



## Cisco IOS Voice Commands: **U**

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This chapter contains commands to configure and maintain Cisco IOS voice applications. The commands are presented in alphabetical order. Some commands required for configuring voice may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

For detailed information on how to configure these applications and features, refer to the *Cisco IOS Voice Configuration Guide*.

**■ unbundle vfc**

## unbundle vfc

To unbundle DSPWare from the VCWare and configure the default file and capability lists with default values, use the **unbundle vfc** command in privileged EXEC mode.

**unbundle [high-complexity | medium-complexity] vfc slot-number**

<b>Syntax Description</b>	<b>high-complexity</b> (Optional) Unbundles the high-complexity firmware set. <b>medium-complexity</b> (Optional) Unbundles the medium-complexity firmware set. <b>slot-number</b> Voice feature card (VFC) slot number.
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<b>Command Default</b>	No default behavior or values
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<b>Command Modes</b>	Privileged EXEC
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<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	11.3(2)NA	This command was introduced on Cisco AS5300.
	12.0(2)XH	The <b>high-complexity</b> and <b>medium-complexity</b> keywords were added.
	12.0(3)T	This command was integrated into Cisco IOS Release 12.0(3)T.

<b>Usage Guidelines</b>	VFCs come with a single bundled image, VCWare, stored in VFC Flash memory. Use the <b>unbundle vfc</b> command to unbundle this bundled image into separate files, which are then written to Flash memory. When VCWare is unbundled, it automatically adds DSPWare to Flash memory, creates both the capability and default file lists, and populates these lists with the default files for that version of VCWare. The default file list includes the files to be used to boot up the system. The capability list defines the available voice codecs for H.323 capability negotiation. These files are used during initial card configuration and for subsequent firmware upgrades.
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Before unbundling a VFC software image that you have just copied over to VFC Flash, use the **clear vfc** command. Unbundling a DSP firmware set rewrites the default-file and capabilities lists. After unbundling, you must reload the router for any changes to take effect.

<b>Examples</b>	The following example unbundles the high-complexity firmware set into slot 2:
	<pre>Router# unbundle high-complexity vfc 2</pre>

<b>Related Commands</b>	<b>Command</b>	<b>Description</b>
	<b>copy flash vfc</b>	Copies a new version of VCWare from the Cisco AS5300 motherboard to VFC Flash memory.
	<b>copy tftp vfc</b>	Copies a new version of VCWare from a TFTP server to VFC Flash memory.

## url

To configure the Internet service provider (ISP) address, use the **url** command in settlement configuration mode. You can configure the address type multiple times. To disable the address, use the **no** form of this command.

**url url-address**

**no url url-address**

<b>Syntax Description</b>	<i>url-address</i> URL address. A valid URL address is as follows: <i>http://fully qualified domain name[:port]/[URL]</i>
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<b>Command Default</b>	No default behavior or values
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<b>Command Modes</b>	Settlement configuration
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<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	12.0(4)XH1	This command was introduced on Cisco 2600 series and Cisco 3600 series, and Cisco AS5300.
	12.1(1)T	This command was integrated into Cisco IOS Release 12.1(1)T.
	12.2(11)T	The settlement configuration for this command was modified. The settlement provider must be shutdown before the <b>url</b> command is entered.

<b>Usage Guidelines</b>	You can configure the address type multiple times. If you configure multiple URLs for the settlement server, the gateway attempts to send the request to each URL in the order in which you configured these addresses.
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If the first URL is unsuccessful, the gateway tries the next URL. If the first URL becomes available, the gateway does not switch back until it loops through the list of URLs, for example:

```
url http://servicepoint1.com
url http://servicepoint2.com
url http://servicepoint3.com
```

If `http://servicepoint1.com` fails, the gateway sends the request to `http://servicepoint2.com`. If `http://servicepoint1.com` comes back online, the gateway continues to send requests to `http://servicepoint2.com`. Later on, if `http://servicepoint2` is down, the gateway sends requests to `http://servicepoint3.com`.

When `http://servicepoint3.com` is down the gateway routes its requests back to `http://servicepoint1.com`.

**Examples**

The following example shows four URLs configured for the settlement server:

```
settlement 0
url http://1.2.3.4/
url http://1.2.3.4:80/
url https://1.2.3.4:4444/
url https://yourcompany.com:443/
```

**Related Commands**

<b>Command</b>	<b>Description</b>
<b>connection-timeout</b>	Sets the connection timeout.
<b>customer-id</b>	Sets the customer identification.
<b>device-id</b>	Sets the device identification.
<b>encryption</b>	Specifies the encryption method.
<b>max-connection</b>	Sets the maximum simultaneous connections.
<b>response-timeout</b>	Sets the response timeout.
<b>retry-delay</b>	Sets the retry delay.
<b>retry-limit</b>	Sets the connection retry limit.
<b>session-timeout</b>	Sets the session timeout.
<b>settlement</b>	Enters settlement configuration mode.
<b>show settlement</b>	Displays the configuration for all settlement server transactions.
<b>shutdown/no shutdown</b>	Brings up the settlement provider and then shuts it down.
<b>type</b>	Specifies the provider type.

# url ( SIP)

To configure URLs to either the Session Initiation Protocol (SIP), SIP secure (SIPS), or telephone (TEL) format for your VoIP SIP calls, use the **url** command in SIP configuration mode. To reset to the default, use the **no** form of this command.

**url {sip | sips | tel [phone-context]}**

**no url**

Syntax Description	<b>sip</b> Generates URLs in SIP format for VoIP calls. <b>sips</b> Generates URLs in SIPS format for VoIP calls. <b>tel</b> Generates URLs in TEL format for VoIP calls. <b>phone-context</b> (Optional) Appends the phone-context parameter to the TEL URL.
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<b>Command Default</b>	SIP URLs
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<b>Command Modes</b>	SIP configuration (conf-serv-sip)
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Command History	Release	Modification
	12.2(2)XB	This command was introduced.
	12.2(2)XB1	This command was implemented on the Cisco AS5850.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T. Support for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 was not included in this release.
	12.2(11)T	This command was implemented on the Cisco AS5300, Cisco AS5350, and Cisco AS5400 platforms.
	12.4(6)T	The <b>sips</b> keyword was added.
	12.4(22)YB	The <b>phone-context</b> keyword was added.
	15.0(1)M	This command was integrated into Cisco IOS Release 15.0(1)M.

<b>Usage Guidelines</b>	This command affects only user-agent clients (UACs), because it causes the use of a SIP, SIPS, or TEL URL in the request line of outgoing SIP INVITE requests. SIP URLs indicate the originator, recipient, and destination of the SIP request; TEL URLs indicate voice call connections.
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The **voice-class sip url** command takes precedence over the **url** command configured in SIP global configuration mode. However, if the **voice-class sip url** command is configured with the **system** keyword, the gateway uses what was globally configured with the **url** command.

Enter SIP configuration mode after entering voice-service VoIP configuration mode, as shown in the “Examples” section.

**Examples**

The following example generates URLs in SIP format:

```
voice service voip
sip
  url sip
```

The following example generates URLs in SIPS format:

```
voice service voip
sip
  url sips
```

The following example generates URLs in TEL format:

```
voice service voip
sip
  url tel
```

The following example generates URLs in TEL format and appends the phone-context parameter:

```
voice service voip
sip
  url tel phone-context
```

**Related Commands**

<b>Command</b>	<b>Description</b>
<b>sip</b>	Enters SIP configuration mode from voice-service VoIP configuration mode.
<b>voice-class sip url</b>	Generates URLs in the SIP, SIPS, or TEL format.

# usage-indication

To enter the Annex G neighbor usage mode used to configure optional usage indicators, use the **usage-indication** command in Annex G neighbor configuration mode. To return to the default setting, use the **no** form of this command.

## **usage-indication**

## **no usage-indication**

<b>Syntax Description</b>	This command has no arguments or keywords.
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<b>Command Default</b>	Disabled
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<b>Command Modes</b>	Annex G neighbor
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<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	12.2(11)T	This command was introduced.

<b>Usage Guidelines</b>	Use <b>usage-indication</b> command to enter the mode to set usage indication characteristics. Repeat this command for each border element neighbor that you configure.
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The **no shutdown** command must be used to enable each service relationship.

<b>Examples</b>	The following example shows how to enter the Annex G neighbor usage mode:
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```
doc-rtr3 (config-nxg-neigh-usg)# usage-indication
```

<b>Related Commands</b>	<b>Command</b>	<b>Description</b>
	<b>access-policy</b>	Requires that a neighbor be explicitly configured.
	<b>inbound ttl</b>	Sets the inbound time-to-live value.
	<b>outbound retry-interval</b>	Defines the retry period for attempting to establish the outbound relationship between border elements.
	<b>retry interval</b>	Defines the time between delivery attempts.
	<b>retry window</b>	Defines for how long a border element will attempt delivery.
	<b>shutdown</b>	Enables or disables the border element.

**use-proxy**

## use-proxy

To enable proxy communications for calls between local and remote zones or the H.225 Annex G border element, use the **use-proxy** command in gatekeeper configuration mode. To remove either a proxy configuration entry for a remote zone or the H.225 Annex G border element, to disable proxy communications between local and remote zones or H.225 Annex G border element, use the **no** form of this command.

```
use-proxy local-zone-name {default | h323-annexg | remote-zone remote-zone-name}  
[inbound-to | outbound-from] {gateway | terminal}
```

```
no use-proxy local-zone-name {default | h323-annexg | remote-zone remote-zone-name}  
[inbound-to | outbound-from] {gateway | terminal}]
```

### Syntax Description

<i>local-zone-name</i>	Name or zone name of the gatekeeper, which is usually the fully domain-qualified host name of the gatekeeper.
<b>default</b>	Default proxy policy for all calls that are not defined by a <b>use-proxy</b> command with the <b>remote-zone</b> keyword or <b>h323-annexg</b> keyword.
<b>h323-annexg</b>	Proxy policy for calls to or from the H.225 Annex G border element co-located with the gatekeeper.
<b>remote-zone</b> <i>remote-zone-name</i>	Proxy policy for calls to or from a specific remote gatekeeper or zone.
<b>inbound-to</b>	Proxy policy as it applies to calls that are inbound to the local zone from a remote zone. Each <b>use-proxy</b> command defines the policy for only one direction.
<b>outbound-from</b>	Proxy policy as it applies to calls that are outbound from the local zone to a remote zone. Each <b>use-proxy</b> command defines the policy for only one direction.
<b>gateway</b>	Type of local device to which the policy applies. The <b>gateway</b> option applies the policy only to local gateways.
<b>terminal</b>	Type of local device to which the policy applies. The <b>terminal</b> option applies the policy only to local terminals.

### Command Default

The local zone uses proxy for both inbound and outbound calls to and from the local H.323 terminals only. Proxy is not used for both inbound and outbound calls to and from local gateways. For releases prior to Cisco IOS Release 12.3(7)T, both inbound and outbound calls using the H.225 Annex G border element do not use the proxy.

### Command Modes

Gatekeeper configuration

### Command History

Release	Modification
12.0(5)T	This command was introduced on the Cisco AS5300.
12.1(5)XM2	The command was implemented on the Cisco AS5350 and Cisco AS5400.

Release	Modification
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T. Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T.
12.3(7)T	The <b>h323-annexg</b> keyword was added.

**Usage Guidelines**

This command replaces the **zone access** command used in previous versions of the gatekeeper. When a previous version of a gatekeeper is upgraded, any **zone access** commands are translated to **use-proxy** commands. You can use the **show gatekeeper zone status** command to see the gatekeeper proxy configuration.

If the domain name is cisco.com, the gatekeeper name might be gk1.cisco.com. However, if the gatekeeper is controlling multiple zones, the name of the gatekeeper for each zone should be a unique string.

**Examples**

In the following example, the local zone sj.xyz.com is configured to use a proxy for inbound calls from remote zones tokyo.xyz.com and milan.xyz.com to gateways in its local zone. The sj.xyz.com zone is also configured to use a proxy for outbound calls from gateways in its local zone to remote zones tokyo.xyz.com and milan.xyz.com.

```
use-proxy sj.xyz.com remote-zone tokyo.xyz.com inbound-to gateway
use-proxy sj.xyz.com remote-zone tokyo.xyz.com outbound-from gateway
use-proxy sj.xyz.com remote-zone milan.xyz.com inbound-to gateway
use-proxy sj.xyz.com remote-zone milan.xyz.com outbound-from gateway
```

Because the default mode disables proxy communications for all gateway calls, only the gateway calls listed above can use the proxy.

In the following example, the local zone sj.xyz.com uses a proxy for only those calls that are outbound from H.323 terminals in its local zone to the specified remote zone germany.xyz.com:

```
no use-proxy sj.xyz.com default outbound-from terminal
use-proxy sj.xyz.com remote-zone germany.xyz.com outbound-from terminal
```



Any calls inbound to H.323 terminals in the local zone sj.xyz.com from the remote zone germany.xyz.com use the proxy because the default applies.

The following example removes one or more proxy statements for the remote zone germany.xyz.com from the proxy configuration list:

```
no use-proxy sj.xyz.com remote-zone germany.xyz.com
```

This command removes all special proxy configurations for the remote zone germany.xyz.com. After you enter a command like this, all calls between the local zone (sj.xyz.com) and germany.xyz.com are processed according to the defaults defined by any **use-proxy** commands that use the **default** option.

To prohibit proxy use for inbound calls to H.323 terminals in a local zone from a specified remote zone, enter a command similar to the following:

```
no use-proxy sj.xyz.com remote-zone germany.xyz.com inbound-to terminal
```

**use-proxy**

This command overrides the default and disables proxy use for inbound calls from remote zone `germany.xyz.com` to all H.323 terminals in the local zone `sj.xyz.com`.

In the following example, the local zone `sj.xyz.com` is configured to use a proxy for inbound calls and outbound calls that use the H.225 Annex G border element co-located with the gatekeeper:

```
use-proxy sj.xyz.com h323-annexg inbound-to gateway
use-proxy sj.xyz.com h323-annexg outbound-from gateway
```

In the following example, the local zone `sj.xyz.com` is configured not to use a proxy for inbound calls and outbound calls that use the H.225 Annex G border element co-located with the gatekeeper:

```
no use-proxy sj.xyz.com h323-annexg inbound-to terminal
no use-proxy sj.xyz.com h323-annexg outbound-from terminal
```

The following example removes one or more proxy statements for the H.225 Annex G border element from the proxy configuration list:

```
no use-proxy sj.xyz.com h323-annexg
```

Related Commands	Command	Description
	<b>show gatekeeper zone status</b>	Displays the status of zones related to a gatekeeper.

## user-id

To match a call based on the user-id field in the Session Initiation Protocol (SIP) uniform resource identifier (URI), use the **user-id** command in voice URI class configuration mode. To remove the match pattern, use the **no** form of this command.

**user-id** *username-pattern*

**no user-id**

<b>Syntax Description</b>	<i>username-pattern</i>	Cisco IOS regular expression pattern to match against the user-id field in a SIP URI. Can be up to 32 characters.
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<b>Command Default</b>	No default behavior or values
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<b>Command Modes</b>	Voice URI class configuration
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<b>Command History</b>	<b>Release</b>	<b>Modification</b>
	12.3(4)T	This command was introduced.

<b>Usage Guidelines</b>	<ul style="list-style-type: none"> <li>You can use this command only in a voice class for SIP URIs.</li> <li>You cannot use this command if you use the <b>pattern</b> command in the voice class. The <b>pattern</b> command matches on the entire URI, whereas this command matches only a specific field.</li> </ul>
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<b>Examples</b>	The following example defines a voice class that matches on the user-id field in a SIP URI:
	<pre>voice class uri r100 sip   user-id abc123</pre>

<b>Related Commands</b>	<b>Command</b>	<b>Description</b>
	<b>destination uri</b>	Specifies the voice class used to match the dial peer to the destination URI for an outgoing call.
	<b>host</b>	Matches a call based on the host field in a SIP URI.
	<b>incoming uri</b>	Specifies the voice class used to match a VoIP dial peer to the URI of an incoming call.
	<b>pattern</b>	Matches a call based on the entire SIP or TEL URI.
	<b>phone context</b>	Filters out URIs that do not contain a phone-context field that matches the configured pattern.

**■ user-id**

<b>Command</b>	<b>Description</b>
<b>voice class uri</b>	Creates or modifies a voice class for matching dial peers to calls containing a SIP or TEL URI.
<b>voice class uri sip preference</b>	Sets a preference for selecting voice classes for a SIP URI.