Understanding Codecs: Complexity, Hardware Support, MOS, and Negotiation

Contents

Introduction

This document provides an overview to the different coder-decoders (codecs) used with Cisco IOS® Voice over IP (VoIP) gateways. Prior to Cisco IOS Software Release 12.0(5)T, VoIP gateways only supported the G.729 and G.711 codecs and only one voice/fax-relay call per digital signal processor (DSP). With the introduction of Cisco IOS 12.0(5)T, Cisco VoIP gateways support a larger number of codecs and DSP modules. They can also support up to four voice/fax-relay calls per DSP.

For more information on DSPs, refer to the Voice Hardware: C542 and C549 Digital Signal Processor (DSP) document.

Prerequisites

Requirements

There are no specific requirements for this document.

Components Used

This document is not restricted to specific software and hardware versions.

Conventions

For more information on document conventions, see the Cisco Technical Tips Conventions.

Codec Complexity

Some codec compression techniques require more processing power than others. Codec complexity is broken into two categories, medium and high complexity.

- Medium complexity allows the C549 DSPs to process up to four voice/fax-relay calls per DSP.
- High complexity allows the C549 DSPs to process up to two voice/fax-relay calls per DSP.

Medium Complexity (4 calls / dsp)	High Complexity (2 calls / dsp)
G.711 (a-law and m -law)	G.728
G.726 (all versions)	G.723 (all versions)
G.729a, G.729ab (G.729a AnnexB)	G.729, G.729b (G.729-AnnexB)
Fax-relay	Fax-relay
	Medium Complexity codecs (see Note 1)

Note: The difference between medium and high complexity codecs is the amount of CPU utilization necessary to process the codec algorithm, and therefore, the number of voice channels that can be supported by a single DSP. For this reason, all the medium complexity codecs can also be run in high complexity mode, but fewer (usually half) of the channels are available per DSP.

Note: Fax-relay (2400 bps, 4800 bps, 7200 bps, 9600 bps, 12 kbps, and 14.4 kbps) can use medium or high complexity codecs.

On platforms which support the C549 DSP technology, the codec complexity is configured under the voice-card (for example, the 2600/3600/VG-200 High Density Voice Network Module). Some platforms only support high complexity because they have enough DSPs onboard to support all T1/E1 channels using the high complexity mode.

An example of the complexity configuration is shown below:

```
Cisco-router #config t
Enter configuration commands, one per line. End with CNTL/Z.
Cisco-router(config)#voice-card 1
Cisco-router(config-voicecard)#codec complexity ?
        high    Set codec complexity high. High complexity, lower call density.
        medium  Set codec complexity medium. Mid range complexity and call density.
        <cr>
Cisco-router(config-voicecard)#codec complexity high
```

The following is an excerpt from the **show running-config** output to determine which complexity is configured:

```
!voice-card 1
  codec complexity high
!
```

Codec	1750	26xx/36xx NM-1V/2V	26xx/36xx NM-HDV	3810	AS5300 AS5800	AS5350 AS5400	7200	7500
G.711 a-law & u-law PCM (64 kbps)	12.0.5XQ1	Yes	12.0.5XK1	12.0.7XK	Yes	Yes	12.0.5XE3	12.1.3T
G.726 ADPCM (32, 24,16 kbps)	12.1.2T	12.0.5T	12.0.5XK1	12.0.7XK	Yes	No	12.0.5XE3	12.1.3T

G.728 LD-CELP (16 kbps)	No	12.0.5T	12.0.5XK1	12.0.7XK[7]	Yes	No	12.0.5XE3	12.1.3T
G.729 CS-ACELP (8 kbps)	12.1.2T	Yes	12.0.5XK1	12.0.7XK	Yes	No	12.0.5XE3	12.1.3T
G.729a CS-ACELP (8 kbps)	12.0.5XQ1	Negotiated (see Note 4)	12.0.5XK1	12.0.7XK	Yes	Yes	12.0.5XE3	12.1.3T
G.729 Annex-B (8 kbps) [VAD]	No	12.0.5T	12.0.5XK1	12.0.7XK[7]	Yes	No	12.0.5XE3	12.1.3T
G.729a Annex-B (8 kbps)	No	Negotiated (see Note 4)	12.0.5XK1	12.0.7XK[7]	Yes	Yes	12.0.5XE3	12.1.3T
G.723.1 MP-MLQ (6.3 kbps)	12.1.2T	12.0.5T	12.0.5XK1	12.0.7XK[7]	Yes	Yes	12.0.5XE3	12.1.3T
G.723.1 ACELP (5.3 kbps)	12.1.2T	12.0.5T	12.0.5XK1	12.0.7XK[7]	Yes	Yes	12.0.5XE3	12.1.3T
G.723.1 Annex-A MP-MLQ (6.3 kbps)	12.1.2T	12.0.5T	12.0.5XK1	12.0.7XK[7]	Yes	Yes	12.0.5XE3	12.1.3T
G.723.1 Annex-A ACELP (5.3 kbps)	12.1.2T	12.0.5T	12.0.5XK1	12.0.7XK[7]	Yes	Yes	12.0.5XE3	12.1.3T

Codec Compression Method
PCM = Pulse Code Modulation
ADPCM = Adaptive Differential Pulse Code Modulation
LDCELP = Low-Delay Code Excited Linear Prediction
CS-ACLEP = Conjugate-Structure Algebraic-Code-Excited Linear-Prediction
MP-MLQ = Multi-Pulse, Multi-Level Quantization
ACELP = Algebraic Code Excited Linear Prediction

Codec Mean Opinion Score (MOS)

Each codec provides a certain quality of speech. The quality of transmitted speech is a subjective response of the listener. A common benchmark used to determine the quality of sound produced by specific codecs is the mean opinion score (MOS). With MOS, a wide range of listeners judge the quality of a voice sample (corresponding to a particular codec) on a scale of 1 (bad) to 5 (excellent). The scores are averaged to provide the MOS for that sample. The following table shows the relationship between codecs and MOS scores.

Compression Method	Bit Rate (kbps)	MOS Score	Compression Delay (ms)

G.711 PCM	64	4.1	0.75
G.726 ADPCM	32	3.85	1
G.728 LD-CELP	16	3.61	3 to 5
G.729 CS-ACELP	8	3.92	10
G.729 x 2 Encodings	8	3.27	10
G.729 x 3 Encodings	8	2.68	10
G.729a CS-ACELP	8	3.7	10
G.723.1 MP-MLQ	6.3	3.9	30
G.723.1 ACELP	5.3	3.65	30

Although it might seem logical from a financial standpoint to convert all calls to low-bit rate codecs to save on infrastructure costs, you should exercise additional care when designing voice networks with low-bit rate compression. There are drawbacks to compressing voice. One of the main drawbacks is signal distortion due to multiple encodings (called tandem encodings). For example, when a G.729 voice signal is tandem encoded three times, the MOS score drops from 3.92 (very good) to 2.68 (unacceptable). Another drawback is codec-induced delay with low bit-rate codecs.

G.729 Codec Issues

The following two sections clarify many of the common compatibility issues concerning the G.729 (8 kbps) codec implementation.

Cisco pre-IETF G.729 and Standardized G.729 implementation

Cisco released a G.729 pre-Internet Engineering Task Force (IETF) codec implementation before the G.729 codec was standardized. Beginning with Cisco IOS 12.0(5)T, the default bit-ordering of the G.729 codec is changed from the pre-IETF standard to the IETF standardized format. The two formats do not interoperate, and in fact, will result in an unintelligible "gulping sound" to the end-users.

For compatibility with other vendor's G.729 implementations, Cisco IOS 12.0.5T and later releases default to the standardized implementation of G.729. For backwards compatibility with Cisco software releases prior to Cisco IOS 12.0.5T , enable the pre-IETF G.729 implementation with the following command:

```
maui-vgw-01(config)#dial-peer voice 100 voip
maui-vgw-01(config-dial-peer)#codec g729r8 pre-ietf
```

High Complexity: G.729, G729 Annex-B & Medium Complexity: G.729A, G.729A Annex-B

G.729 is a high complexity algorithm, and G.729A (also known as G.729 Annex-A) is a medium complexity variant of G.729 with slightly lower voice quality. All platforms that support G.729 also support G.729A.

On Cisco IOS gateways, the variant to use (G.729 or G.729A) is related to the codec complexity configuration on the voice card, and it does not show up explicitly in the Cisco IOS command line interface (CLI) codec choice. For example, the CLI will not show g729ar8 ("a" code) as a codec option, but if the voice-card is defined as medium-complexity, then the **g729r8** option is the G.729A codec.

Note: For the MC3810, prior to Cisco IOS Software Release 12.0.7XK, there is an explicit CLI choice between 24 channels of G729A or 12 channels of G729.

G.729 Annex-B is a high complexity algorithm, and G.729A Annex-B is a medium complexity variant of G.729 Annex-B with slightly lower voice quality. The difference between the G.729 and G.729 Annex-B codec is that the G.729 Annex-B codec provides built-in IETF voice activity detection (VAD) and Comfort Noise Generation (CNG).

The following G.729 codec combinations interoperate:

- G.729 and G.729A
- G.729 and G.729
- G.729A and G.729A
- G.729 Annex-B and G.729A Annex-B
- G.729 Annex-B and G.729 Annex-B
- G.729A Annex-B and G.729A Annex-B

Note: There is no explicit way to configure G.729A on the Cisco 2600/3600/VG-200 NM-1V and NM-2V (voice network module) since these voice modules do not support the "codec complexity" configuration supported on the NM-HDV (High Density Voice Network Module). However, if a G.729A call is set up by another endpoint terminating on the NM-1V/2V, the call will be successfully connected.

G.723.1 Codec Issues

There are two versions of G.723.1, Annex-A and non Annex-A. These versions do not interoperate. G.723.1 Annex-A includes a built-in IETF VAD algorithm and CNG.

Also, beginning with Cisco IOS 12.0(5)T, the G.723.1 codec is supported with a 5.3 kbps and 6.3 kbps rate. When a Cisco VoIP gateway sets up a call between devices using G723.1, it is only concerned that the far-end is using G.723.1. Neither side is concerned with the 5.3 kbps or 6.3 kbps rate that is supported by the other side. This means that, while it is beneficial to have both sides support the same rate, it is possible that one side will transmit at 5.3 kbps and the reverse direction will transmit at 6.3 kbps. The speed being used is viewed with the **show call active voice brief** command as shown below:

```
Cisco-router# show call active voice brief
47 : 494514hs.1 +473 pid:0 Answer active
tx:210/5040 rx:219/4380
IP 5.5.0.1:16534 rtt:3ms pl:890/0ms lost:0/0/0 delay:70/70/70ms g723r63
47 : 494514hs.2 +473 pid:1 Originate 4750001 active
  TX:230/1840 rx:230/8280
  Tele 2/0:0 (35): TX:6870/2290/0ms g723r63

  !--- In this example the G.723.1 is operating at 6.3 kbps

  noise:0 acom:0 i/0:-79/-5 dBm
```

The G.723.1 standard will allow stations to change rates between 6.3 kbps and 5.3 kbps during a call to adjust to network traffic loads. The Cisco VoIP gateways do not support this functionality, but they understand if the remote device (such as a Cisco IP Phone) transmits at a different rate than was originally negotiated.

The following G.723.1 codec combinations interoperate:

- G.723.1 (5.3 kbps) and G.723.1 (6.3 kbps)
- G.723.1 (5.3 kbps) and G.723.1 (5.3 kbps)
- G.723.1 (6.3 kbps) and G.723.1 (6.3 kbps)
- G.723.1 Annex-A (5.3 kbps) and G.723.1 Annex-A (6.3 kbps)
- G.723.1 Annex-A (5.3 kbps) and G.723.1 Annex-A (5.3 kbps)
- G.723.1 Annex-A (6.3 kbps) and G.723.1 Annex-A (6.3 kbps)

Codec Negotiation

With the introduction of 12.0(5)T, Cisco VoIP gateways support the codec negotiation feature. This feature provides the ability for a Cisco VoIP gateway to connect to other VoIP devices without necessarily knowing which codec is used for a call-setup. Also, this feature allows Cisco VoIP gateways to dynamically adjust to changes on remote devices. As long as the codec used by the remote VoIP device matches the capabilities-list of the Cisco VoIP gateway, the VoIP call is completed. Codec negotiation is supported on both the C542 and C549 DSPs.

The following example shows how to configure codec negotiation:

```
Cisco-router# config t
Cisco-router(config)# voice class codec 1

!--- This sets up class 1 to be assigned to the dial peer.

Cisco-router(config-class)#codec preference 1 g723r63
Cisco-router(config-class)#codec preference 2 g729ar8
Cisco-router(config-class)#codec preference 3 g711ulaw
Cisco-router(config-class)#codec preference 4 g726r32 bytes 240

!--- These commands define the preferred codec list using 1,2,3, and 4 to set the
preference.


Cisco-router(config)#dial-peer voice 1 voip
Cisco-router(config-dial-peer)#voice-class codec 1

!--- This assigns voice-class codec 1 to the dial-peer

Cisco-router(config-dial-peer)#destination-pattern 4723155
Cisco-router(config-dial-peer)#session target ipv4:192.168.100.1
```

Related Error Messages

%DSPRM-5-SETCODEC:

The **%DSPRM-5-SETCODEC** error is caused by having a high complexity codec configured on a VoIP dial-peer while still having the voice card set for the default of medium complexity. To fix this problem, you must remove the ds0-group configuration from the controller which will cause the voice-port to be removed. After you have removed the ds0-group, follow the procedures earlier in this document to change the complexity.

Related Information

- **Voice Hardware: C542 and C549 Digital Signal Processor (DSP)s**
- **Voice over IP - Per Call Bandwidth Consumption**
- **Voice Technologies**
- **Voice, Telephony and Messaging Devices**
- **Voice Software**
- **Technical Support - Cisco Systems**

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