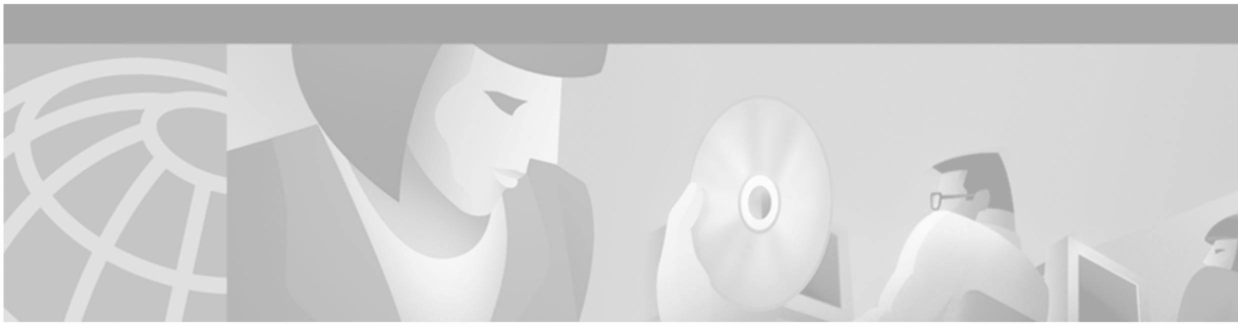

Enhanced Codec Support for SIP Using Dynamic Payloads

Document Update Alert

This document was originally produced for Cisco IOS Release 12.2(11)T. This feature has been updated in subsequent releases, and more recent documentation is available.

If you are using Cisco IOS Release 12.2(11)T or higher, refer to the following feature in the SIP Features Roadmap chapter of the *Cisco IOS SIP Configuration Guide*, Release 12.3:

- *Enhanced Codec Support for SIP Using Dynamic Payloads*

Feature History

Release	Modification
12.2(11)T	This feature was introduced on the Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850.

This document describes the Enhanced Codec Support for SIP Using Dynamic Payloads feature in Cisco IOS Release 12.2(11)T and includes the following sections:

Feature Overview

The Enhanced Codec Support for SIP Using Dynamic Payloads feature enhances codec selection and payload negotiation between originating and terminating Session Initiation Protocol (SIP) gateways. The new feature provides the SIP enhancements described in the following sections:

- Additional Codec Support
- Payload Type Selection
- Advertising Codec Capabilities

Additional Codec Support

Codecs are a digital signal processor (DSP) software algorithm used to compress or decompress speech or audio signals. Previous implementations of the SIP stack on Cisco IOS gateways supported only a subset of the available codecs for each platform. The Enhanced Codec Support for SIP Using Dynamic Payloads feature adds support for eight additional codecs:

- Clear-channel
- G723ar53
- G723ar63
- G723r53
- G726r16
- G726r24
- G729br8
- GSM-EFR

Support for these codecs varies on different platforms. See Table 1 for a listing of SIP codec support by platform, and by previous and current Cisco IOS release. Use the **codec ?** command to determine the codecs available on a specific platform.

Table 1 SIP Codec Support by Platform and Cisco IOS Release

Codecs	Cisco Series 2600, 3620, 3640, 3660 Previous/ Current Support	Cisco Series 7200 Previous/ Current Support	Cisco AS5300 Previous/ Current Support	Cisco AS5350, AS5400, AS5850 Previous/ Current Support
Clear-channel	Yes/Yes	No/No	No/Yes	No/Yes
G711alaw	Yes/Yes	Yes/Yes	Yes/Yes	Yes/Yes
G711ulaw	Yes/Yes	Yes/Yes	Yes/Yes	Yes/Yes
G723ar53	No/Yes	No/Yes	No/Yes	No/Yes
G723ar63	No/Yes	No/Yes	No/Yes	No/Yes
G723r53	No/Yes	No/Yes	No/Yes	No/Yes
G723r63	Yes/Yes	Yes/Yes	Yes/Yes	No/Yes
G726r16	No/Yes	No/Yes	No/Yes	No/Yes
G726r24	No/Yes	No/Yes	No/Yes	No/Yes
G726r32	Yes/Yes	Yes/Yes	Yes/Yes	No/Yes

Codecs	Cisco Series 2600, 3620, 3640, 3660 Previous/ Current Support	Cisco Series 7200 Previous/ Current Support	Cisco AS5300 Previous/ Current Support	Cisco AS5350, AS5400, AS5850 Previous/ Current Support
G728	Yes/Yes	Yes/Yes	Yes/Yes	No/No
G729br8	No/Yes	No/Yes	No/Yes	No/Yes
G729r8	Yes/Yes	Yes/Yes	Yes/Yes	Yes/Yes
GSM-EFR	No/Yes	No/No	No/Yes	No/No
GSM-FR	Yes/Yes	No/No	Yes/Yes	No/Yes

Payload Type Selection

Payload types define the content and format of Real-Time Transport Protocol (RTP) packets and the resulting stream of data generated by the RTP flow. The payload type defines the codec in use and is identified in the payload type field of the header of each RTP packet. There are two mechanisms for specifying payload type, static and dynamic.

Static payload types are assigned to specific RTP formats by RFC 1890 and these mappings are registered with the Internet Assigned Numbers Authority (IANA). Although not required, static payload types can also be mapped to RTP encodings using the rtpmap attribute. The following SIP supported codecs have static payloads values defined by the IANA:

- G711ulaw
- G711alaw
- G723r63
- G726r32
- G728
- G729r8
- GSM-FR

Dynamic payload values are used for codecs that do not have static payload values defined. Dynamic payload types do not have fixed mappings, and must be mapped to RTP encodings within the Session Description Protocol (SDP) itself using the **a=rtpmap:** line. The new feature allows dynamic payload values to be used for the following codecs with no static payload values defined:

- Clear-channel
- G726r16
- G726r24
- GSM-EFR

Of the four codecs listed that allow dynamic payload values to be assigned, only the payload type for the clear-channel codec can be configured using the command-line interface (CLI). The remaining G726r16, G726r24 and GSM-EFR codecs are selected on a per-call basis by the SIP subsystem. The dynamic payload range is assigned by the IANA, with values from 96 to 127. The SIP subsystem looks for and uses the first value in the range that is both available and not reserved for Cisco IOS applications. Once a dynamic payload value is picked for a particular payload type, it cannot be used for other payload types. Of the 32 available IANA values, those reserved for special Cisco IOS applications are listed in Table 2. To configure dynamic payload values for the payload types listed in Table 2, use the **rtp payload-type** command. Otherwise, the default payload values defined in Table 2 are used.

Table 2 Default Dynamic Payload Values

Dynamic Payload Type	Default Dynamic Payload Value	Supported by SIP
Cisco-rtp-dtmf-relay	121	Yes
Named Signal Event	100	Yes
Named Telephony Event	101	Yes
Cisco-cas-payload	123	No
Cisco-clear-channel	125	No
Cisco-codec-fax-ack	97	No
Cisco-codec-fax-ind	96	No
Cisco-fax-relay	122	No
Cisco-pcm-switch-over-alaw	127	No
Cisco-pcm-switch-over-ulaw	126	No

Note Once a dynamic payload value has been assigned from the reserved range, it cannot be used for any other payload types.

Advertising Codec Capabilities

The dynamic payload value selected by the SIP subsystem is advertised in the outgoing SIP INVITE request. The Enhanced Codec Support for SIP Using Dynamic Payloads feature supports dynamic payloads by expanding the SIP subsystem ability to advertise and negotiate available codecs. SIP uses the connection, media, and attribute fields of the SDP message during connection negotiation.

The new feature supports the following Internet Engineering Task Force (IETF) drafts:

- draft-ietf-avt-rtp-mime-06.txt, *MIME Type Registration of RTP Payload Formats* (further developed and later published as RFC 3555).
- draft-ietf-avt-profile-new-12.txt, *RTP Profile for Audio and Video Conferences with Minimal Control* (further developed and later published as RFC 3551).

The following sample SIP INVITE message shows the payload value and codec selection resulting from the payload negotiation process. The media m= field includes the added payload value. The attribute a= field includes the selected codec. In this outgoing INVITE message, the first available dynamic payload value of 115 is selected by the SIP subsystem for a GSM-EFR codec.

```
INVITE sip:36602@172.18.193.120:5060;user=phone SIP/2.0
    Via: SIP/2.0/UDP 172.18.193.98:5060
    From: "36601" <sip:36601@172.18.193.98>
    To: <sip:36602@172.18.193.120;user=phone>
    Date: Mon, 01 Mar 1993 00:05:14 GMT
    Call-ID: 4326879A-14EF11CC-80069792-19DC655A@172.18.193.98
    Cisco-Guid: 1092278192-351211980-2147784594-433874266
    User-Agent: Cisco-SIPGateway/IOS-12.x
    CSeq: 101 INVITE
    Max-Forwards: 6
    Timestamp: 730944314
    Contact: <sip:36601@172.18.193.98:5060;user=phone>
```

```
Expires: 180
Content-Type: application/sdp
  Content-Length: 228

v=0
o=CiscoSystemsSIP-GW-UserAgent 6973 8772 IN IP4 172.18.193.98
s=SIP Call
c=IN IP4 172.18.193.98
t=0 0
m=audio 17928 RTP/AVP 18 115
a=rtpmap:18 G729/8000
a=rtpmap:115 GSM-EFR/8000
```

G723 Codec Versions

In addition to the previously supported G723r63 version of the G723 codec, the new feature supports the following new versions:

- G723r53, where the number 53 indicates the bit rate of 5.3 kbps
- G723ar53, where the letter a indicates support for Annex A, which specifies voice activity detection (VAD)
- G723ar63, where the number 63 indicates a bit rate of 6.3 kbps

A static payload value of 4 is used for all versions of the G723 codec.

Expanded codec support allows the originating and terminating gateways to advertise and negotiate additional codec capabilities. Cisco implements support for multiple G723 codec versions by using a=fmtp and a=rtpmap attributes in the SDP body of outgoing INVITE requests to define the G723 codec version. For the G723 codec, the value of a=fmtp is 4 (the IANA assigned static value), and the annexa value is either yes or no. The default for annexa is yes.

Table 3 lists the possible codec configurations, that, taken together with Annex A support at the remote end, result in selecting the negotiated codec.

Table 3 G723 Codecs

Configured Codec(s)	Remote End Supports Annex A	Negotiated Codec
G723r63	annexa = no or no fmtp line	G723r63
G723r53	annexa = no or no fmtp line	G7223r53
G723r53 and G723r63	annexa = no or no fmtp line	G723r63
G723ar63	annexa=yes or no fmtp line	G723ar63
G723ar53	annexa=yes or no fmtp line	G723ar53
G723ar53 and G723ar63	annexa=yes or no fmtp line	G723ar63
G723ar53 and G723r53	annexa=yes or no fmtp line	G723ar53
G723ar63 and G723r63	annexa=yes or no fmtp line	G723ar63
G723ar63 and G723r53	annexa=yes or no fmtp line	G723ar63
G723ar53 and G723r63	annexa=yes or no fmtp line	G723ar63
G723ar53, G723r53, G723ar63, and G723r63	annexa = no or no fmtp line	G723ar63

The following partial SDP body shows the media m= field and attribute a= field for a gateway with G723 codecs and Annex A specified.

m=audio 62986 RTP/AVP 4
a=rtpmap:4 G723/8000
a=fmtp:4 annexa=yes

G729 Codec Versions

The new feature supports the following versions of G729 codecs:

- G729r8, where r8 indicates the bit rate of 8 kbps
- G729br8, where b indicates support for Annex B, which specifies VAD, Discontinuity Transmission (DTX), and Comfort Noise generation (CNG).

A static payload value of 18 is used for all versions of the G729 codec.

Cisco implements support for multiple G729 codec versions by using a=fmtp and a=rtpmap attributes in the SDP body of outgoing INVITE requests. For the G729 codec, the value of a=fmtp is 18 (the IANA assigned static value), and the annexb value is either yes or no. The default for annexb is yes.

Table 4 lists the possible codec configuration that, taken together with Annex B support at the remote end, result in selecting the negotiated codec.

Table 4 G729 Codecs

Configured Codec(s)	Remote End Supports Annex B	Negotiated Codec
G729r8	annexb= no or no fmtp line	G729r8
G729br8	annexb = yes or no fmtp line	G729br8
G729r8 and G729br8	annexb= yes or no fmtp line	G729br8
G729r8 and G729br8	no fmtp line	G729br8
G729r8 and G729br8	annexb=no or no fmtp line	G729r8
G729r8 and G729br8	annexb=yes	G729br8

The following partial SDP body shows the media m= field and attribute a= field for a gateway with G729 codecs and Annex B specified:

m=audio 17928 RTP/AVP 18
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no

Benefits

- Expanded dynamic payload support on Cisco IOS gateways, resulting in enhanced bandwidth control
- Expanded ability to advertise and negotiate all codecs available on a given platform
- Expanded interoperability and interconnectivity between gateways, applications, and services in the network

Restrictions

Dynamic payload values can be configured using the **rtp payload-type** command only for the payload types listed in Table 2; dynamic payloads cannot be configured for the codecs shown in Table 1.

Related Features and Technologies

- Cisco VoIP
- Cisco SIP proxy server
- Cisco IP Phones

Related Documents

- RFC 2543, *SIP: Session Initiation Protocol*
- RFC 2833, *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*
- *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2
- *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2
- *Cisco SIP Proxy Server Administrator Guide,* Version 1.2, "Configuring the Cisco SIP Proxy Server" chapter
- *Cisco IOS IP Configuration Guide*, Release 12.2
- *RTP Parameters*
- draft-ietf-avt-rtp-mime-06.txt, *MIME Type Registration of RTP Payload Formats* (further developed and later published as RFC 3555).
- draft-ietf-avt-profile-new-12.txt, *RTP Profile for Audio and Video Conferences with Minimal Control* (further developed and later published as RFC 3551).

Supported Platforms

- Cisco 2600 series
- Cisco 3600 series
- Cisco AS5300
- Cisco AS5350
- Cisco AS5400
- Cisco AS5850
- Cisco 7200 series

Determining Platform Support Through Cisco Feature Navigator

Cisco IOS software is packaged in feature sets that are supported on specific platforms. To get updated information regarding platform support for this feature, access Cisco Feature Navigator. Cisco Feature Navigator dynamically updates the list of supported platforms as new platform support is added for the feature.

Cisco Feature Navigator is a web-based tool that enables you to quickly determine which Cisco IOS software images support a specific set of features and which features are supported in a specific Cisco IOS image. You can search by feature or release. Under the release section, you can compare releases side by side to display both the features unique to each software release and the features in common.

To access Cisco Feature Navigator, you must have an account on Cisco.com. If you have forgotten or lost your account information, send a blank e-mail to cco-locksmith@cisco.com. An automatic check will verify that your e-mail address is registered with Cisco.com. If the check is successful, account details with a new random password will be e-mailed to you. Qualified users can establish an account on Cisco.com by following the directions found at this URL:
http://www.cisco.com/register

Cisco Feature Navigator is updated regularly when major Cisco IOS software releases and technology releases occur. For the most current information, go to the Cisco Feature Navigator home page at the following URL:
http://www.cisco.com/go/fn

Availability of Cisco IOS Software Images

Platform support for particular Cisco IOS software releases is dependent on the availability of the software images for those platforms. Software images for some platforms may be deferred, delayed, or changed without prior notice. For updated information about platform support and availability of software images for each Cisco IOS software release, refer to the online release notes or, if supported, Cisco Feature Navigator.

Supported Standards, MIBs, and RFCs

Standards

- draft-ietf-avt-rtp-mime-06.txt, *MIME Type Registration of RTP Payload Formats* (further developed and later published as RFC 3555).
- draft-ietf-avt-profile-new-12.txt, *RTP Profile for Audio and Video Conferences with Minimal Control* (further developed and later published as RFC 3551).

MIBs

No new or modified MIBs are supported by this feature.

To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL:
http://tools.cisco.com/ITDIT/MIBS/servlet/index

If Cisco MIB Locator does not support the MIB information that you need, you can also obtain a list of supported MIBs and download MIBs from the Cisco MIBs page at the following URL:
http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml

To access Cisco MIB Locator, you must have an account on Cisco.com. If you have forgotten or lost your account information, send a blank e-mail to cco-locksmith@cisco.com. An automatic check will verify that your e-mail address is registered with Cisco.com. If the check is successful, account details with a new random password will be e-mailed to you. Qualified users can establish an account on Cisco.com by following the directions found at this URL:
http://www.cisco.com/register

RFCs

No new or modified RFCs are supported by this feature.

Prerequisites

The following are general prerequisites for SIP functionality:

- Ensure that your Cisco 2600 series, Cisco 3600 series, or Cisco 7200 series router has at least 16 MB Flash memory and 64 MB DRAM. A Cisco AS5300 must have a minimum of 16 MB Flash memory and 128 MB DRAM. A Cisco AS5400 must have a minimum of 32 MB Flash memory and 256 MB DRAM.
- Ensure that the gateway has voice functionality that is configured for SIP.

  For more information about configuring SIP, refer to *Cisco IOS Voice, Video, and Fax Configuration Guide,* Release 12.2, "Configuring SIP for VoIP" chapter.
- Establish a working IP network.

  For more information about configuring IP, refer to *Cisco IOS IP Configuration Guide*, Release 12.2.
- Configure VoIP.

  For more information about configuring VoIP, refer to *Session Initiation Protocol for VoIP on Cisco Access Platforms.*

Configuration Tasks

The new feature is enabled by default. See the following sections for configuration tasks for the new feature. Each task in the list is identified as either required or optional:

- Configuring Dynamic Payload Values (optional)
- Verifying Dynamic Payload Values (optional)

Configuring Dynamic Payload Values

This procedure is optional and selects a dynamic payload value from the IANA defined range of 96 - 127. To modify the payload value for a dynamic payload type, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)#` **`dial-peer voip`** *`number`*	Enters dial peer configuration mode for a VoIP peer.
Step 2	`Router(config-dial-peer)#` **`rtp payload-type`** *`type number`*	Configures the dynamic payload type and value.
Step 3	`Router(config-dial-peer)#` **`exit`**	Exits dial peer configuration mode.

Verifying Dynamic Payload Values

This section describes the process for verifying dynamic payload value configuration. The **show running-config** command is used to display dynamic payload configuration. In the following partial output a dynamic payload value of 115 is configured, freeing up the reserved value of 101:

```
router# show running-config

Building configuration...
Current configuration: 2024 bytes
```

```
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
hostname r4
ip subnet-zero
ip tcp synwait-time 5
no ip domain-lookup
ipx routing 0000.0000.0004
no voice hpi capture buffer
no voice hpi capture destination
fax interface-type fax-mail
mta receive maximum-recipients 0
interface Loopback0
 ip address 10.0.0.0 255.255.255.0
interface FastEthernet0/0
 ip address 10.0.0.1 255.255.255.0
 speed 100
 full-duplex
interface Serial0/0
 ip address 10.0.0.4 255.255.255.0
 encapsulation frame-relay
.
.
.
call rsvp-sync
voice-port 3/0/0
voice-port 3/0/1
mgcp ip qos dscp cs5 media
mgcp ip qos dscp cs3 signaling
no mgcp timer receive-rtcp
mgcp profile default
dial-peer cor custom
dial-peer voice 1234 voip
 rtp payload-type nte 115
alias exec co config t
alias exec br show ip int brief
alias exec i show ip route
alias exec sr show run
alias exec sri sh run interface
alias exec sio show ip ospf
alias exec sioi show ip ospf int
alias exec sion show ip ospf nei
alias exec cir clear ip route *
alias exec ix show ipx route
alias exec b show ip bgp
alias exec sis show isdn status
alias exec fm show frame map
alias exec dm show dialer map
line con 0
 exec-timeout 0 0
 privilege level 15
 password ccie
 logging synchronous
line aux 0
line vty 0 4
 exec-timeout 0 0
 privilege level 15
 password ccie
 logging synchronous
 no login
end
```

Configuration Examples

This section provides the following configuration example:

- RTP Payload Named Telephony Event Example

RTP Payload Named Telephony Event Example

To configure a dynamic payload value, use the following commands beginning in global configuration mode. This procedure is optional and changes the default dynamic payload value assigned from the reserved range for a dynamic payload type.

```
router(config)# dial-peer voip 1
router(config)# rtp payload-type nte 115
router(config)# exit
```

Command Reference

There are no new commands for this feature. All commands used with this feature are documented in the Cisco IOS Release 12.2 command reference publications.

Glossary

CLI—command-line interface.

codec—coder-decoder. Transforms analog signals into a digital bit stream, and digital signals back into analog signals. In VoIP applications, it specifies the voice coder rate of speech for a dial peer.

gateway—A gateway allows SIP terminals to communicate with terminals configured to other protocols, by converting protocols. A gateway is the point at which a circuit-switched call is encoded and repackaged into IP packets.

INVITE—A method that initiates a session. It indicates that a user is invited to participate, provides a session description, indicates the type of media, and provides insight regarding the capabilities of the called and calling parties.

Payload type—A field within the fixed header portion of the RTP packet that defines the content, format, and encoding scheme of the RTP payload. When a SIP call is set up, the payload type values that are used in this field are listed in the m= line of the session description. These values can be either statically assigned to encoding names, or dynamically assigned within the session description. If a payload type is dynamically assigned, the session description will also include an rtpmap attribute that maps the payload type number to an encoding name. The encoding name identifies the format of the RTP packets (for example, a codec or telephone event).

RTP—Real-Time Transport Protocol.

SDP—Session Description Protocol.

SIP—Session Initiation Protocol. An application-layer protocol originally developed by the Multiparty Multimedia Session Control (MMUSIC) working group of the Internet Engineering Task Force (IETF). Their goal was to equip platforms to signal the setup of voice and multimedia calls over IP networks. SIP features are compliant with IETF RFC 2543, published in March 1999.

VoIP—Voice over IP. The ability to carry normal telephone-style voice over an IP-basednetwork.