Answering the Call for Voice Transformation
Voice services are changing as we speak. Now, the basic calling services we once took for granted are part of advanced voice services we can take anywhere. Voice and data, converged on the same IP network, are answering the call for less complexity and more cost control.

And, enterprises are saying hello to hyperconverged voice: Voice services integrated with real-time communication and collaboration tools and the applications that drive their operations – fueling transformation as they become more mobile and social.

VoIP: The Great Enabler
While hyperconverged voice is changing the way we interact, it wouldn’t be possible without Voice Over Internet Protocol (VoIP) changing the technology of talk first.

Instead of pushing voice packets through the pipe of a traditional phone network, they’re transported over an Internet connection. While legacy TDM-based voice services are based on aging, costly and hard-to-replace telephony equipment, VoIP helps lower costs by letting you leverage your existing data network and the investments you’ve already made.

As a result, you can pay for and manage a single voice and data network. You can also combine the dial tone reliability employees expect with the bandwidth scalability you need for high-quality voice services. And, you can ease the expansion of voice services to multiple locations, because they can go anywhere your IP connections and your mobile workers do.

With mobile VoIP, anytime, any where, any device access to voice services rings true: Desktop PCs and phones, laptops and smartphones can all access VoIP services. Instead of missed phone calls and endless games of phone tag, a single phone number can ring on multiple devices simultaneously or sequentially – closing the gap between fixed and mobile work environments.

SIP Trunking: An Easy Transition
While VoIP enables you to converge voice and data, Session Initiated Protocol (SIP) trunking enables you to converge older voice infrastructures with newer IP technology.

SIP-Enabled Applications
Session Initiation Protocol (SIP) connects VoIP transport services to legacy voice environments and with the communication and collaboration applications you need for today and tomorrow.
As a VoIP service, SIP uses the Internet to connect legacy PBX, key systems and TDM trunk lines to the public phone network. So, you can protect existing voice investments while you adopt IP telephony at your own pace.

For large enterprises delivering voice services to many distributed locations, the cost savings potential of SIP is loud and clear: A single IP-based SIP trunk can replace multiple TDM trunks, enabling you to consolidate and simplify your entire voice infrastructure.

With the dynamic provisioning and bandwidth allocation of an IP network, it can take only minutes to add SIP connections, versus months in a TDM environment.

So, you can support the voice traffic patterns of a typical business day, but you can also meet unpredictable boosts in calling activity with burstable network capacity. As call volumes decrease, that capacity can be quickly and cost-effectively dialed back or allocated elsewhere.

Joining VoIP on the evolutionary ladder of voice transformation, SIP leads to the next critical rung in convergence: unified communications (UC).

UC: The Ultimate Convergence
UC integrates voice calling with multiple communications and collaboration tools – from presence, instant messaging and conferencing to email and unified messaging, team collaboration and shared calendars – and makes them easily accessible behind a single user interface.

It’s here that convergence shifts into hyperconvergence. It’s where UC applications like audio, video and web conferences enable real-time collaboration between employees, partners and customers. It’s where mobilizing UC applications provides access from nearly any device, anywhere.

More and more, hyperconverged voice and UC are moving to the cloud.

It’s where transitioning communication sessions from a voice call with one to an audio conference with many happens seamlessly – between tools and across fixed and mobile networks. Further, it’s where voice and UC are integrated with your core applications to accelerate business processes, or with social networks that embed IM and “click to call” connections without leaving the page.

More and more, it’s also where hyperconverged voice and UC applications reside in the cloud to provide IP telephony and UC as a Service – once again transforming how you deliver services to your enterprise.

Hyperconverged Voice and UC: Here to Stay
Whether it’s the convergence of voice and data on the same network, the merging of IP and legacy voice environments, or the pinnacle of hyperconvergence with UC, voice transformation is here to stay as more enterprises rediscover the value of voice.

With a rich legacy of voice and one of the world’s largest IP networks, AT&T is uniquely qualified to help you transform your business. For your next step on the hyperconverged voice highway, talk to your AT&T representative.

For more information contact an AT&T Representative or visit networkingexchangeblog.att.com/topics/voice-transformation/