Convergence

Informing voice and data convergence initiatives with analytics

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Introduction

Convergence is a technology advancement that carries much promise and expectation. As an industry buzz-term it's been used whenever technologies combine - such as the new wave of mobile phones with built-in digital camera, GPS and MP3 player. And in the enterprise communications sector, the term was first applied to voice and data integration across a single network. In this white paper, we take a look at the drivers towards converged voice and data networks and provide recommendations on deciding the best options, managing evaluation and how to achieve a smooth implementation.

Voice and data united

The momentum behind voice and data convergence migration has been rapid, with some global enterprises having already migrated, and many evaluating migration for the near future. The perceived advantage of a one system does all service comes with a host of attractive benefits that appear straight-forward and logical for any business, large or small, to adopt. However, the perception that migrating to a converged network will be a win-win deal, delivering cost savings and increased productivity, is a perception that, in reality, comes with a host of challenges and can be difficult to achieve. With enterprises under increasing pressure to identify cost savings and deliver benefits to overall productivity, technology decision makers have clearly identified with the industry-hyped benefits of convergence.

Reduction in costs

The main driver for enterprises investing in convergence is undoubtedly cost reduction. The expected savings to be made from uniting data and telephony requirements into one infrastructure has fuelled much of the current adoption, the common belief being that a single network must surely be cheaper to run than separate voice and data networks. However, realising these cost savings can be a complicated exercise, and without thorough analysis and planning can lead to unplanned surprises further down the line.

The platform of the future

As current analogue PBX systems come to the end of their serviceable life, enterprises will naturally consider migration to convergence. The enhanced functionality of convergence provides two main benefits:

- **Improved functionality**
  Converged technology provides business benefits such as improved call management, mobility and free calls between sites. The growth of presence telephony popularised by instant messaging tools becomes possible to implement, enabling calls to be intelligently routed to staff, whatever their location or availability.

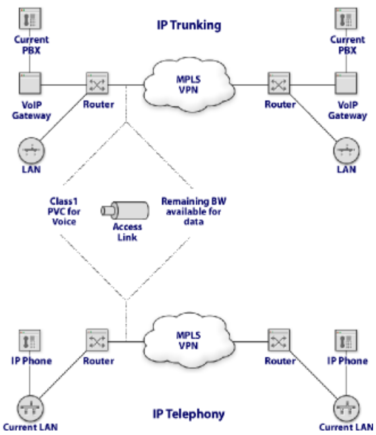
- **Simplified management**
  With separate voice and data networks, two or more different providers are often required, and with this, two teams of technical support staff with different skill sets. With a new converged network, enterprises can potentially reduce this to one provider and merge staff into one support team that utilises a single set of tools.

The choices available

There are two main types of convergence architecture commonly deployed in enterprises: IP trunking and IP telephony. Both utilise voice over IP (VoIP), but with very different hardware requirements.

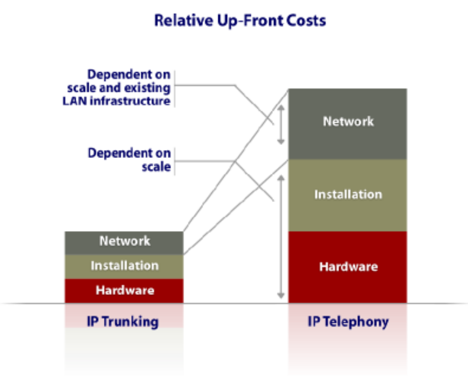
IP trunking

IP trunking uses VoIP over the wide area, with a traditional in-office telephone infrastructure. The existing desk phones and PBX can therefore remain, meaning that no additional investment is required on telephony hardware. The analogue-to-digital (VoIP) conversion occurs after the signal reaches the PBX by way of a VoIP gateway. IP trunking delivers many of the benefits of VoIP, but without a major upgrade to existing telephony systems. As a result, no investment is required in new telephones, LAN infrastructure or user training.



IP telephony

IP telephony requires investment in a whole new telephony system. The PBX is replaced by a call management server, and VoIP is directed straight to the users VoIP-enabled desk phones. There are significant increases to the upfront costs associated with IP telephony - the main contributing factors being the hardware, installation and network investment required. IP telephony systems provide a direct IP address for each phone, providing flexibility through plug and play for office moves and home working. With IP Telephony being central to many communication providers' roadmaps, the adoption is future-proofed for the coming years; however, the size of the upfront costs can make it difficult to realise the cost the cost savings that enterprises are seeking.



Understanding the challenges

The anticipated benefits of voice and data convergence are compelling. However, the quality and performance of both voice and data systems are critical to the running of any business, and convergence presents a risk to both. A clear understanding of the risk, and an open dialogue of requirements between the IT organisation and business, is therefore a fundamental and necessary part of any migration plan.

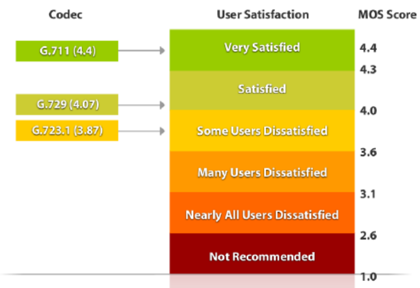
So, with communication as the lifeblood of any business, quality and reliability of service is a pre-requisite that has little room for compromise and is where many important questions need to be addressed:

- ***Will the voice quality be as good as our current analogue system?***
- ***How can we manage the bandwidth between voice and data?***
- ***How will it affect network availability?***
- ***What changes are required to the network to incorporate the voice traffic?***
- ***What new functionality can it bring to the business?***
- ***What are the cost implications?***

Voice quality vs. bandwidth

In planning voice-data convergence, the first difficult task is determining just how much bandwidth to give over to voice packets, carefully balancing the needs of both voice and data requirements. Detailed analysis of the existing voice and data networks is required to accurately ascertain current and future demands.Overall voice quality is determined by how much compression the voice signal goes through when it is digitised. Straightforward digitising uses 64 Kbps of bandwidth and thus provides a quality of voice service that most users would be very satisfied with.

The drawback however is that much of the network would be clogged up with voice packets, resulting in diminished performance for other data requirements. When the voice is compressed, it uses less bandwidth - but the more it is compressed, the more voice quality is compromised. The challenge for decision makers is to determine if voice quality can be sacrificed to achieve lower running costs. Sumerian has found that bandwidth costs can vary widely depending on the voice codec implemented. For example, implementing code G.711 (highest quality) can actually result in significant cost increases. In contrast, codec G.729 (medium quality) resulted in marked cost savings and provided users with a satisfactory voice quality. With G.723 (lowest quality) voice quality can be compromised, but delivers even higher cost savings.



Codec Options and Associated User Perceived Voice Quality (MOS)

Quality of Service (QoS)

Conversations over the telephone happen in real time. Unlike instant messaging and email where a small delay will go unnoticed, in phone conversation a delay or jitter is instantly recognised, disrupts the flow of communication and results in dissatisfied users. To provide a seamless voice delivery, successive packets need to be played out at a constant rate. To achieve this, networks must be QoS-enabled to meet delay and jitter targets and mark the voice packets for special treatment.

Without proper assessment of network traffic and bandwidth demands before implementation, enterprises may also find that the quality of voice service can be inconsistent at certain times when traffic across the network is heavy. In addition, voice compression is lossy, which means that it can be severely reduced when the voice packet is compressed and decompressed several times. This process takes place in several different scenarios, such as calls to mobiles or calls through voicemail. With voicemail, the original voice signal packet is compressed and decompressed when leaving a message, and then compressed and decompressed when retrieving the message. Each time the message goes through this process its original quality is weakened, resulting in - at worst - an inaudible message being retrieved.

New functionality - presence telephony

Instant messaging has become a popular and widely used form of real-time communication within enterprises. The use of status markers such as online , away and busy has contributed to its success, enabling staff members to quickly ascertain a co-worker's availability. Presence telephony is the integration of instant messaging with call routing, resulting in a call management process that routes calls by location and availability. Whilst the benefits of presence telephony are appealing, there are several challenges that can beset its successful adoption.

There is an overwhelming choice of proprietary solutions to choose from, which can result in a high potential for vendor lock-in. Additionally, the solutions have a strong coupling with Microsoft Windows authentication infrastructure, meaning that Active Directory schemas are required to be updated regularly. Incremental costs can be highly variable, and without adequate research into inbound and outbound call patterns, enterprises can be faced with higher call costs. Scalability is a concern and enterprises seeking to implement presence telephony need a clear technical and commercial exit strategy should the migration not be successfully deployed.

Developing a convergence strategy

With a myriad of challenges awaiting any convergence migration, a detailed strategy addressing these challenges is highly recommended. Although some telephony providers will offer an upfront assessment as part of their pre-sales service, it is recommended that decision makers ensure this incorporates the following series of programmatic steps.

1. Conduct an audit of your existing system

A detailed assessment of your current separate voice and data networks should be carried out to ascertain what the systems look like structurally, how they are used by staff and where the systems are in the investment cycle. For example, by understanding how your analogue voice system is used within your business in terms of service delivery and functionality, you can start to build a case on whether investment in a brand new IP Telephony system is justified, or whether a transparent shift to IP Trunking is a better solution. In addition, a service assessment dialogue with business management will serve to increase internal business satisfaction and aid an effective converged service delivery.

2. Analyse the consumption of your systems

The second step involves understanding how the current resources are being consumed by existing voice and data traffic. In essence, this will assess your technology readiness and determine if your existing WAN, LAN(s) and CAT5 telephony cabling are capable of carrying the additional voice data. To implement this successfully, a detailed measurement of consumption is required across all networks. On the current voice system, you need to determine the current and future call types, patterns and volumes carried across the system. This information can be obtained through analysis of call data records (CDRs) produced by most modern PBXs, or supplied by your service provider. Similarly, detailed consumption measurements of the data network will provide indicators on the capacity required for the planned migration: how much bandwidth is currently used, by which applications, and what headroom is available. For both voice and data networks it is important to identify busy hours and seasonal variations to ensure that the converged network is appropriately sized.

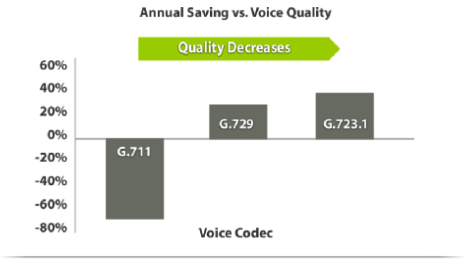
3. Undertake a capacity and cost analysis

The next step requires the analysis from the analogue voice consumption to be converted into the expected levels of digitised voice data for the converged system, taking into account voice quality requirements. The peak call load from busy hour measurements can be converted into an IP bandwidth requirement that depends on a number of factors: the percentage of calls to be carried on-net (the grade of service, or GoS); the codec to be used (which will directly impact call quality); and codec and network parameters such as coding delay and jitter.

From this, a judgement can be made on the capacity required to successfully run the converged voice and data, and whether the current capacity headroom is sufficient to cover it. If the current network is not sufficient to carry the expected converged data, decisions will need to be made on what changes are required to upgrade the network and what the cost implications of that would be. In this costing stage it is important to be mindful of the different network requirements for voice and data. The former requires delay guarantees, which typically requires the allocation of high-priority class of service (CoS) bandwidth. This bandwidth is typically charged at a premium over physical bandwidth, and service providers will typically impose structural conditions on the combination of CoS-1, CoS-2 and physical bandwidth. For example, CoS-1 + CoS-2 bandwidth may be required to be less than 50% of physical bandwidth.

4. Evaluating requirements and future

The results gathered from the planning phase will provide a consolidated set of requirements to make informed decisions on. But as with all business-critical services, the strategy does not end at deployment. Ongoing measurement and monitoring of the voice and data network and its demands are imperatives moving forward, as are maintaining quality of service and monitoring ongoing service costs. In converged networks it is important that measured and reported parameters are relevant to the type of traffic being carried. For example, reporting on the actual voice quality (for example, Mean Opinion Score, or MOS) is much more useful to the enterprise than reporting on packet delay or jitter.



Source: Sumerian Networks 2006

Realising the benefits

Voice and data convergence migration is happening now and is being implemented across many global enterprises. However, if enterprises are to realise the benefits they hope to achieve, thorough assessment of current and future business demands must be undertaken not only at the planning phase, but as part of ongoing service delivery. Without fully understanding the implications of a chosen architecture or call charging model, enterprises may find that much anticipated cost savings are swallowed up by upfront investment costs or inappropriate network and tariff arrangements To demonstrate increased productivity and cost savings, enterprises must develop and monitor their convergence strategy carefully to satisfy both business and IT requirements and to achieve the positive outcome that convergence migration promises.

More information

For further information on Sumerian or to arrange a demonstration of our services, contact us on 0141 229 7580, e-mail us at **info@sumerian.com** or visit our Web site at **www.sumerian.com**