WHITE PAPER

Making the Journey to SIP Contact Centers

From the experience of early adopters
a tried and true path to success emerges

Compiled by: McGee-Smith Analytics

Sponsored by: AudioCodes, Genesys

WHITE PAPER

WHITE PAPER

Flexibility, Scaling and Cost Savings

The Foundation of the Contact Center Evolution

As organizations have grown, completed acquisitions or moved their operations, the customer care provided by their now aging digital-based ACDs has proven to become a barrier to further growth. The expense of expansion and maintenance along with legacy T1 and ISDN network architectures is a serious limitation to growth and a catalyst to re-examine the underlying technology of the customer care contact center.

Businesses that have abandoned legacy equipment and moving to a SIP-based contact center architecture have realized greater flexibility to meet their business needs. With SIP, operations can be centralized in private or cloud data centers, while customer care agents can be pooled and sit virtually anywhere where Internet connectivity is available. A SIP network also allows for public network access to be consolidated with more flexible and less expensive SIP trunks. SIP also enables enhanced IVR sessions - using speech recognition and call routing applications - that facilitate more efficient and friendlier handling of caller traffic. Behind the scenes, centralized call management and improved reporting allow greater visibility to customer care operations, the lifeblood of many businesses.

Call centers may have been born on circuit-switched digital equipment, but a SIP foundation is needed to deliver the promise of an exceptional customer experience.

Customer Expectations, and Technology Improvements

Closing the Gap

While there are compelling technical reasons to migrate legacy ACDs to SIP, there are equally compelling business imperatives. Line of business executives understand that they need to satisfy increasingly "connected" consumers who expect to call, chat, SMS, tweet and "like" (or not!) the enterprises with which they conduct business. Consumers increasingly presume that an enterprise will remember their prior interactions and proactively personalize their experience across these various channels. The term "Customer Experience" has emerged as the phrase used to encompass these demands for convenient, competent, proactive and personalized customer service.

> *To quote business author Steven Covey, "Begin with the end in mind." As you chart your migration to SIP, make sure each step takes you closer to your ultimate goal of improved customer experience*

This Customer Experience imperative is often leading to the conclusion that IT organizations will need to evolve their contact center infrastructure. Fortunately, it is becoming easier to build and manage contact centers that meet these high expectations. Leveraging new technology and standards, IT organizations can provide faster responses to the needs of their business counterparts while reducing operational costs. The "gap" between customer expectations and enterprise capabilities can be successfully met with greater ease than ever before.

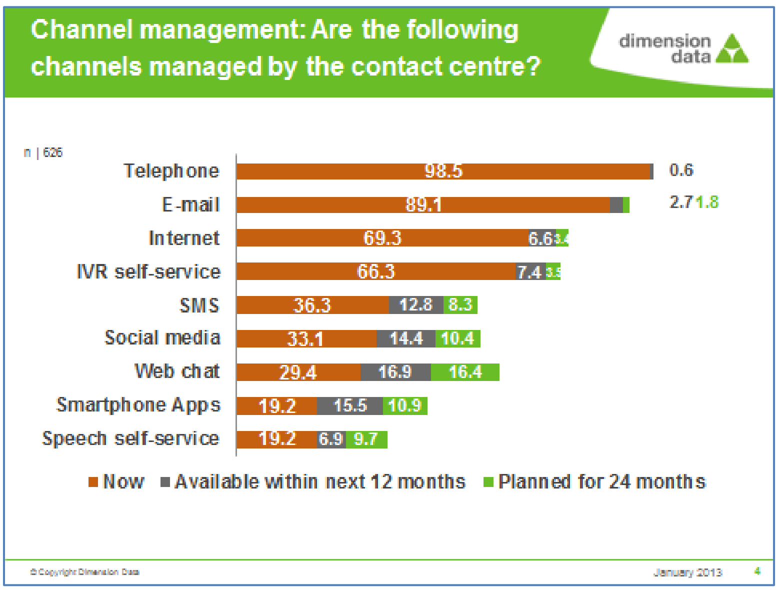
What does success look like? A senior IT executive that has completed this SIP journey for their mainline business recently told her story. A sister line of business was caught in a jam: their technology provider declared the contact center platform to be end-of-life and they quickly needed to develop alternative capabilities. The executive's team studied the

WHITE PAPER

requirements and traffic flows for this sister business, and found they could quickly adapt the primary contact centers to meet those needs – at virtually zero incremental capital expense. The flexible SIP infrastructure that had been put in place for one part of the business was able to address the customer experience imperative of another line of business – quickly.

Customers Expect to Have Personalized Cross-Channel Conversations

There is pressure on customer service centers to be accessible from every type of consumer device – smartphones and less-smartphones, tablets, email anywhere and web browsers everywhere. This transition is evidenced in the results of Dimension Data's 2012 Global Contact Centre Benchmark Report. As seen in the accompanying chart, the handling of email has become well-established in contact centers, and support for social media, web chat, and mobile applications is predicted to rise dramatically over the next two years.



As the data portrays, many organizations have added communication channels such as e-mail and chat in the past few years, but what the data doesn't show is that often this was accomplished with add-on, siloed systems staffed by separate agent groups. As companies plan over the next two years to add more and varied interaction channels to support customer demand, the complexity of separate systems will multiply. The solution is to support cross-channel customer interactions that could simultaneously require voice, chat, video or web co-browsing with a single software application.

One technology that promises to support this cross-channel customer interaction model is based on an emerging standard called WebRTC. With a significant push from web giant Google, WebRTC's design makes it easier for people to communicate very simply through their web browser using audio, video, text and other services. A demonstration of WebRTC working between a Google Chrome user and a Mozilla Firefox user is available here.

Don't think flash-cut – think Journey. Many successful deployments, especially of multiple centers in multiple cities and counties, were done in stages. Each stage can deliver ROI, helping fund the next stage

So what is the business impact of this technology change? Imagine if customer service agents and customers had such a simple way to share images, text and descriptions. The time to display, mutually understand and resolve any product or service problems could be vastly shortened. This would improve both the customer experience and also elevate the customer service agent's sense of job satisfaction.

However, leveraging WebRTC requires a SIP-based contact center platform, capable of integrating the voice and video formats native to WebRTC. Legacy TDM ACDs are simply incompatible with WebRTC and the promise it holds.

Steps Along the Journey

Although the precise starting point of this journey – at least in terms of existing contact center assets and network topology – will be different for each enterprise, there are a few common lessons that apply. For example, few companies try to jump-forward to the SIP end state in a single bound. Moving sequentially through a series of steps can serve two purposes: it mitigates risk while allowing the return on investment of early steps to fund later ones.

Here is a typical journey to SIP for an organization that has multiple operational centers.

The Journey Begins: a Representative Virtual Contact Center

Generally, operations centers have individual ACDs with their own public network connections, linked by TDM trunks to pass voice calls between centers. Multi-site call routing software (e.g., Genesys) and associated customer service applications such as reporting and CRM are placed in a centralized location, usually a data center.

Migration to SIP Server

The virtual contact center architecture described above, using Genesys routing software, does deliver benefits of scale and better access to resources than legacy single-site ACD environments. However, the maintenance expense of legacy ACD systems has become prohibitive, often in conjunction with the technology being declared end-of-life by its manufacturer. Hundreds of customers have found that converting from multiple ACDs to SIP Server architecture is a step along the TDM to SIP journey that results in a dramatic reduction in maintenance expenses.

> *While keeping existing telephones may be a short term money saver, in the long term modern devices with modern capabilities makes more sense.*

Within a center, the migration from legacy ACD to SIP Server involves converting both network connections and agent endpoints. A proven technique is to convert the network facilities in a "like-for-like" mode that permits testing and graceful migration between the old and new. In this mode, a media gateway converts the PRI circuits to SIP as needed by the SIP Server. A migration to SIP trunks would be initiated at a later time after the migration to SIP Server is complete and the legacy ACDs are taken out of service.

A key consideration for this architecture is supporting a progressive migration from TDM circuits to SIP trunks. AudioCodes Mediant Media Gateways facilitate a migration to SIP trunks with optional Enterprise-Session Border Controller features, providing a secured demarcation between the enterprise and public network domains. This allows support for the both PRI and SIP trunk network connections during the transition period.

The transition to SIP Server creates the opportunity to switch to standard IP phones, much more cost effective than vendor-dictated proprietary ACD or PBX phones. One criterion to consider in selecting phones is the ability to support central provisioning and support – important in a centralized contact center to preserve the benefits of consolidation. Another increasingly important consideration is native support of the OPUS codec by the contact center SIP phones. OPUS is the preferred WebRTC wideband codec, designed to deliver superior end-to-end voice quality over the Internet, facilitating transcode-free calls with WebRTC-enabled end-points.

WHITE PAPER

Network Readiness Assessment:

Before moving voice traffic from their dedicated trunk facilities onto a converged network, a key success factor is assessing the existing data network capacity and performance. Countless post-implementation issues have been eventually traced to network performance problems, typically only found and corrected after something stops working or behaves incorrectly. Neglecting this basic step invariably leads to costly delays in a solution deployment.

> *"Voice quality – and hence the customer experience -- is highly dependent on the network it rides on. Ensuring your network is ready to support a SIP deployment is imperative"*
>
> *Elizabeth Gotto,*
> *Ticketmaster,*
> *Sr. VP Global Contact Center Technology*

A network readiness assessment proactively investigates a network to understand its potential bottlenecks and whether it is suitable for the deployment of a Genesys Contact Center solution. It measures the impact of both application data traffic and voice on the existing data network through the origination and termination of simulated test calls. As SIP voice traffic consists of many small packets, it can result in additional network traffic up to six times more demanding than most other types of data traffic.

A Genesys Network Readiness Assessment, conducted by AudioCodes, includes specific qualitative measurements of network and carrier elements that affect voice-network quality. This approach applies best practices and tools to proactively identify potential issues and point the way to a successful SIP deployment.

In a world of sessions, the end-to-end management of voice quality becomes a key consideration. As soon as best effort networks become part of the mix, it is prudent to monitor and proactively manage resource allocation. AudioCodes Session Experience Manager provides such a solution.

Migration to SIP Trunking

Many enterprises have found the business case for SIP trunking to be quite compelling, particularly in comparison with their existing PRI contracts with often legacy carriers. Part of what drives the cost reduction is the new carriers that exclusively deliver SIP trunks are introducing more competition to the industry.

As compelling as dramatic reduction in monthly expenses can be, there are equally compelling technical advantages contact centers can gain with SIP trunking. These include simplified control for alternate and backup routing and a reduction in the amount of premises equipment required.

With SIP trunks, enterprises must secure communications between the enterprise and public domains, best accomplished by the use of Session Border Controllers (SBCs). Genesys can provide a software SBC pre-integrated with its SIP Server. As cited in the section on SIP Server, IT staff may anticipate the future need for SBCs by choosing media gateways that can migrate to SBCs, such as those available from AudioCodes.

WHITE PAPER

As contact center management begins the SIP trunking migration step, they may find that it has already been deployed across the enterprise. In the past ten years, it has been common for organizations to replace TDM PBXs with enterprise telephony and unified communications applications across a distributed enterprise or between collaborative organizations. Often it has been only the contact center that remained on TDM technology. In that case, IT's task will be simplified, needing only to extend the existing SIP trunking network to include the contact center.

> *"Don't underestimate the value of centralized reporting – we live and die by our reporting."*
>
> *Elizabeth Gotto, Ticketmaster*

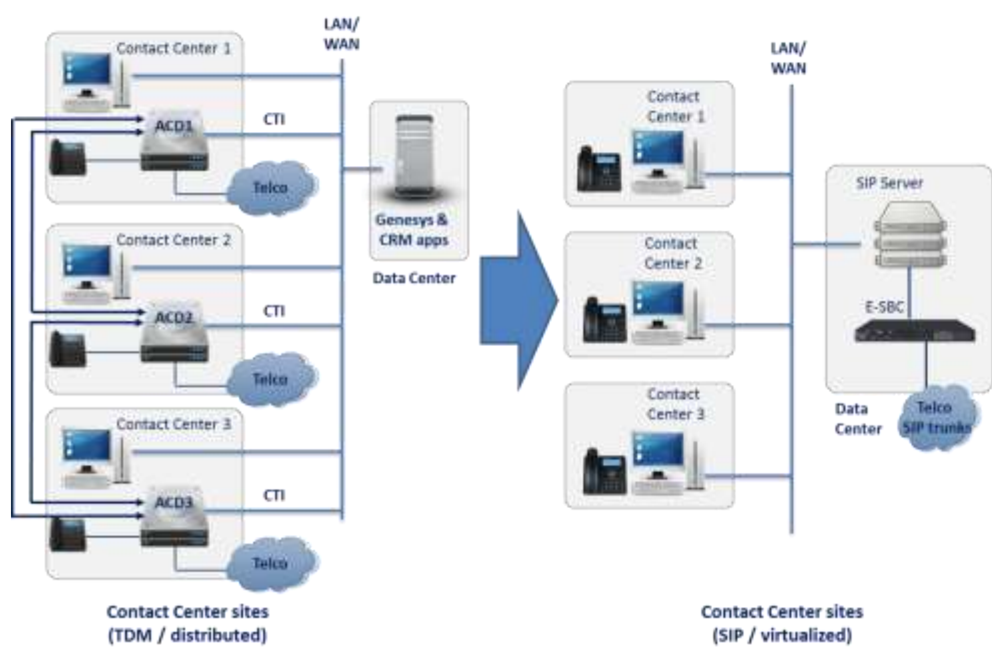
Stepping into the Cloud

It is typical for enterprises to deploy SIP contact center systems in data centers, where IT staff manage customer care solutions side-by-side with other enterprise software and networking. Larger enterprises with contact centers in multiple cities tend to deploy SIP contact center systems in a redundant configuration. Redundancy comes through additional data centers.

Increasingly, however, options to work with a cloud-service provider to provide the primary and/or redundant aspects of an SIP contact center deployment are available. As you define your journey to SIP, make sure to investigate both vendor-provided cloud alternatives (e.g., Genesys Platform on Demand) as well as carrier-based and other service provider offerings.

The journey of a thousand miles begins with a single step

Sometimes it's difficult to see how to get from point A to point B, from a myriad of equipment boxes and trunks in locations all over the country, even the world, to a simple data center-driven architecture. But it is a journey that others have already successfully taken and one that is well worth the admittedly hard work along the way.



The journey to SIP contact centers described here includes examples of best practices from leading practitioners, but every technology traveler will have their own specific points of interest and, of course, their budget. For example, business executives and the front line

WHITE PAPER

managers value the efficiencies and effectiveness that come from virtualizing the contact center resources. The payback here (read: budget) comes from the pooling effect of more people managed from a common center.

All CPE today? This is also the right time to evaluate your cloud options. Not ready to go all cloud? Consider hybrid alternatives

However, the underlying network topology may limit the degree to which customer queries can be easily moved from center to center, and this limitation puts a cap on the pooling effect and the access to specific skills that may exist in one center but not another. So the incentive for IT and operations managers to converge on a SIP network is to harmonize the technology across sites and make it simpler to match/move customer interactions to company resources. A network readiness assessment yields actionable insights into the possibilities and impacts.

Finally, a migration to SIP trunking yields present and future technology benefits; the specific financial gains should be investigated by examining existing network vendor contracts and commitments.

This journey to an SIP contact center can be exciting. We invite you to share your travelogue story so that all industry practitioners can find the swiftest and most rewarding paths forward.

Resources for the Road Trip:

Looking for more details, tools or resources?

Read the complete story and view a video interview with Elizabeth Gotto, Sr. VP Global Contact Center Technology Ticketmaster, discussing their journey to SIP at: http://www.audiocodes.com/Ticketmaster

For more information on AudioCodes networking solutions for Genesys SIP, visit: http://www.audiocodes.com/genesys

WHITE PAPER

Compiled by:

Sheila McGee-Smith
President & Principal Analyst
McGee-Smith Analytics, L.L.C.
603 672 2982
87 Horace Greeley
Amherst, NH 03031
sheila@mcgeesmith.com
www.mcgeesmith.com

About AudioCodes

AudioCodes Ltd. (NasdaqGS: AUDC) designs, develops and sells advanced Voice over IP (VoIP) and converged VoIP and Data networking products and applications to Service Providers and Enterprises. AudioCodes is a VoIP technology market leader focused on converged VoIP & data communications and its products are deployed globally in Broadband, Mobile, Cable, and Enterprise networks. The company provides a range of innovative, cost-effective products including Media Gateways, Multi-Service Business Routers, Session Border Controllers (SBC), Residential Gateways, IP Phones, Media Servers and Value Added Applications. AudioCodes' underlying technology, VoIPerfectHD™, relies on AudioCodes' leadership in DSP, voice cod- ing and voice processing technologies. AudioCodes High Definition (HD) VoIP technologies and products provide enhanced intelligibility and a better end user communication experience in Voice communications.

International Headquarters
1 Hayarden Street, Airport City
Lod 70151, Israel
Tel: +972-3-976-4000
Fax: +972-3-976-4040

AudioCodes Inc.
27 World's Fair Drive, Somerset, NJ 08873
Tel:+1-732-469-0880
Fax:+1-732-469-2298

Contact us: www.AudioCodes.com/info
Website: www.AudioCodes.com

©2014 AudioCodes Ltd. All rights reserved. AudioCodes, AC, HD VoIP, HD VoIP Sounds Better, IPmedia, Mediant, MediaPack, OSN, SmartTAP, VMAS, VoIPerfect, VoIPerfectHD, Your Gateway To VoIP and 3GX are trademarks or registered trademarks of AudioCodes Limited. All other products or trademarks are property of their respective owners. Product specifications are subject to change without notice.

Ref # LTRM-80034 01/14 V.1

WHITE PAPER