use the MGCP protocol for routing voice calls. This transfers the dial mapping to an external call agent or to a Media Gateway Controller, so that the VoIP gateways do not have to be individually configured with the dial mappings.

**Note**

The Cisco uBR925 cable access router can use MGCP in either DOCSIS-bridging mode or routing mode.

You must do the following to configure the Cisco uBR925 cable access router for MGCP routing of voice calls:

- Enable MGCP operation on the Cisco uBR925 cable access router.
- Specify the MGCP call agent's IP address.
- Configure the local dial peers to be MGCP applications.
- Optionally specify the MGCP packages to be supported.
- Optionally change a number of MGCP parameters.

**Note**

No configuration of remote dial peers is needed when using MGCP.

These functions are done using the commands shown in the following table:

| | Command | Purpose |
|---------|---|--|
| Step 1 | To configure incoming calls on voice port V1: uBR925(config)# dial-peer voice <i>id-number</i> pots | Specify a unique <i>id-number</i> for this incoming dial peer and enter dial-peer configuration mode. |
| Step 2 | uBR925(config)# application MGCPAPP | Specify that this dial peer is handled as an MGCP application. |
| Step 3 | uBR925(config-dial-peer)# port 0 | Specify that voice port V1 is attached to this telephony equipment. |
| Step 4 | uBR925(config-dial-peer)# exit | Exit dial-peer configuration mode. |
| Step 5 | To configure incoming calls on voice port V2: uBR925(config)# dial-peer voice <i>id-number</i> pots | Specify a unique <i>id-number</i> for this incoming dial peer and enter dial-peer configuration mode. |
| Step 6 | uBR925(config)# application MGCPAPP | Specify that this dial peer is handled as an MGCP application. |
| Step 7 | uBR925(config-dial-peer)# port 1 | Specify that voice port V2 is attached to this telephony equipment. |
| Step 8 | uBR925(config-dial-peer)# exit | Exit dial-peer configuration mode. |
| Step 9 | uBR925(config)# mgcp | Enable MGCP operations on the router. |
| Step 10 | uBR925(config)# mgcp call-agent <i>ip-address</i> [<i>port</i>] [service-type <i>sgcp</i> <i>mgcp</i>] | Specify the IP address and optional UDP port number for the MGCP call-agent. If no port number is given, the default is 2427. The default service-type is mgcp , but sgcp can be specified to ignore RSIP error messages. |
| Step 11 | uBR925(config)# mgcp dtmf-relay { <i>codec</i> low-bit-rate } mode { <i>cisco</i> out-of-band } | (Optional) Enables the accurate forwarding of touchtone digits during a voice call. Use codec to specify the G.711 codec or low-bit-rate to specify the G.729 codec. Use a mode of cisco to transmit the tones with the Cisco proprietary method; if the remote gateway is not a Cisco router, use out-of-band instead. |
| Step 12 | uBR925(config)# mgcp ip-tos { high-reliability high-throughput low-cost low-delay precedence <i>value</i> } | (Optional) Enable IP Type of Services (TOS) for the voice connections, and specify the value for the IP precedence bit (the default IP precedence is 3). |
| Step 13 | uBR925(config)# mgcp max-waiting-delay <i>value</i> | (Optional) Specify the number of milliseconds to wait after a restart (default of 3000) before connecting with the call agent. If used, these values should be staggered among gateways to avoid having large numbers of gateways connecting with the call agent at the same time after a mass restart. |
| Step 14 | uBR925(config)# mgcp modem passthru { <i>cisco</i> <i>ca</i> } | (Optional) Enable the transmission and reception of modem and fax data. If the remote gateway is a Cisco router, specify cisco ; otherwise, specify ca (default) to allow the data to pass through the call-agent. |

| | Command | Purpose |
|---------|--|---|
| Step 15 | <code>uBR925(config)# mgcp package-capability { line-package dtmf-package gm-package rtp-package }</code> | (Optional) Specify that the Cisco uBR925 cable access router supports a particular package capability. Give this command multiple times to enable multiple packages. Use this command before using the mgcp default-package command. |
| Step 16 | <code>uBR925(config)# mgcp default-package { line-package dtmf-package gm-package }</code> | (Optional) Specify the default package type for the media gateway; defaults to line-package . |
| Step 17 | <code>uBR925(config)# mgcp playback { adaptive <i>init-value min-value max-value</i> fixed <i>init-value</i> }</code> | (Optional) Change the jitter buffer packet size in milliseconds for MGCP calls, using either an adaptive range or a fixed value. The default is adaptive 60 4 200 . |
| Step 18 | <code>uBR925(config)# mgcp request retries <i>count</i></code> | (Optional) Specify the number of times a call request message is transmitted to a call agent before timing out. The default is 3 times. |
| Step 19 | <code>uBR925(config)# mgcp request timeout <i>timeout</i></code> | (Optional) Specify the number of milliseconds to wait for a response to a request before retransmitting or timing out the request. The default is 500 milliseconds. |
| Step 20 | <code>uBR925(config)# mgcp restart-delay <i>value</i></code> | (Optional) Specify the value (in seconds) used in Restart in Progress (RSIP) messages to indicate the delay before the connection is torn down. The default delay is 0 seconds. |
| Step 21 | <code>uBR925(config)# mgcp vad</code> | (Optional) Enable Voice Activity Detection (VAD) to turn silence suppression on. The default disables VAD. |
| Step 22 | <code>uBR925# copy running-config startup-config</code> Building configuration... | Save the configuration to nonvolatile memory so that it will not be lost in the event of a reset, power cycle, or power outage. |
| Step 23 | <code>uBR925# show startup-config</code> | Display the configuration file that was just created. |

The following configuration shows a Cisco uBR925 cable access router configured in DOCSIS-bridging mode that uses MGCP for controlling its voice calls. The relevant commands are shown in bold.

```

version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname uBR925
!
!
clock timezone - 0 6
ip subnet-zero
no ip routing
ip domain-name cisco.com
ip name-server 10.0.0.32
!
mgcp
mgcp call-agent 10.186.1.36
mgcp modem passthru ca
mgcp package-capability dtmf-package

```

```

mgcp package-capability line-package
mgcp default-package line-package
!
xgcp snmp sgcp
!
!
voice-port 0
!
voice-port 1
!
dial-peer voice 100 pots
  application MGCPAPP
  port 0
!
dial-peer voice 101 pots
  application MGCPAPP
  port 1
!
process-max-time 200
!
interface Ethernet0
  no ip directed-broadcast
  no ip route-cache
  no ip mroute-cache
  bridge-group 59
  bridge-group 59 spanning-disabled
!
interface cable-modem0
  ip address docsis
  no ip directed-broadcast
  no ip route-cache
  no ip mroute-cache
  bridge-group 59
  bridge-group 59 spanning-disabled
!
interface USB0
  ip address negotiated
  no ip directed-broadcast
  no ip route-cache
  no ip mroute-cache
  arp timeout 0
  bridge-group 59
  bridge-group 59 spanning-disabled
!
ip classless
no ip http server
no service finger
!
!
line con 0
  transport input none
line vty 0 4
  login
!
end

```