White Paper

THE LAST MILE CHALLENGE AND TRANSCODING

Why Carriers are adopting Low-Bit-Rate coding in their infrastructure and the impact on application developers

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AudioCodes

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Abstract

As next-generation VoIP telephone carriers like Vonage, AT&T and others have started to deploy their services into the market, the "last mile" to the customer has once again proven to be the hardest to cross. Both technical and business hurdles have required some sudden changes in deployment models, equipment vendors and application solutions. While the VoIP service market is still up for grabs, the carriers are scrambling to retool their offerings and solve the "last mile challenge". This article will examine why the last mile has caused the VoIP carriers to retool, how they are addressing the challenge and what it means to equipment and application vendors.

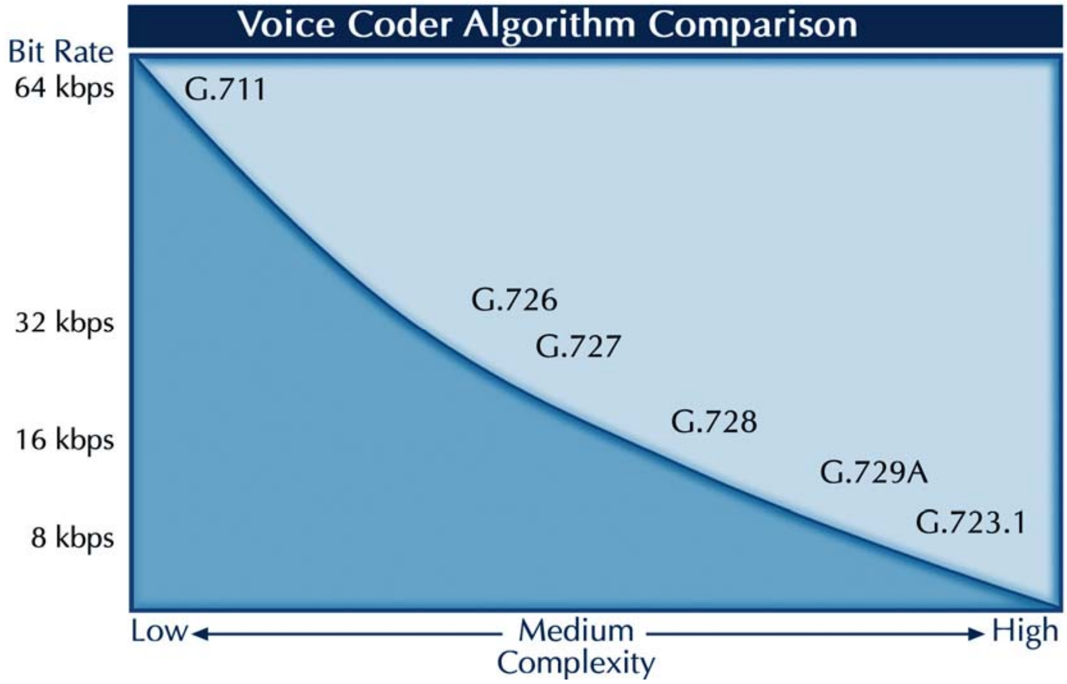
The Year of VoIP

This last year will surely go down in history as the year VoIP came out of the lab and into the home. Consider that Vonage, AT&T, BellSouth, Verizon, SBC and others have all launched their enterprise VoIP-based broadband telephone services during this year. Every home in the world that has a broadband internet connection is a potential customer. The VoIP carriers were not limited by their geography or regional networks.

With these heavy-weight contenders now providing telephone service over the customer's existing broadband connection, one can only imagine the competition on pricing and service offerings that are soon to come.

Deployment Challenges

One of the challenges VoIP Carriers have had to overcome early in their network planning was choosing a voice coding standard for their services. Unlike traditional TDM networks, VoIP offers a wide range of voice coding and compression algorithms.



From uncompressed G.711 at 64 kbps to highly compressed G.723.1 at 5.3 kbps, the VoIP carriers could now choose the level of voice compression that would be applied to their customers.

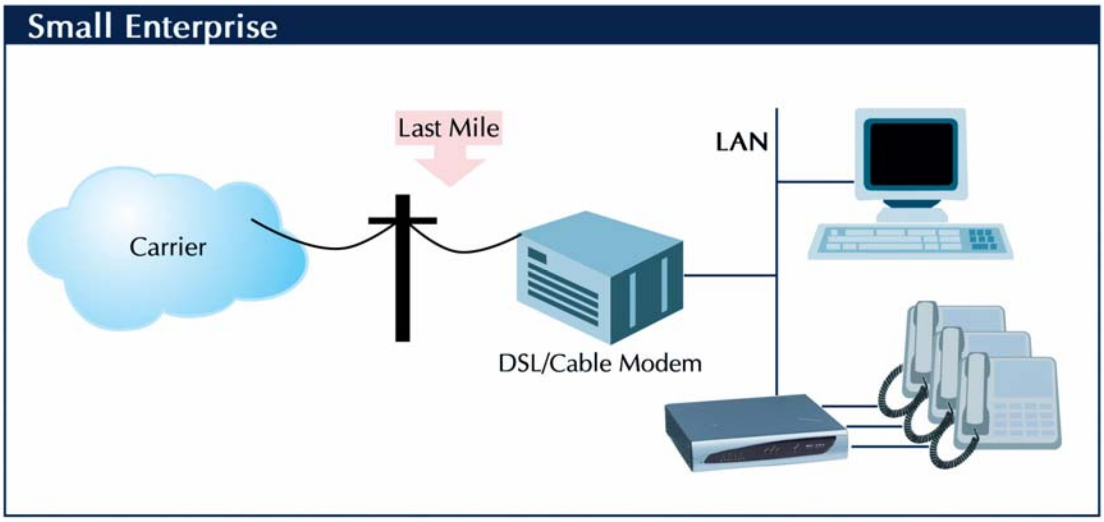
Initially, many of the VoIP carriers were convinced that they would have plenty of bandwidth on their network backbones and transporting uncompressed G.711 speech would be preferred.

Advantages that G.711 has over other low-bit-rate coding algorithms:
Good voice quality
Reduced cost media gateways
Easiest to manipulate
Most reliable with speech recognition applications

Disadvantages of G.711:
Poor network efficiency
Lacks missing packet interpolation

The Last Mile is the Hardest Mile

As the VoIP carriers started to deploy their services in field trials, an interesting (and in some cases, unexpected) challenge slowed their progress. Virtually all of the VoIP carriers planned on leveraging the existing broadband infrastructure to the customer's premise. This means their service was going to piggy back on the customer's existing DSL, Cable Modem or wireless broadband connection as it went from the Point of Presents (POP) to the customer's home or business.



Using the existing infrastructure is imperative to make the business model work. The VoIP Carriers have to leverage the existing broadband infrastructure.

Choke Point

Once the first trials were installed, a real-world choke point was quickly recognized. For a customer to have one or more reliable conversations over a broadband connection, the circuit had to be shared. It turned out that the uncompressed G.711 payload was just too great to make voice telephone calls reliable.

Whether a Wi-Fi network is in a public space or a cable-modem link to a small business, every link has its cost/performance limitations.

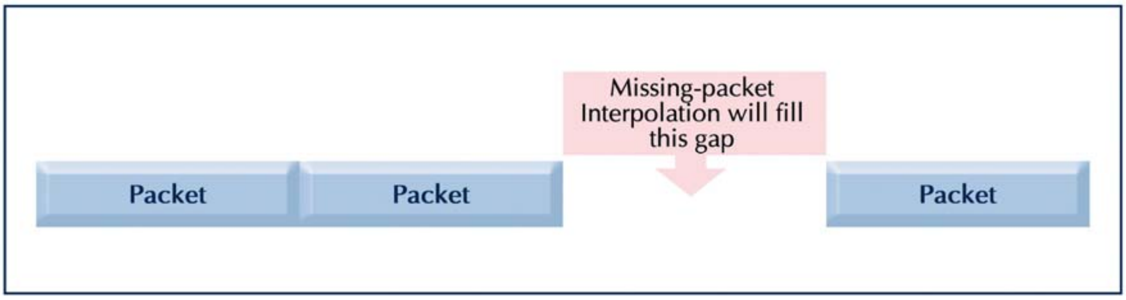
Additionally, customers running small businesses want to have two or more telephone conversations on the same broadband connections. Having the ability to deliver four, six, or even eight voice conversations on one broadband circuit was an important requirement.

Low Bit Rate Coders Advantages for VoIP Carriers

To solve the "Last Mile Challenge", the VoIP carriers have adopted Low Bit Rate (LBR) coders for their voice traffic for a number of reasons:

First and most obvious, they reduce the network bandwidth requirements. Depending on the coder, they can cut the required bandwidth to $1/10^{th}$ of the bandwidth needed by G.711. This allows a VoIP carrier to operate over a much wider range of broadband services and allow for multiple voice conversations on one circuit. This is especially important if any of the network connectivity is accomplished via wireless technologies like Wi-Fi.

Second, the LBR coders typically offer better voice quality in real world applications as they include missing-packet interpolation. Real-world networks drop and delay packets due to congestion in the network. Missing-packet interpolation features smooth over the audible gaps left by missing packets, providing near-toll quality in networks with imperfect engineering.



Based on both the business and quality advantages that LBR coders offer, many of the current VoIP carriers have shifted some or all of their service to LBR coders. Fortunately, almost all of the consumer-grade gateways, telephones, and soft-client packages offer at least one LBR coder option, making support for LBR coders a non-issue at the end-points.

Wait, what about the Applications?

To make a reasonable telephone service offering viable, carriers need applications. Basic applications one takes for granted like announcements, voicemail, IVR, directory assistance, etc. needed major overhauls (if not completely rewritten) to interface into an all-IP infrastructure.

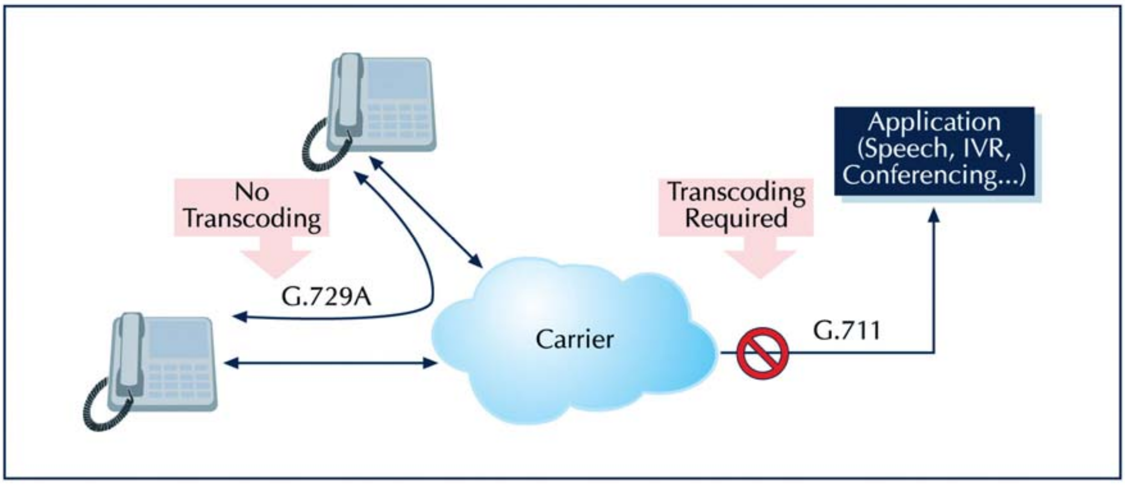
Meanwhile, the VoIP carriers were working on a differentiation strategy. How was their service going to be better than the others? Also, what optional services could they offer that would drive minutes and create loyalty? Just as in the wireless carriers, the VoIP carriers needed creative applications that are "sticky" and make it attractive for customers to stay with the service, despite lower cost offers.

A set of new technologies that has matured making application development faster was Automated Speech Recognition (ASR) and Text–to-Speech (TTS). Both technologies have made very creative and easy to use applications within reach, speeding development and lowering the cost of application development and maintenance.

Unfortunately, ASR and TTS applications aren't directly compatible with low bit rate coders. Somehow, the speech had to be converted to/from G.711 for these applications to operate properly.

Transcoding

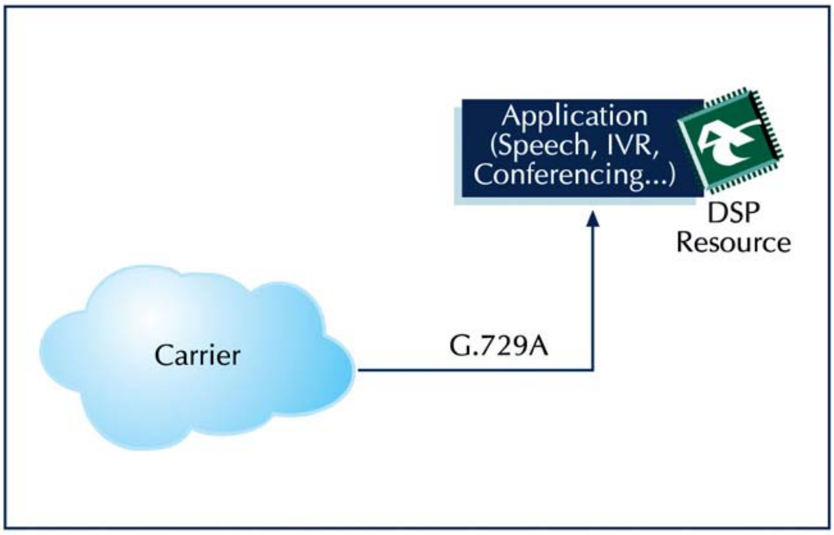
Transcoding is the process of converting from one voice coding algorithm to another, in many cases, from a LBR coder (G.723.1 or G.729a) to G.711 algorithm. Transcoding is very resource intensive and must be done in real-time with little or no delay. Generally, transcoding is accomplished using DSP technologies, either in the form of a DSP accelerator card or via an external media server chassis.



One might ask: "Does every conversation need transcoding?" The answer is "it depends", generally only the conversations that are interacting with the applications or different phone types need transcoding. Other end-to-end voice conversations between similar end points would not require transcoding and thus would use the LBR coder for the duration of the call.

Embedded Transcoding Resources

Some application environments embed the transcoding resource. Many times this is accomplished using a bank Digital Signal Processor (DSP) chips on a standard PCI or cPCI form factor card. Application developers have access to a wide range of DSP "resource" cards for this purpose.



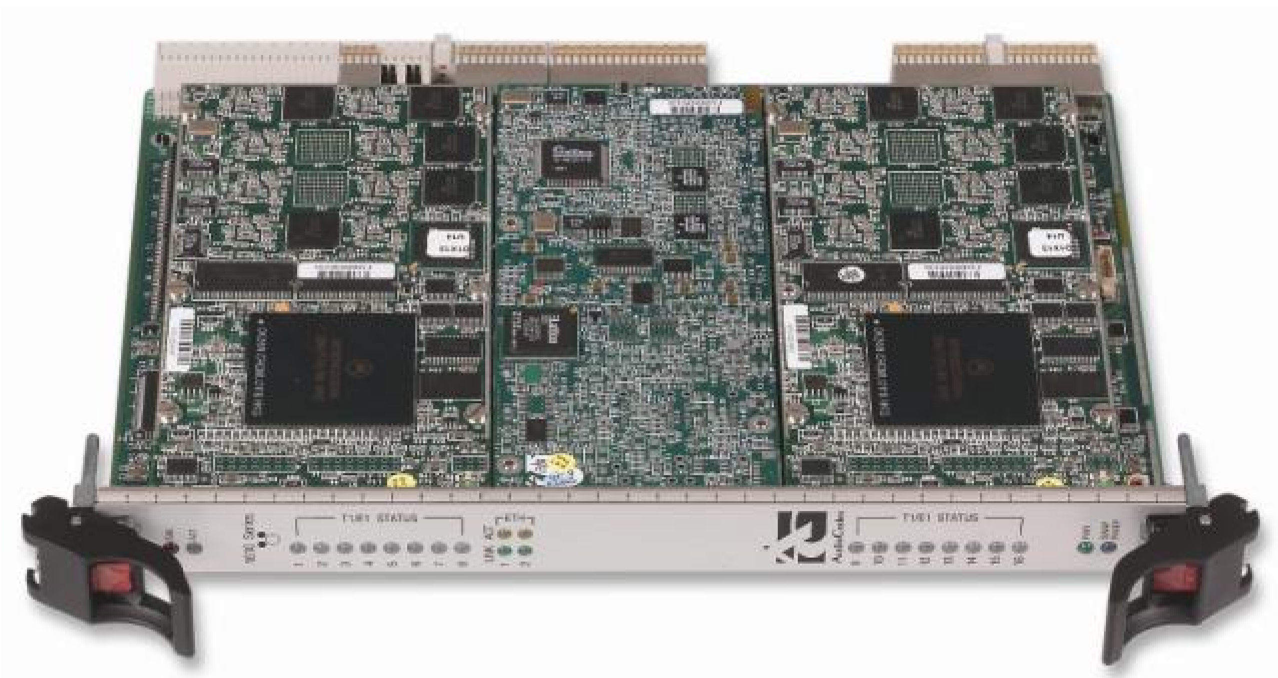
Choosing a Transcoding Solution

As the VoIP carriers continue the deployment of services and sophisticated applications, the challenges of transcoding in the network are real, but can be overcome. Both DSP resource cards and stand-alone media server platforms can close the gap between the LBR coded end-points and high-performance applications.

AudioCodes offers a set of distinctive blade-level solutions that can solve the transcoding needs of carriers and service providers:

IPM-260/TRB

In a PCI form-factor, the IPM-260/TRB offers a highly efficient transcoding resource that can convert from 15 to 128 LBR to LBR sessions in one slot.



IPM-1610/TRB

In a cPCI form-factor, the IPM-1610/TRB offers greater density, transcoding from 30 up to 240 LBR to LBR sessions in one cPCI slot.

IPM-6310/TRB

Also in a cPCI form-factor, the IPM-6310/TRB offers even higher density with up to 1008 LBR to LBR transcoding sessions.

Control – A feature of the IPM-260/TRB is that the transcoding control can be accomplished via the powerful AudioCodes VoPLIB API or via the Session Initiation Protocol (SIP).

Benefits of AudioCodes Transcoding Solutions:
Available as embedded technology in a standard PCI or cPCI form factor
Meets the density needs of the application
Wide range of simultaneous coder options
Low-latency and Low-jitter
Uses standard control protocols or proprietary API
Meets the Target Cost per Channel

Summary

As the market penetration of the VoIP carriers continues to grow and the need for specialized applications grows too, transcoding DSP resource cards and media servers will play a key role. Watch for significant market growth with the transcoding platforms to settling around standard protocols, improvements in performance and increased density.

The carriers have made a significant commitment in VoIP. They can't afford to let transcoding trip them up when crossing the last mile.

About AudioCodes

AudioCodes Ltd. (NASDAQ: AUDC), Your Gateway to VoIP, provides innovative, reliable and cost-effective Voice over Packet (VOP) technology and Voice Network products to OEMs, Network Equipment Providers, Service Providers and System Integrators worldwide. AudioCodes provides a diverse range of flexible, comprehensive media gateway and media processing technologies (based on VoIPerfect™ – AudioCodes' underlying, best-of-breed, core media gateway architecture) and Session Border Controllers (SBCs). The company is a market leader in product development, focused on VoIP Media Gateway, Media Server and SBC technologies and network products. AudioCodes has deployed tens of millions of media gateway and media server channels globally over the past few years and is a key originator of the ITU G.723.1 standard for the emerging Voice over IP market. The Company is a VoIP technology leader focused on quality, having recently received a number one ranking from ETSI for outstanding voice quality in its media gateways and media servers. AudioCodes voice network products feature media gateway and media server platforms for packet-based applications in the converged, wireline, wireless, broadband access, enhanced voice services and video markets. AudioCodes enabling technology products include VoIP and CTI communication blades, VoIP media gateway processors and modules, and CPE devices. AudioCodes' headquarters and R&D facilities are located in Israel with an R&D extension in the U.S. Other AudioCodes' offices are located in Europe, the Far East, and Latin America.

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