Cisco Voice Codec Algorithms

A codec is a coder/decoder algorithm or a compression/decompression algorithm. Codecs are used to encode/decode or compress/decompress various t[redacted] *data that would otherwise use up large amounts of disk space, such as sound and video files. Cisco Unity uses audio codecs with streaming (live conv*[redacted] *content and file-based (WAV file voice message) content.*

Audio codecs supported for use with Cisco Unity:

G.711 Mu-Law (*the default codec*), G.711 A-Law

The G.711 codec is the standard format for digital voice delivery in the public switched telephone network (PSTN) and through PBXes. It is widely used in the teleco[redacted] the signal-to-noise ratio without increasing the amount of data.
There are two subsets of the G.711 codec: Mu-Law and A-Law. Mu-Law is used in North American and Japanese phone networks, while A-Law is used in E[redacted] Law and A-Law subsets use compressed speech carried in *8-bit samples*. They use an 8-kHz sampling rate with **64 Kbps storage**. Pulse code modulati[redacted]

G.726

The G.726 ADPCM voice codec, is used in many applications that require high-quality, robust speech reproduction, such as video conferenci[redacted] communications. The source code for the G.726 codec used by Cisco Unity is released by Sun Microsystems to the public domain.
In Cisco Unity version 4.x with Microsoft Exchange 2000 and in Cisco Unity version 4.0(5) and later with IBM Lotus Domino, the G.72[redacted] 32 Kbps is the format required by the VPIM version-2 specification and is supported by all VPIM-compliant voice messaging syste[redacted] version 4.0(5) and later installations and upgrades.

G.729a

The G.729 codec is a high-complexity algorithm, and G.729a (Annex A) is a medium-complexity variant of G.729[redacted] The advantage of G.729a is that it uses fewer digital signal processor (DSP) resources, making it easier to im[redacted] rate with 8 Kbps storage. If calls go through a *WAN link with limited bandwidth*, a codec with voice compres[redacted]

Intel Dialogic OKI ADPCM 8 kHz / 6 kHz

The adaptive differential pulse code modulation (ADPCM) method of encoding sound data file[redacted] encodes the difference between actual audio sample amplitude and predicted amplitude, a[redacted] Many versions of ADPCM exist. Cisco Unity supports two options for Intel Dialogic OKI[redacted] OKI ADPCM is a *hardware-based compression* method that is used by Intel Dialogi[redacted] However, when used with an IP phone system integration, ADPCM compression[redacted]

GSM 6.10

The Groupe Speciale Mobile (GSM), or Global System for Mobile C[redacted] the European digital cellular phone network to make the most of[redacted]

GSM 6.10 uses an 8-kHz sampling rate with 13 Kbps sto[redacted]

Stored voice messages can consume considera[redacted] storage a WAV file uses depends on what ki[redacted]

Audio Codec	Approximate File Si[redacted]
G.711 Mu/A-Law	480 KB
G.726 32 Kbps	240 KB
OKI ADPCM 8 kHz	240 KB
OKI ADPCM 6 kHz	180 K[redacted]
GSM 6.10	98 [redacted]
G.729a	[redacted]

ADPCM, com[redacted] waveform[redacted] the dis[redacted] AD[redacted] frequencies are properly reproduced, but any high frequencies tend to get distorted.