Overview

It’s an opportune time for smaller or emerging market collaboration service providers to jump into next generation audio conferencing technology. This white paper describes the equipment and features new conferencing service providers (CSPs) need to get started, perhaps starting small and expanding over time.

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Executive Summary

It’s an opportune time for up-and-coming conferencing service providers (CSPs) to jump into conferencing solutions based on VoIP architectures. Early adopters have demonstrated the viability and cost-efficiencies of these technologies in large hosted conferencing deployments.

The transition away from legacy TDM conferencing bridges toward Next Generation Network (NGN) audio conferencing is presenting new opportunities for CSPs. It gives them more control over the expenditures, size and feature-mix of their conferencing systems, while dramatically lowering cost per port. These benefits are enabling CSPs to better serve the hosted conferencing needs of emerging economies (e.g., BRIC countries), small businesses and social networking.

This white paper describes the equipment and features new conferencing service providers need to get started, perhaps starting small and expanding over time. The basis is an easy-to-use, scalable and flexible solution that supports all the necessary functionality. The time is ripe to adopt NGN audio conferencing technology and get to market faster with a modern, cost-effective conferencing solution.

Market Trends

Somewhat of a paradox, the tough economic environment is driving higher audio conferencing demand. According to Frost & Sullivan, "Businesses in 2009 continued to feel the pressures of a tough economy across all the geographies, with the U.S. alone shedding about 12 million jobs. Though audio conferencing usage was high, as conference calls replaced travel, businesses were intent on cutting conferencing costs.”

Even with minutes increasing, CSP revenues still tend to be flat because price per minute is going down and squeezing profits. More encouragingly, Frost & Sullivan sees higher growth rates for web conferencing and desktop video, which will propel the use of audio conferencing services, driving momentum in the CSP market. Facing intense price pressures, incumbent and new CSPs need modern, cost-efficient conferencing solutions in order to be competitive.

High-Level Service Provider Requirements

In addition to equipment acquisition cost, foremost on the minds of CSPs is how well systems can adapt to changing network and competitive conditions. Per se, Frost & Sullivan identifies six categories of critical product features of conferencing solutions, listed in Table 1. This whitepaper suggests how these features can help achieve key goals of CSPs.

Minimizing the acquisition cost of a conferencing solution is clearly very important, and a fundamental metric is price per port, a measure of what CSPs are getting for their money. Thorough CSPs also think long term and consider the total cost of ownership of a solution. Integration is another key goal, whether a CSP has existing infrastructure or plans to add systems at a later date. In both cases, standards conformance improves the interoperability between

Table 1.

<table>
<thead>
<tr>
<th>Features</th>
<th>Goals of Conferencing Service Providers</th>
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<tbody>
<tr>
<td>Cost</td>
<td>Increase competitiveness</td>
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<tr>
<td></td>
<td>Lower price points to market entry</td>
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<tr>
<td>Integration</td>
<td>Avoid getting stuck with a solution that becomes obsolete or can’t be enhanced or expanded</td>
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<td></td>
<td>Prepare for any-to-any network configurations with standards-based equipment</td>
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<tr>
<td>Scalability</td>
<td>Service growth without performance degradation</td>
</tr>
<tr>
<td></td>
<td>Minimize the outlay to expand capacity</td>
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<tr>
<td>Return on Investment</td>
<td>Implement solutions faster, improving time to money</td>
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<tr>
<td></td>
<td>Deploy new services, like video conferencing, at minimal cost</td>
</tr>
<tr>
<td>Security</td>
<td>Safeguard networks and users</td>
</tr>
<tr>
<td></td>
<td>Block unauthorized calling with fraud protection</td>
</tr>
<tr>
<td>Personalization</td>
<td>Provide service differentiation</td>
</tr>
<tr>
<td></td>
<td>Adapt quickly to customer requirements &amp; market shifts</td>
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system elements and enables CSPs to leverage supplier competition and ultimately minimize their infrastructure cost.

Scalability is a crucial feature for new CSPs entering the market, giving them the ability to start small and later expand in response to growing customer demand. Similarly, greater control over capacity enables CSPs to maximize system utilization as well as return on investment (ROI), a primary goal.

Security features—a cost of doing business—must adapt to emerging threats to users and the network. Today, more and more hackers are making illegal calls on CSP networks, hence there’s growing interest in fraud protection. Personalization capabilities enable CSPs to create new services or features that help retain or attract customers and create a competitive advantage.

**Traditional Audio Conferencing Solutions**

Despite the worldwide commercial success of NGN networks based on Voice over IP (VoIP) technology, aging TDM service architectures are still in place, severely limiting a service provider’s market agility to introduce new services.

The traditional way for CSPs to service public switched telephone network (PSTN) customers was to use TDM audio bridges that connected to the PSTN via T1/E1 trunks, as shown in Figure 1, which aggregate conferencing voice circuits. TDM audio bridges are single-purpose elements, typically hard-coded, which give CSPs few options to personalize their systems aside from making special requests of the equipment manufacturer. As a result, adding new features (e.g., branded, self-service web-based GUIs) can require a considerable amount of time, on the order of six to nine months.

In terms of cost, TDM audio bridges are more expensive per port than comparable VoIP solutions, partly because they have roughly one-fifth the port density, while also consuming much more rack space. Moreover, TDM bridges often experience performance degradation when used with some VoIP elements, due to higher latencies caused by TDM and IP-based systems processing data differently. This can create integration issues as CSPs look to adopt new networking technology or deploy services that mix a wide assortment of media (e.g., voice, data and video), which is far easier with packet-based network elements.

Scalability options are limited since TDM bridges are also monolithic, supporting conferencing service logic, media processing and network interfaces altogether, as shown in Figure 2. For example, a CSP wanting to expand just the number of ports (i.e., network interfaces) has to buy an entire TDM bridge and possibly pay for unnecessary capacity in other areas, like media processing. Another drawback is many TDM bridges are not true carrier-grade NEBS compliant, which may be a concern for CSPs trying to win business from discriminating enterprise customers.

**NGN Integrated Conferencing Solutions**

Next generation networks (NGN) reflect the migration away from TDM circuits carried by E1/T1 lines to voice streams transferred as data packets traversing over IP networks. Consequently, an IP-based conferencing solution is naturally suited to service any IP-based endpoint such as an IP Soft Phone on a PC, an IP...
The interface point between the CSP’s trusted IP service network and external IP networks is managed by a session border controller (SBC), which provides security, access control and network address translation (NAT) for clients entering the service provider’s VoIP network. IP-based technology also allows self-service web-based interfaces and VoIP audio conferencing to be performed on the same platform, as illustrated in Figure 3.

However, many CSPs have a large installed base of customers still using circuit-based (i.e., TDM) phones, or 2G cell phones. Supporting existing customers on the PSTN is accomplished by adding media gateways to convert PSTN circuits to VoIP packet streams, as depicted in Figure 4.

The resulting IP-based conferencing system enables the users and subscribers of conferencing services to interact with the service via two methods:

- A TDM and/or SIP endpoint for audio interaction
- An IP data connection for web interaction

### Main Components: Integrated Conferencing Solution

The main components of an IP-based conferencing solution are the media gateway, the application server and the IP media server, as shown in Figure 5 and described in the following:

- **Media Gateways** (line and trunk cards) are deployed at the border of circuit-based access networks, like the PSTN or 2G cellular networks, and the IP core. The primary function of a media gateway is to convert media circuits on one side into Real-time Transport Protocol (RTP) over IP media packet streams on the other side.

- **Application Servers** (features and applications) host the conferencing call flow and service logic; provide database access and IVR dialog logic; support billing and provisioning; and coordinate with Web clients for real-time conference control.
IP Media Servers (media packet processing) connect to an IP network and are controlled by the application server. They perform media processing functions such as media bridging, mixing and format transcoding, acoustic echo cancellation, noise reduction, announcements, IVR, recording and playback, automatic speech recognition (ASR), text to speech (TTS), fax handling and video processing.

Delivering convenience and lower overall cost, some network carriers are supporting the media gateway function (and TDM interfaces), thus allowing CSPs to connect to the carrier network using cost-efficient Session Initiation Protocol (SIP) trunking. In this case, the CSP also simplifies their external network integration by focusing solely on the session border controller to support the IP boundary security requirements (Figure 6). Since carrier-supported media gateways are cost-efficient for SIP-based trunking deployments, gateways are often outside the scope of an IP-based integrated conferencing solution.

The Radisys Integrated Conferencing Solution

An IP-based architecture, using the Session Initiation Protocol (SIP) for call control, delivers to service providers the business model flexibility that was inconceivable with traditional audio bridge equipment and TDM networks. This approach supports more predictable and cost effective scalability, because the conferencing functionality can be easily distributed among different components. For example, application logic can be centralized at a single location (or possibly two geographically distributed sites), while the media gateway function is distributed as close to the subscribers as possible. These elements scale independently, allowing CSPs to dial-in the right amount of capacity where required for features, applications and media processing.

Supporting this model, the Radisys Integrated Conferencing Solution consists of an application server and the IP media server, whose functionality is divided into three components, as shown in Figure 7. It should be pointed out that this solution architecture is based on modular components and open standards-based interfaces and APIs, which facilitate reconfiguring the system to deploy other services. For example, the exact same components can implement new features, like IVR applications, contact centers, video or messaging applications. In contrast, TDM audio bridge equipment is typically single-purpose, delivering only audio conferencing services based on a propriety solution.

The Radisys SIPware Conferencing Application, shown at the top of Figure 7, offers customizable and turnkey SIP-based services, including operator-assisted event and reservationless audio conferencing:
1. Operator-Assisted Event Conferencing
(i.e., large scale events, like seminars or financial earning updates)
- An operator is available to support participants throughout the event
- Offers a higher level of assurance and reliability to the call
- Ideal for one-off events, where participants may be using the conferencing service for the first time

2. Reservationless Conferencing
(i.e., ad-hoc, meet-me meetings)
- High-growth conferencing service with basic, easy-to-consume services, allowing the participant to self-manage meetings beginning to end
- No prior conference booking is required, and meetings can be conducted day or night
- Service activation is often automated and accessed using easy-to-use web-based GUls
- Suitable for small groups or those using conferencing services frequently
- Typically less expensive than operator-assisted, since no operator interaction is required

The SIP-based software runs the Radisys RapidFlex Platform, a fault-tolerant, multi-site services architecture providing carrier-grade reliability and exceptional manageability.

Delivering feature-rich audio mixing under control of the SIPware applications, the Radisys Convedia Media Server supports industry-leading features, scalability, reliability and performance.

While TDM audio bridges in the field are approaching end-of-life, with decreasing levels of support from a shrinking pool of suppliers, the Radisys Integrated Conferencing Solution offers a path forward into next-generation IP-based communication networks.

### Problems Solved by the Radisys Solution

Smaller and emerging market collaboration service providers are looking for easy-to-use, scalable and flexible conferencing solutions that support all the necessary functionality. These needs, previously divided into six categories, are all addressed by the Radisys Integrated Conferencing Solution based on modern VoIP technology.

1. **Cost**: The modularity and high port density of IP-based conferencing components helps the Radisys solution reduce the cost per port to about half that of a typical TDM bridge solution. It’s also possible to implement SIP Peering to take advantage of the lowest available long distance costs. In addition, CSPs can start with a relatively small, low-cost solution as described in the following scalability section.

2. **Integration**: Enabling CSPs to avoid getting locked into a proprietary solution, the Radisys solution is truly standards-based, as demonstrated by support for:
   - **Open Standard Protocols**: IP, SIP and MSML
   - **Network Architecture Standards**: RADIUS and DIAMETER-IMS compliant
   - **Software Standards-Based**: C, JAVA, SOAP, XML, Oracle, HTTP and SNMP
   - **Operating System**: Linux running on a variety of Linux-based compute platforms

3. **Scalability**: The Radisys Integrated Conferencing Solution can be deployed on a single 1RU Linux server, left side of Figure 8, providing an economical starter system for lab trials or small deployments. However, the modularity of the architecture also allows the solution to scale to large, multi-site distributed systems supporting 50,000 or more simultaneous calls, as shown on the right side of Figure 8. Leveraging this solution, CSPs can start small, yet grow their capacity with confidence.
The Radisys Integrated Conferencing Solution is also highly reliable and carrier-grade, integrating the latest failover and availability technologies. The system consists of clustered and redundant NEBS components for high availability and no single point of failure. Further resilience can be achieved by geographic distribution of redundant application servers designed for instant failover protection, and replicating databases across geographically distributed sites; thus multiple sites can supersede in the case of component failure at other sites.

4. Return on Investment: CSPs can be up and running in no time since Radisys offers a comprehensive conferencing solution that improves time to money. In the future, CSPs will also be able to add video conferencing capabilities without hardware upgrades, thereby protecting their investment.

5. Security: The Radisys Integration Conferencing Solution is based on Linux, which offers extensive security and platform management capabilities, proven in large-scale service provider deployments.

The Radisys SIPware Conferencing Application also supports fraud prevention capabilities, such as protecting against unauthorized individuals using a calling card to dial out to international destinations. The system disconnects the caller after a specified number of invalid PIN attempts and blocks future PIN fraud attempts from the same telephone number.

6. Personalization: As a standards-based implementation, the Radisys solution enables the integration of various 3rd party solutions, providing applications flexibility, including:

- Web-based GUI customization for subscriber or operator interfaces
- APIs for integration with common desktop productivity meeting scheduling tools, web conferencing services, or web-based integration with social networking websites looking to add audio conferencing services
- Easy integration with Enterprise Unified Communication Systems

Compared to TDM audio bridges, the Radisys solution costs less, requires less space and energy, and offers more integration flexibility and scalability options.

**Move to IP-based Conferencing Solutions**

The transition away from legacy TDM conferencing bridges to next-generation IP-based platforms is presenting new opportunities for conferencing service providers. They have more control over the cost, size and feature-mix of their conferencing systems and can dramatically lower cost per port. Likewise, the Integrated Conferencing Solution from Radisys was designed to meet the needs of CSPs wishing to start with smaller entry-level systems, but with the capabilities to service the mass market with respect to cost, integration, scalability, ROI, security and personalization. This new watermark is ideal for smaller and emerging market collaboration service providers who want to go after new opportunities in more diverse and cost sensitive markets.
Glossary

The following Glossary is in the order of the acronyms appearing in the paper.

**AS:** Application Server
**ASR:** Automatic Speech Recognition
**CDR:** Call Detail Record
**CSP:** Conferencing Service Providers
**BRIC Countries:** Brazil, Russia, India, and China
**E1:** European Data Circuit Running at 2.048 Mbit/s Line Rate, supporting 30 voice circuits
**GUI:** Graphical User Interface
**IP:** Internet Protocol
**IVR:** Interactive Voice Response
**MRF:** Media Resource Function
**MS:** Media Server
**MSML:** Media Server Markup Language (RFC 5707)
**NAT:** Network Address Translation
**NGN:** Next Generation Network
**PSTN:** Public Switched Telephone Network
**VoIP:** Voice over IP
**RTP:** Real-time Transport Protocol
**SBC:** Session Border Controller
**SIP:** Session Initiation Protocol
**SNMP:** Simple Network Management Protocol
**TDM:** Time Division Multiplexing
**T1:** Data Circuit Running At 1.544 Mbit/s Line Rate, supporting 24 voice circuits
**TTS:** Text-to-Speech
**XML:** Extensible Markup Language

References