A Roadmap To Convergence

Gary Audin

The road to convergence will be at least partially different for each organization. Every organization begins with unique assets, legacy operations and its own goals. There are, however, general considerations and decisions common to all organizations:

Existing IT Environment

1. What are the legacy IT applications, data networks and connections to the PSTN and carrier facilities?
2. Which applications will remain in their present form, and which will move to the converged environment?
3. Can the existing IT platforms, servers, PCs, workstations, operating systems and software be upgraded, or do they need to be replaced?
4. Are the LAN and WAN ready for convergence? Is their performance adequate for voice and video transmission?
5. How will the vendor(s) help in the migration to newer technical solutions and standards?

Financial Issues

6. How many of the assets of 1 above will be retained?
7. Does the total cost of ownership (TCO) reduce costs while improving the business process?
8. Does the return on investment (ROI) deliver positive business results?

Staff Considerations

9. Will the user and customer interfaces increase productivity?
10. Are the IT and communications staffs educated and trained well enough to implement a converged environment?

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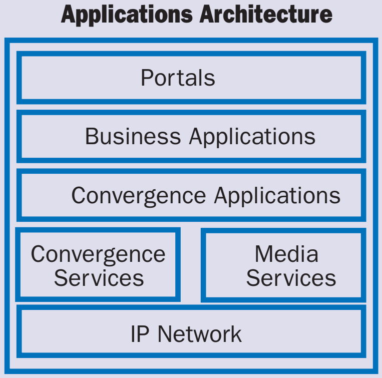
FIGURE 1 Convergence Applications Architecture



Convergence Applications Architecture

All of the approaches to construct a converged environment must recognize the fact it will be composed of a set of hardware, software and transmission building blocks. No single vendor will be able to provide all the components and services. Figure 1 presents five levels of functionality for the implementation of a converged environment.

This architecture diagram describes a roadmap for deploying converged applications and networks. The five levels have varying degrees of: availability; standardization; commodity offerings vs. custom design; generic vs. vertical market products; in-sourcing and outsourcing.

Each level requires corresponding elements that offer common:
1. Layered security features
2. Systems and network management for multiple sites with element management systems (EMS)
3. Database services and administration
4. Application builder tools that enable the creation of applications that straddle traditional boundaries (e.g., virtual assistant).

The hardware and software used for construction should be based upon open standards, Linux-based and SIP-enabled servers, routers and gateways. End users should be able to access applications through public and private (in-house) wired and wireless networks.

Network Convergence

A common method for defining the stages of convergence is to start, at the bottom of a diagram, with the IP network, because the network will be based on the IP protocol family. IP networks are application independent, and were originally designed to

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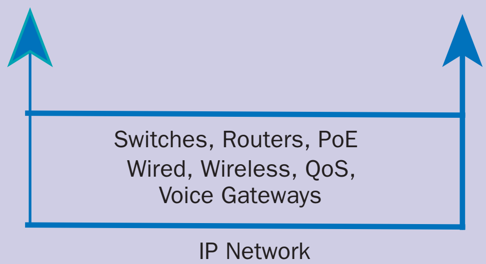
treat all traffic the same—there was no quality of service (QOS) differentiation among traffic. However, a converged network must provide the performance required for various types of traffic, which will need different levels of QOS. The components are illustrated in Figure 2.

FIGURE 2 Network Convergence



The network components include LAN switches, routers and wired transmission facilities. Recent additions to the mix are wireless transmission (cellular, wireless LAN), firewalls, power over Ethernet and QOS. With convergence, the mix will include voice gateways connecting the IP network with legacy phones, faxes and other devices, as well as T1 and PRI trunk transmission facilities. These gateways also introduce voice and video signaling protocols such as H.323 and the Session Initiation Protocol (SIP), which will be discussed in the "IT Infrastructure" section of this article. Other support standards at this level should include the Lightweight Directory Access Protocol (LDAP), Java Database Connectivity (JDBC) and Simple Network Management Protocol (SNMP).

The major task for this level is preparing the data network for the QOS requirements of voice and eventually video traffic. Today's public switched telephone network (PSTN) delivers excellent voice quality (Table 1). To ensure acceptable voice quality in converged networks, five performance issues must be addressed: transmission errors, one-way delay, delay variance (also called jitter), packet loss and out-of-sequence packets. All these impairments are far more severe on the WAN; LAN performance may not be an issue because LAN switches with high bandwidth (10/100 Mbps) generally eliminate these problems.

Voice over IP (VOIP) speech can tolerate poorer conditions than those encountered on the PSTN and still deliver voice quality that sounds the same as the PSTN to the average listener. Managed IP and private intranets can provide the network performance required for VOIP, but they must be tested and possibly modified before voice traffic is carried. Three solutions are available for improving performance:

1. Reduce the data traffic (not likely).
2. Increase bandwidth.
3. Add QOS features to the LAN switches and especially the routers.

The likely QOS features offered by LAN switch vendors are IEEE-standard 802.1p for QOS and 802.1q for virtual LAN (VLAN) support, allowing voice and data traffic to run on separate VLANs. Differentiated Services (DiffServ) is the likely choice for router/WAN QOS. Finally, the Internet can deliver adequate performance for VOIP, but the uneven range of delivered performance makes it hard to predict at what moment the Internet will be adequate for any particular call. It turns out that jitter will be the most difficult impairment to fix on the Internet. At this level, standards adoption and interoperable products and services are mandatory.

IT Infrastructure Convergence

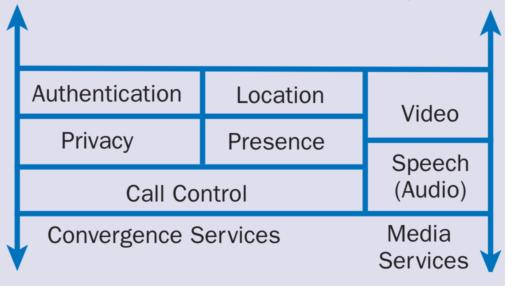
The next level above Network Convergence, IT Infrastructure Convergence, delivers a series of services implemented in end devices such as servers and workstations. Elements of the IT infrastructure will also be part of the IP phones, voice gateways and firewalls (Figure 3, p. 11).

These services are divided into two parts, Media Services and Convergence Services. Media Services are further subdivided into speech (voice) and video services. The digitization of uncompressed speech for transmission over the IP network should be implemented using the G.711 standard, operating

TABLE 1 TDM vs. IP Networks

Factor	PSTN	VoIP Speech Tolerance	Managed IP or Intranet	Internet
Errors	Very low and ignored	Ignored; No retransmission	Low Corrected by retransmission	Low Corrected by retransmission
One-way Delay	1–30ms	40–120ms	20–200ms	40–2,000ms
Delay Variance (Jitter)	0–5ms	10–25ms	10–75ms	10–100+ms
Packet Loss	0%	1–4%	1–5%	1–30%
Out-of-sequence packets	Does not occur	Correction required but adds to delay	Correction required but adds delay	Correction required but adds delay

FIGURE 3 IT Infrastructure Convergence



at 64 kbps. The most common standard for compressed voice is G.729. In the future, the G.722 standard for high fidelity (broadband) speech should be part of the voice services. All vendors appear to be compatible with G.711, but different vendors' implementations of G.729 and G.722 may not interoperate because of the differences that exist in standard versions and software implementations.

Video support will require compression, because uncompressed video demands bandwidth far greater than any converged network can economically support. The compression standards for videoconferencing over ISDN (H.261 and H.263) have been adopted for use with IP converged networks. An emerging standard, H.264, will offer greater picture resolution with reduced bandwidth requirements. H.264 has been introduced by a few vendors, but is not commonly used.

The Convergence Services (CS) include five components: call control, privacy, presence, authentication and location. Convergence Services also supports Call Center XML (CCXML) and Voice XML (VXML).

Call control is performed by signaling protocols. There are four choices:

1. ITU's H.323, versions 1, 2, 3 and 4.
2. SIP from the IETF.
3. MGCP from the IETF.
4. MEGACO/H.248 from both the ITU and the IETF.

Most VOIP vendors introduced H.323 products first, many using version 1, which is now obsolete ("Too Many Protocols", p.12). Other vendors developed non-standard versions of H.323. As the VOIP industry matured, the Session Initiation Protocol (SIP) became more attractive. SIP is a better platform for converged applications, and any roadmap to convergence must either include SIP, or else the non-SIP vendor must provide a migration path to SIP. Table 2 provides a comparison of the two protocols. MGCP and H.248 do not appear to be primary choices.

Convergence Services provides four other functions:

1. Authentication—Software combined with a directory that ensures that the accessing devices and users are allowed to interface with the portals using a single sign-on.
2. Location—The ability of the network to determine where a device/user are located.
3. Privacy—The ability to limit knowledge of the accessed services to the user and to the network and applications management. This is especially important because of legal requirements such as the Health Insurance Portability and Accountability Act (HIPAA).
4. Presence—The implementation of the SIMPLE (SIP for instant messaging) standard shared across all applications with capabilities such as context, instant messaging (IM) and media preference, and integration with Lotus and Microsoft servers.

TABLE 2 H.323 vs. SIP

FACTOR	H.323	SIP
Design	Complex (736-page spec)	Simple (128-page spec)
Number of Elements	100s	37
Messages	Based on ASN.1	HTTP and RTSP
Call Setup	Multiple Requests	Single Request
Extensibility	Non-standard	Use Session Description Protocol
Large Phone Number Domains	Designed for LAN	Designed for IP Networks
Conferencing	Limited	Open to all sizes
Firewall Support	Difficult	Easier
Interoperability Among Vendors	Poor	Good

Business Applications

Business applications have been accessible through various IT devices for decades. However, many designs of business applications have actually been segregated structures that could work in parallel but not in a unified manner. The user had to switch and sign on to multiple interfaces, most of which have different appearances and logic. The goal of convergence is to embed communications into these business applications. This makes the workflow easier, simpler and more effective. This will have to combine converged access tools with software from third party vendors. Examples of this approach are:

1. Groupware integration with IP messaging
2. Lotus Messenger integration
3. Presence server integration with Microsoft Live Communications Server (LCS)
4. Standards adoption such as XML,VXML and Java
5. Common subscriber database and administration

Too Many Protocols

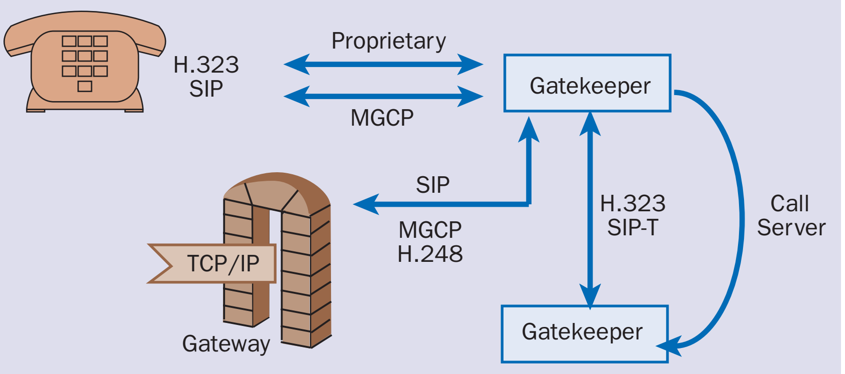
VOIP began with one standard signaling protocol, the ITU's H.323 version 1 in 1996. Version 1 worked, but inefficiently, stimulating an upgrade to produce version 2 (1998), which now is the common choice. Version 1 is not upward-compatible with versions 2, 3 and 4. Some vendors that entered the VOIP market later have chosen version 3 (1999) or 4 (2000).

While this was transpiring, Telecordia (the former Bellcore) and Level 3 developed proprietary protocols that were combined and expanded to create the Media Gateway Control Protocol (MGCP), another standard signaling protocol. Many vendors, however, used H.323 as a model and developed proprietary IP phone signaling protocols, the best-known of which is SCCP (Skinny) from Cisco.

Independently, the IETF has developed the Session Initiation Protocol (SIP), with a draft document in 1996 and formal proposal in 1999. SIP shares a number of design elements found in other IP-related protocols and can be used with the family of existing standard IP-related protocols. A variation called SIP-T was designed for gatekeeper (i.e., call server)-to-gatekeeper signaling.

The ITU worked with the IETF and effectively combined an advanced version of H.323—called H.GCP—with MGCP, producing MEGACO, which has evolved into the ITU standard H.248. Figure A shows where all these protocols work in the network□

FIGURE A Protocol Usage



Portals are used to access these business applications. In a sense, the business applications present a unified structure whereby the user/customer can be productive with multiple applications simultaneously, without having to continually switch from screen to screen.

Portals

The portal is the audio, visual and tactile interface between the user and the converged environment. It is the top level of Figure 1. A major feature of the portal is to provide a standardized user interface (UI) across the organization. The result is reduced training and increased usage.

The portal design should deliver a consistent user experience in different contexts. This is accomplished through different media (data, voice and video) that will produce a "Unified Client" with higher productivity. The portal design should integrate business applications and accelerate application adoption. Poor design will limit the success, and the justification for the new applications will not be realized. Some of the design goals for the portal are:

1. Unified graphic user interface (GUI)
2. Single sign-on access to the services
3. Wireless clients (local and cellular)
4. Speech assistants for true voice-in, voice-out access to applications

New Construction

This roadmap has identified many areas to be considered on the path to a converged environment. This path will be never-ending; new software, hardware, development tools and vendors will always be candidates for inclusion. The best design will build upon the five levels outlined. Adherence to standards and avoiding single-vendor proprietary solutions will ensure longevity.

Additions to the roadmap that will be important as converged environments mature should include:

1. Speech applications, probably delivered in partnership with third-party vendors.
2. Mobile applications support with:
- Standards-based security—802.1i.
- Standards-based QOS—802.1e.
- Handoff between access points in different subnets.
- Dual-mode devices to work across carrier and in-building networks.

3. Business process integration with:
- Microsoft Exchange and Lotus Notes with IP messaging and IP conferencing.
- Lotus SameTime Web Collaboration.
- Enterprise resource planning (ERP) and customer relationship management (CRM) applications (e.g., Siebel, SAP etc.)