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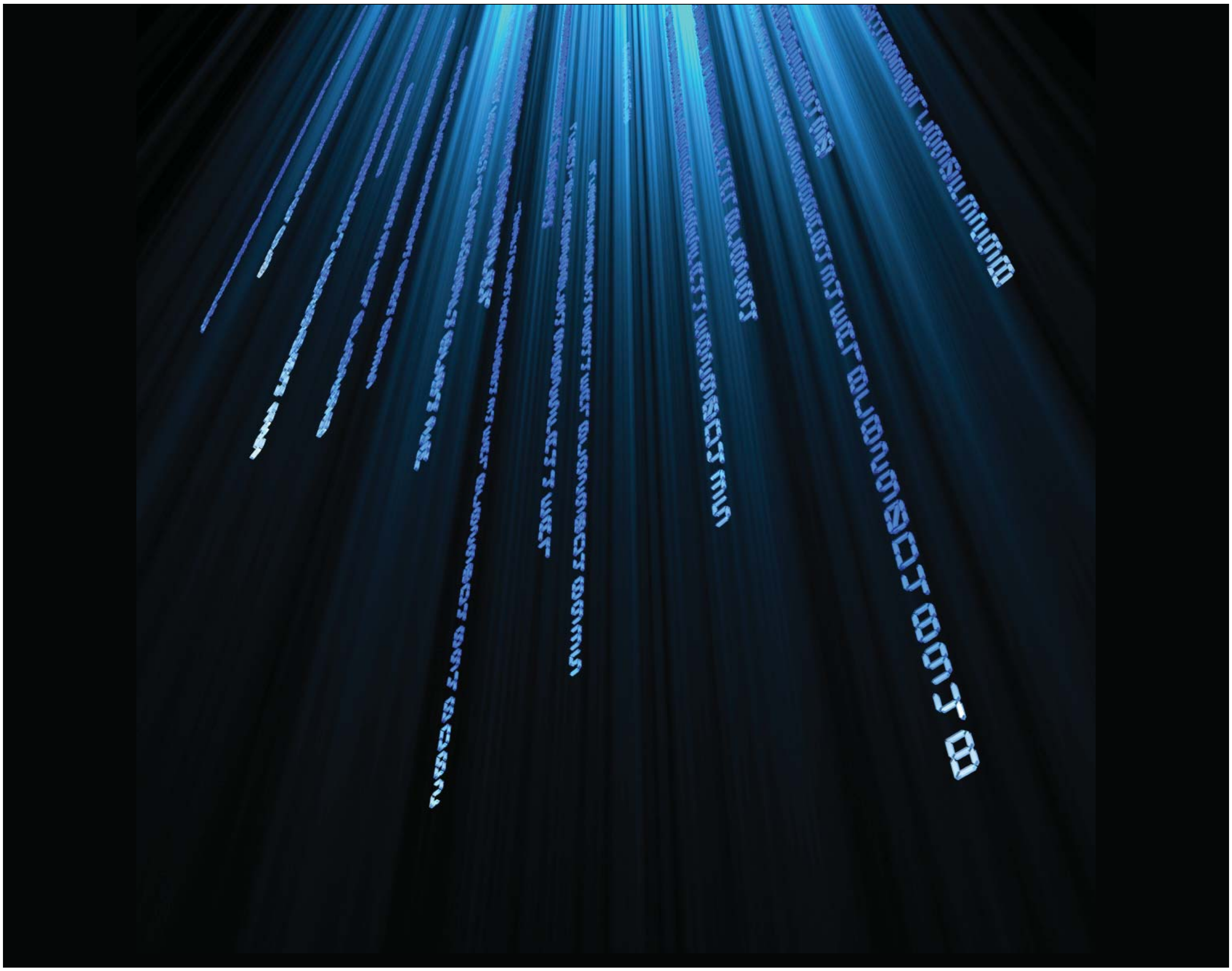
Decoding the Meaning Behind Codec



Decoding the meaning behind Codec

So you have decided to go with the times and consider VoIP for your business' voice needs. The low cost of VoIP, its "unified communication" options, and the freedom to take your business' calls anytime anywhere are all ample enough reasons to be happy with this decision. However, there are still a few things that the savvy well-informed decision maker must know in order to choose the best VoIP configurations to get the most bang for their buck. Understanding your Codec options is one of these key concepts.

What is Codec?

Codec stands for "Coder"/"Decoder". Simply put, Codec is what breaks voice down and encodes it on one end of a voice call, transmits it down a wire (physical or digital), and then decodes and makes it understandable on the other end of the call. Before the creation of VoIP voice options, businesses had to put up with the high costs associated with voice services because of the lack of Codec options available.

Nowadays however, one of the greatest perks of choosing VoIP over traditional methods of making phone calls is that you can take your Codec options into your own hands. With a little background knowledge on Codec, you can pick the best Codec configurations for your business, saving you money and bandwidth, without compromising call quality.

How does a Codec break down sound?

First, Codecs break sound down into slivers of time. Then, these slivers of time are compressed and coded into samples known as "packets" which can be transmitted. The longer a voice conversation is, the more "packets" are produced and the more bandwidth is needed to transmit these packets at a normal consistent rate which is 50 packets per second for most codecs.

A poor Codec configuration can prevent packets from being received at a constant rate which may either result in a broken connection because packets are dropped, or in a dropped call altogether because of the connection's inability to handle the sheer volume of packets.

What are the most important factors which affect my Codec needs?

When making a wise decision regarding the appropriate Codec options for your business, you must consider 3 very important things:

- **Sound Quality**- What are your sound quality needs? Do you need your calls to be in high-definition or is traditional tone quality acceptable for your needs?

- **Bandwidth**- How much bandwidth does your business currently have available to expend on your VoIP service?

- **Call Volume**- How many calls are active at any given point in a day?

What are the most common forms of voice Codec and which one is most appropriate for my business' needs?

CODECS	BANDWIDTH	QUALITY	DROP SENSITIVE?
1. G711	64 Kbps	Tone (Traditional)	No
2. G722	64 Kbps	High Definition	No
3. G729	8 Kbps	Filtered	No
4. iLBC	13.3 Kbps	Filtered	Yes

Codec #1- G711: This is the standard Codec of choice used in most VoIP systems. It has a fairly high bandwidth usage averaging 64 Kbps and produces audio quality consistent with a traditional analog telephone system. This is the most widely used codec in the VoIP world primarily because of history. The G711 codec was originally introduced by Bell Systems in the 1970s and is the codec of choice in public-switched telephone networks and integrated services digital network lines.

Codec #2- G722: This is a wide-band codec offering high definition voice. G722 uses the same amount of bandwidth as G711, though the resulting audio output is of a much higher quality ("high definition"). Calls made with this configuration are very crisp and clear. G722 is a newer codec (standardized in 1998) compared to G711 and is rapidly gaining momentum as a replacement to G711 in the VoIP world. A codec of this caliber ("high definition") requires the support of HD phones or phones which explicitly state supporting G722.

Codec #3- G729: This is a low bandwidth codec that compares to G711 in audio quality. This Codec compresses calls to only require 8 Kbps of bandwidth, making it perfect for businesses with low network bandwidth and/or high call volumes. G729 manages to use a lower bandwidth by compressing audio data and by efficiently transmitting silence and background noise. Analog hiss is simulated digitally by inserting comfort noise during silence to assure the receiver that links are active and operational.

Codec #4- iLBC (internet Low-Bit Codec): This is a low bandwidth codec utilizing 13.3 Kbps of bandwidth and offering equivalent audio quality as G729. It handles the case of loss of frames through graceful speech quality degradation. Therefore, if a packet is dropped on your end of a call, this codec accommodates for this drop so that your listener never experiences a break in the conversation. This codec is ideal for business locations where the underlying network connections is poor (for example microwave links) or for low bandwidth or high call volume scenarios.

Still Unsure?

Making a decision regarding your Codec options can be very confusing. If you would like to have us at Crexendo contact you and have one of our professional consultants help you make this decision, please feel free to reach out to us at **sales@crexendo.com** or call **1.877.517.7772** for a free consultation.