

What is CODEC?

Audio (Voice) Compression Protocols(CODEC)

With respect to voice over IP, a codec is an algorithm used to encode and decode the voice conversation. Since voice and sound as we hear it is analogue, it needs to be converted (or encoded) to a digital format suitable for transmission over the Internet. Once at the other end, it needs to be decoded again so the other person can hear what you are saying. There are a variety of different ways this encoding and decoding can be done - many of which utilize compression in order to reduce the required bandwidth of the conversation. A key thing to remember with VOIP, is that encoding, particularly when heavy compression is used, takes time, which adds a delay to the conversation. Thus, its important to utilize a codec which not only maintains good quality with compression, but is able to do the encoding and decoding in a minimal amount of time.

What Bandwidth is Required for Each Voice Call?

The bandwidth used varies depending on the compression method chosen.

IP Office supports a wide range of compression standards, including the most popular G.723.1 and G.729a. These will occupy approximately 10K and 13K of bandwidth respectively.

Use the following chart to choose the most appropriate compression algorithm for your available bandwidth.

Audio	RTP Voice Data Payload	Packets per secon	LAN (bps)	% Overhead LAN	WAN (bps)	% Overhead WAN	Algorithmic Delay
G.723.1	24 Bytes	33.33	20,800	225%	9,867	54%	8
G.729a	20 Bytes	50	29,600	270%	13,200	65%	40
G.711 (64k)	160 Bytes	50	85,600	34%	69,200	8%	20

G.711 vocoder provides a compression /decompression (companding) algorithm to deliver precision transmissions of encoded speech signals sampled at the rate of 8 kHz in conformance with the ITU-T G.711 standard. G.711 implementation includes independent user-callable functions that perform all of the μ -law and Alaw encoding and decoding operations. The most common application for G.711 vocoder is in telephone networks.(Required large bandwidth, thus at least 128kbps bandwidth in each direction is required but will deliver precise transmission of speech)

G.723.1 is for low bit rate speech compression for transmissions at either 5.3, or 6.3 kbit/s. It is the recommended codec for H.324 compatible PSTN videotelephony, and has recently been recommended by the VoIP Group of the IMTC as the default low bit rate codec for Voice-over-IP networks, such as the Internet. (Very high compression whilst maintaining high quality audio Requires a lot of processor power Often used by dialup VoIP users for optimal quality.)

G.729A offers low complexity, G.729-compatible, speech compression at 8kbit/s. It is the recommended codec for V.70 DSVD (Digital Simultaneous Voice Data) Modems, and is being considered by the Voice over Frame Relay Forum as the recommended codec for voice over frame relay applications. The coder processes signals with 10 ms frames and has a 5 ms look-ahead which results in a total of 15 ms algorithmic delay (Excellent bandwidth utilization for toll quality speech Performs well under random bit errors with a minimum delay 15 ms)

How Many Simultaneous Calls Can go Down My Link?

The following chart illustrates the theoretical maximum number of simultaneous voice calls that can be delivered over a WAN for a given link speed. This does not take into account any bandwidth that may be required for data traffic between sites or the physical limit of VoIP calls for the specific version of IP Office in use. The number of simultaneous voice calls can be in excess of the capabilities of the individual platform, where the calls transit the switch as data traffic. In this situation compression resources are not used but obviously must be catered for in the overall bandwidth provision.

Compression		G.729a(8k)	G.711(64k)
Algorithmic Delay (seconds)	0.08	0.04	0.02
Number of Calls			
64Kbps Link	6	4	0
128Kbps Link	12	9	1
256Kbps Link	25	19	3
512Kbps Link	51	38	7
1Mbps Link	103	77	14
2Mbps Link	207	155	29