White Paper | July 2009

radisys

Next Generation Network

Audio Conferencing Business Case

Overview

This white paper will help service providers understand the benefits and economics associated with implementing a Next Generation Network (NGN) audio conferencing service using IP Media Server technology. The paper will start by comparing and contrasting TDM with IP-based service architectures, and then continue with an overview of NGN audio conferencing. A detailed business case model will be used to help determine and summarize the economic benefits of the IP-based approach by exploring two NGN service deployment scenarios.

Audience

This white paper is intended for network operations staff, senior level network operations management, and product (services) marketing staff at service providers looking to implement next generation network infrastructure.

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Executive Summary

Both traditional and emerging telecommunication service providers are facing some challenging business and economic times. Most are looking to increase their revenues while reducing infrastructure budgets—i.e. doing more with less. An industry-wide movement to deploy IP-based network infrastructure and services architecture is now opening up new markets and revenue opportunities. This Next Generation Network (NGN) approach enables service providers to rapidly introduce multiple, high-value services to their networks to increase revenues by attracting new subscribers and keeping them.

In spite of the commercial success of NGN networks around the world, aging TDM service architectures are still in place, severely limiting a service provider's market agility to introduce new services. The advent of Voice over IP (VoIP) technology, open protocols, and distributed architectures have created a revolution in how both core and enhanced services are implemented. VoIP was coined as an industry term to represent both packetized voice and call signaling. The Session Initiation Protocol (SIP) emerged to become the signaling protocol of choice from inside the core network all the way out to end user devices. These network elements were built around IP protocols enabling a new wave of innovative product vendors to build network elements that adhered to these protocols. This fundamentally changed the way all voice services would get delivered.

So for service providers the question is—when is the right time to switch to an IP services architecture? This white paper will show what investment is needed, the benefits of making this decision sooner rather than later, and the opportunity to deliver on the promise of profitably creating a bundle of multiple services that addresses new market opportunities.

Trends in Service Infrastructure Technologies

TDM Service Architectures

At the core of traditional circuit-switched voice service networks were the Class 4/5 switch (referred to as C5 switch in the rest of this document). C5 switches were monolithic devices built from the ground up on proprietary hardware and software components much like a mainframe computer. All of the functions necessary to run the service logic and connect to the network were built into these switches (Figure 1). These functions fell into one of four categories:

- Service Logic
- Signaling and Call Control
- Network Interfaces
- Media Processing

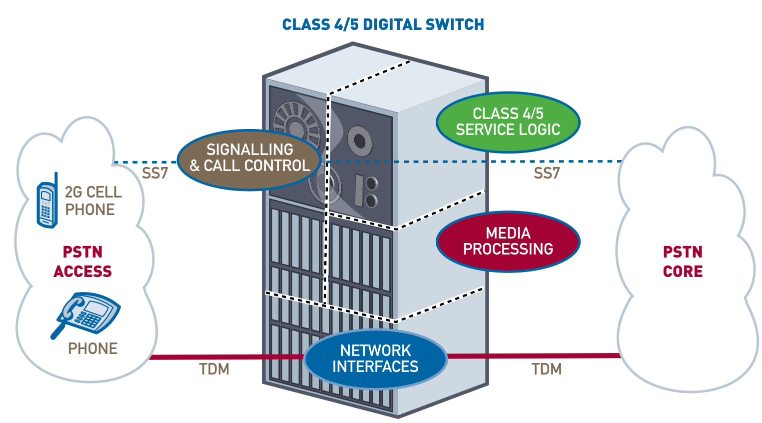

Figure 1. *TDM Service Architectures–The Four Key Funtions of Class 4/5 Digital Switch*

Proprietary programming languages were used to build out the service logic by accessing these internal functions. These products were very complicated, real-time call processing engines with limited computing resources that made it very difficult to add new features. And only the product vendor's development teams had access to these switch internals. This meant service provider requests to C5 switch vendors for customizations and new features were queued up for development and test cycles that could take anywhere from 12-18 months to complete. And the costs were often exorbitant.

This service inflexibility forced service providers to implement AIN-based services. AIN did deliver on its promise to bring incremental services to C5 switch environments, such as toll-free services, cellular roaming, or call screening, but it did not improve the economics of deploying services. And even though the industry saw the first Service Creation Environments, these development environments were tied to specific products that cost millions of dollars.

Due to the high costs for adding new services to the C5 switches, and the failure of AIN to improve service introduction economics, a market developed for product vendors to create dedicated enhanced services platforms (Figure 2). These platforms delivered highly functional services but they were limited in how they could interact with service logic running on a C5 switch.

Whereas AIN services were designed to integrate with C5 service logic through the use of formalized application triggers and database requests, these enhanced service platforms had to rely on call states such as unconditional call forwarding or forwarding the call on a "ring no answer" to get invoked.

And once again this approach required duplication of network interfaces and media processing resources that added cost and complexity to the network. This perpetuated stovepipe service architectures but provided service providers some short term relief in their quest for a more open service architectures with some customization possible.

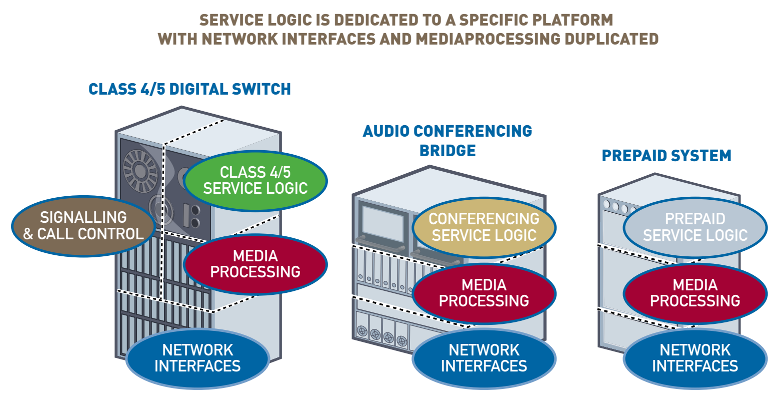

Figure 2. *TDM Service Architectures–Adding Enhanced Services Platforms*

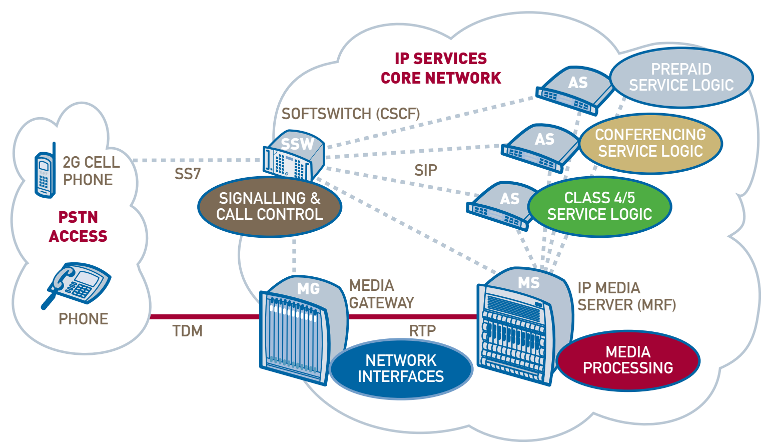

Figure 3. *IP Service Architectures–Multiple Services Sharing Network Interfaces and Media Processing*

IP-based Services Architecture

IP-based communication components used in NGN networks have made it possible to deliver on the promise of lower network costs and service flexibility. The foundation of NGN networks, including IP Multimedia Subsystem or IMS, lies in the decomposition of key functions of the C5 switch and service architecture into separate network resource components so they can be shared by all network services using open IP communication protocols (Figure 3). This has resulted in the following mapping of TDM C5 functions:

- *Call Control & Signaling* became the Softswitch
- *Network Interfaces* became the Media Gateway
- *Service Logic* became the Application Server
- *Media Processing* became the IP Media Server

This architecture enables both core network and enhanced services. There is no need to classify services as Class 4 or Class 5 services as in the TDM model because all service logic now runs on separate Application Servers, regardless of whether it's used to simply place a call or support a bundle of multiple services on a single call. And because Application Servers use industry standard server hardware and software components, this software environment is much more open and accessible to service provider and third party development teams.

This approach also supports scalability in a more predictable and cost effective manner since only the components incorporating the resources that are required need to be added to the network. This has dramatically changed the way service provider networks are traffic engineered and provisioned.

The IP Media Server is a slave media processing device controlled by a call agent such as the Softswitch or Application Server. The Media Server performs media processing manipulations on Real Time Protocol (RTP) streams such as announcements, IVR, recording and playback, bridging/mixing, automatic speech recognition (ASR), text to speech (TTS), fax handling, and video processing. Because a Media Server does not contain any application logic, it is inherently reusable for all media processing tasks in an IP services network.

The IP Media Server is controlled by a Softswitch or Application Server using a control protocol. The large majority of the market today uses SIP for Media Server control, often augmented with an XML-based scripting language like VoiceXML (for IVR dialogs), or Media Server Markup Language (MSML—suitable for feature-rich Media Server control—especially conferencing).

IP Multimedia Subsystems (IMS)

Today's industry discussion around IMS architectures is not a revolution—instead it should be viewed as an evolution from the NGN principles and architecture outlined earlier in this section.

In an IMS environment, the Media Gateway function outlined in the earlier section is further decomposed into a number of different and specialized gateway elements, facilitating the integration of PSTN and 2G circuit-switched cellular networks, with IP-based access networks such as WiMax, WiFi, DSL, Cable Modems, or 3G wireless access networks. But the underlying principle is the same—to convert the various last-mile access technologies into a common IP services core based on SIP signaling and RTP media streams.

The Softswitch function gets loosely mapped into the IMS Call State Control Function (CSCF) elements.

The role and need for application servers is largely unchanged from the original Softswitch architecture and an IMS.

And finally, the role of the IP Media Servers gets loosely mapped into the IMS Media Resource Function (MRF), which provides a similar role of providing RTP media stream processing under CSCF or application server control.

Benefits of a Next Generation Services Architecture

The move to this pure-IP services infrastructure can help service providers to reach their goal of increasing revenues while lowering infrastructure costs, shortening the payback on their investment as we will see in business case model scenarios coming up.

NGN networks have finally emerged to support a single set of technologies that can support multiple services. The basic guiding principle of NGN network design and usage is reuse. This flexible, efficient and innovation-friendly infrastructure model is a sharp departure from the monolithic stovepipe model, and provides much greater opportunities; it enables and simplifies service application adaptations, customizations and extensions, and lets the services infrastructure simultaneously and economically support multiple services and markets. And we can all now focus on a singular "services architecture" as opposed to what we referred to as multiple "service architectures" in TDM networks.

Service Provider Investment Trends

While the technology case for the deployment of NGN network components including IP media servers is compelling, investments in traditional TDM infrastructure continued to make up the majority of service provider Capex spending in recent years. This is somewhat explained by the inertia of an industry built around TDM technology, particularly in developed nations, as well as the long depreciation cycles for traditional telecom services equipment. However, industry data is clearly showing that TDM investment spending is decreasing, and being replaced by a growing investment in NGN equipment (Figure 4).

The technology case for deployment of NGN network components including IP Media Servers is compelling. All networks are moving towards implementing a common, reusable IP communication services infrastructure. Their implementation is just a matter of time according to all of the industry analyst forecasts. In fact, recent industry data suggests we are already in the transition period where NGN investments finally exceed TDM investments.

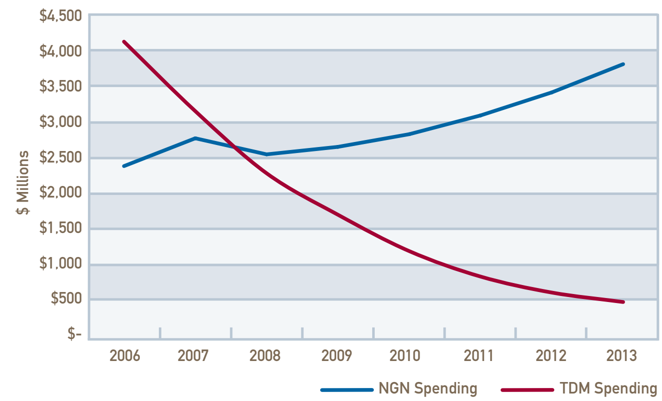

Figure 4. *Worldwide TDM/NGN Infrastructure Service Provider Spending 2006-2013 "Source: IDC 2008"*

NGN Audio Conferencing Deployments

An IP services architecture using the Session Initiation Protocol (SIP) gives service providers business model flexibility that was inconceivable with traditional TDM networks. In this section, we will describe how an audio conferencing service would map across an NGN services architecture, along with operational and deployment benefits compared to traditional TDM-based audio conferencing bridge technology.

IP Audio Conferencing has two main service models:

1. Operator-assisted large-scale Event Conferencing, and

2. Reservation-less meet-me conferencing.

Both service models can be supported on the same IP-based audio conferencing architecture, providing for re-use of key network resources.

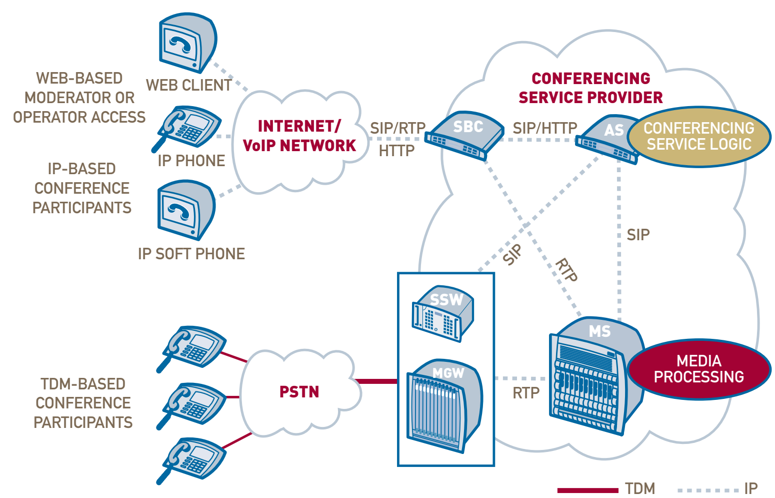

Figure 5. *IP Audio Conferencing Deployment with Media Gateway PSTN Access*

IP-based conferencing systems (Figure 5) typically enable 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

The main network components of an IP-based conferencing service include:

- *Softswitch:* for routing, call control and billing,
- *Media Gateways:* signaling and media interworking between TDM and IP networks

- *Conferencing Application Servers:* call control, database access, IVR dialog logic, and coordination with Web clients for real-time conference control
- *IP Media Servers:* media mixing, and media format transcoding

What is referred to as "the bridge" in the TDM world is a combination of functions performed by the Media Server, Application Server, and Media Gateway in an NGN deployment. The interface for VoIP calls originating in the IP network and for delivery of web interaction is facilitated using the following elements:

- *Session Border Controller (SBC):* security, access control, NAT for clients entering the provider's VoIP network
- *Web Servers:* conference control and provisioning

While the audio interaction is primarily via the TDM network today, direct interaction with VoIP clients is also possible, and actually expected to grow, facilitating capabilities that only an IP network can support, such as high-quality wideband audio codecs (such as G.722).

The Web interaction and features are delivered primarily by web conferencing servers that are part of the IP-based Application Server environment. These handle all aspects of managing the interaction of moderators and operators with Application Servers for real-time conference control.

The Conferencing Service Provider (CSP) provisions toll-free and/or local DID numbers as access numbers to terminate on Media Gateways in its core IP network. Calls to these access numbers are directed via the Softswitch as SIP commands to the Conferencing Application Servers, while the RTP media streams are terminated on the IP media server for conference mixing. The Conferencing Application Servers may also outdial (place an outbound call) to the moderator and/or participants.

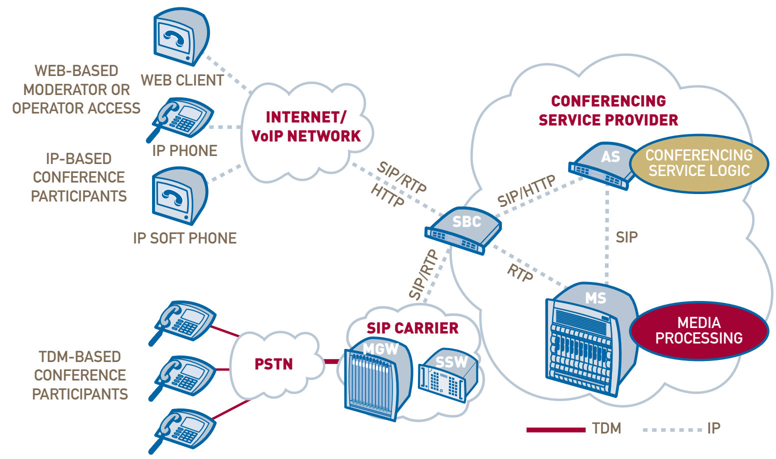

Figure 6. *IP Audio Conferencing Deployment Using SIP Peering for PSTN Access*

Alternatively, a next generation architecture also allows a second traffic termination approach known as SIP peering (Figure 6). Using this approach, the CSP would work with a wholesale long distance service provider—the "SIP carrier"—who would terminate toll free and local DID calls on their own distributed network of media gateways, and then backhaul the voice traffic back to the CSP data center through their Session Border Controller. Because the voice traffic arrives at the CSP as VoIP already, this approach saves the CSP the media gateway Capex, and ongoing TDM circuit operational costs.

IP-based conferencing solutions offer significant benefits compared to their TDM counterparts:

- *Higher density per square foot of floor space:* Complete IP-based conferencing systems with capacities of fifteen to twenty thousand ports can be implemented within a single 42U telecom rack. A single conferencing site can have tens of thousands of ports.
- *Flexible deployment models:* Media Gateways and IP Media Servers can all be sized and distributed separately from the Application Servers. The option to geographically distribute Media Gateways facilitates regional/local conversion of TDM traffic to VoIP, saving backhaul costs. Distributing IP Media Servers also allows regional audio mixing, so that backhaul voice traffic is further reduced.

- *Leverages SIP peering to establish the most economical call routes.* SIP peering also eliminates the need for service provider to purchase/host Media Gateway equipment and support the on-going expense of TDM facilities and trunking.

Evolution to Next Generation Services: IP-based conferencing provides a natural path towards providing additional next generation services. These services are key to increasing Average Revenue Per User (ARPU). Examples of additional value added conferencing services include high-fidelity wideband audio codecs, collaboration, social networking, white boarding, presence, application sharing, and unified communications. Or it could be a completely different service such as prepaid calling cards, ringback tones, IP multimedia contact centers, or hosted Unified Communication (UC) services which can reuse a large majority of the IP audio conferencing service infrastructure.

Elements of the Business Case

So how do service providers make the business case for implementing an IP services architecture? For Greenfield service providers building an IP services infrastructure from scratch, IP is the only way to go. In fact, developing countries such as China which did not have an extensive legacy TDM infrastructure, have been early adopters and seized the fast track to introduce NGN communication networks. However, service providers that have established services and subscribers built on legacy TDM service platforms faced the decision to either replace current services infrastructure, or pursue a "cap and grow" approach, where the service provider caps new investment in TDM equipment, and deploys an IP services architecture for new growth.

The main evaluation criteria for service providers making this decision are:

- Capex
- Opex
- Netex
- On-going customization
- New business opportunities
- Time to change infrastructure
- New service applications

The first three items in this list—Capex, Opex, and Netex—are cost considerations and represent quantifiable variables in the business model to deploy IP Media Servers in an NGN environment. These will be addressed in the next Section of this white paper.

The last four items are driven by internal business goals and represent cross-organizational objectives, so by there nature are a bit more difficult to quantify in costs. On-going customization is a critical need that needs budget allocated to satisfy needs identified as a service matures in the market. In a next generation infrastructure, open standards help to reduce the cost of on-going customization. New incremental business opportunities represent brand-new or adjacent markets that reach new subscribers and require adding new service features. Again, NGN infrastructures allow a service provider to introduce new services quicker, with lower entry-level costs. Time to change infrastructure is driven by many issues including the fact the TDM infrastructure may in fact be at the end of its product life cycle. So going forward at current course and speed will definitely impact growing service capacity, and possibly negatively impact current capacities and quality of service because there is no OEM support for the TDM infrastructure products.

The last item—new service applications—perhaps represents the biggest opportunity to business justify an IP services infrastructure. As we mentioned earlier, TDM enhanced services were technology silos that included of all the hardware and software needed to deliver a single service. But as we have discussed, IP communications infrastructure is built in horizontal layers that are separated by industry standard open protocols. This is in sharp contrast to the TDM approach which was in effect a proprietary vertical technology slice where no sharing of hardware or software was possible with the other vertical technology silos. In fact it was usually discouraged by the OEM suppliers in order to lock in service providers to a supplier's product and services.

Business Case for NGN Audio Conferencing

Radisys® has developed a spreadsheet-based business case model that takes into account Capex, Opex, and Netex costs for deployment of IP communication services using an NGN services architecture. This financial model takes a three year revenue forecast of audio conferencing minutes by month that is used to calculate Capex requirements from call profile inputs. It then allows for input of Netex and Opex costs to calculate service profitability and a payback period based on these assumptions.

This model focuses specifically on the facilities-based service provider deciding to purchase a turnkey conferencing solution including all of the network components needed for an NGN services architecture to support the call capacity needed for a conferencing service.

Service Deployment Scenario A

The first business case analysis is for the service deployment scenario where all of the users and subscribers of the conferencing service come into a conference call through PSTN access. This is still the most common way of accessing a conferencing service even today, but as we have seen in previous sections VoIP clients could be connected. But for this particular business model, we are tracking costs associated with deploying Media Gateways, Softswitches, Application Servers, and IP Media Servers to implement this approach (Figure 7).

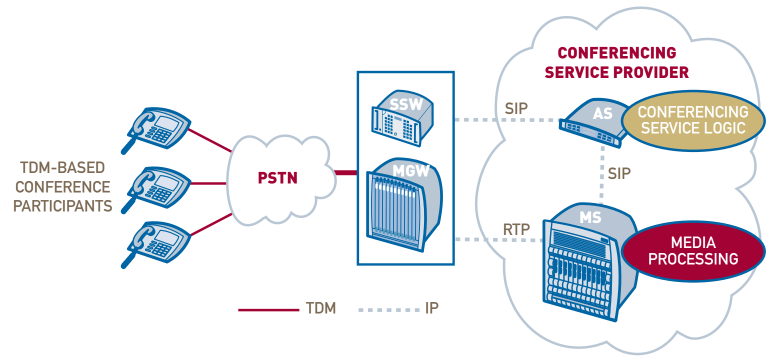

Figure 7. *Service Deployment: Scenario A*

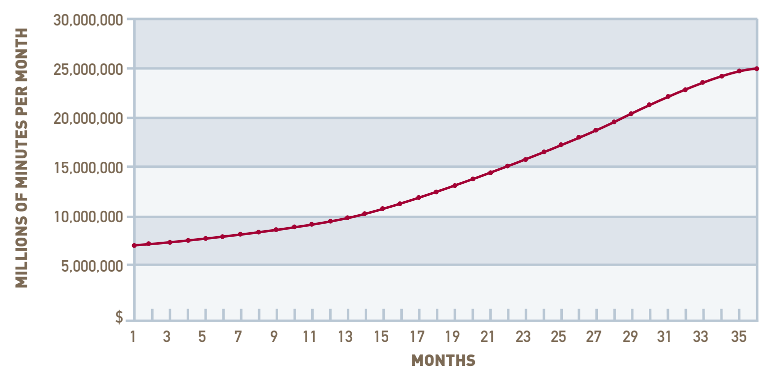

Figure 8. *Scenario A: Service Forecast Model*

Revenue Inputs

The first set of detailed inputs entered in the model is for how many minutes of use the conferencing service generates per month over a three year period.

The service forecast used in this model is shown in Figure 8.

This data is then used to calculate the service revenue for each month by values entered for each of the three years for the retail price per minute for the conferencing service. This allows the model to account for a possible decline in retail pricing over the three year period.

Capex Inputs and Calculations

This service growth forecast not only drives the revenue-side of the business case model, but also drives the underlying Media Server, Application Server, and Media Gateway capacity and Capex needed to meet the forecasted service volume growth.

By using industry standard assumptions for a typical conference call profile, capacity for each of these components can determined in terms of ports or sessions. Call profile inputs include:

- % Busy Hour Calls
- Average Conference Call Hold Time (minutes)

Pricing for the components is then specified and used to generate the initial Capex expenditure and incremental monthly Capex spending to support service growth. This establishes product costs in the model that are then used to calculate component depreciation costs based on the length of the depreciation schedule.

Opex Inputs

Opex is specified in the model by entering the following costs:

- Facilities—physical plant space, power, TDM or IP network connectivity
- Personnel salaries
- Maintenance of all components
- Installation of components
- Sales & Marketing expenses
- Subscriber acquisition expenses

Netex Inputs

The next set of inputs are for Netex costs:

- T1/DS3 Call Cost per Minute
- SIP Peering Call Cost per Minute

In this first deployment scenario using Media Gateways the T1/DS3 costs per minute will be used.

	Startup (T0)	Year 1	Year 2	Year 3
Call Traffic Inputs				
% Busy Hour Calls		45%	45%	45%
Average Conference Call Hold Time (minutes)		53	53	
Revenue Inputs				
Audio Conference Calling Per Minute Price (reservation-less, on demand)		$0.050	$0.045	$0.040
Audio Conference Service Minutes Forecast				
Capex Inputs				
Initial Capex for AS + MS + MGW Components		$1,256,437		
AS + MS + MGW Per Port Pricing		$490	$460	$430
Depreciation Rate per Year		33%	33%	33%
Netex Inputs				
T1/DS3 Call Cost per Minute		$0.005	$0.005	$0.005
Opex Inputs				
Facilities		$230,400	$230,400	$230,400
Personel Annual Salaries		$470,000	$470,000	$470,000
Other Assumptions				
Conferencing App Customization	$200,000			
Billing & OSS System Integration	$100,000			
Network Management Integration	$100,000			

Table 1. *Scenario A: Key Inputs*

Assumptions for Scenario A

Table 1 is a summary of key inputs we have chosen for this scenario. Like all business model tools, the results are based on our own assumptions that reflect our experiences. Of course any of these inputs can be changed to fit a particular deployment scenario and Radisys would be able to model your specific case by drawing on your experiences.

The Other Assumptions category includes initial project start up costs that are typical when deploying an audio conferencing service. There will always be some for of customization of the service application from the Application Server vendor chosen. Also call detail record (CDR) formats will need to be changed to conform to a CSP's billing system. And depending on the network management environment the CSP has, changes may need to be made by the Application Server and IP Media Server vendors to integrate with SNMP network monitoring environments. All of this work will require software customization and in fact may be charged for separately by the component vendors. This set of inputs accounts for this possibility.

Scenario A Business Case Analysis

The following charts are output from the business case spreadsheet based on these inputs. These first two pie charts (Figure 9) represent cost breakouts for the first year the service is in operation. Broken out first individually and then by the three major cost categories represented in the model.

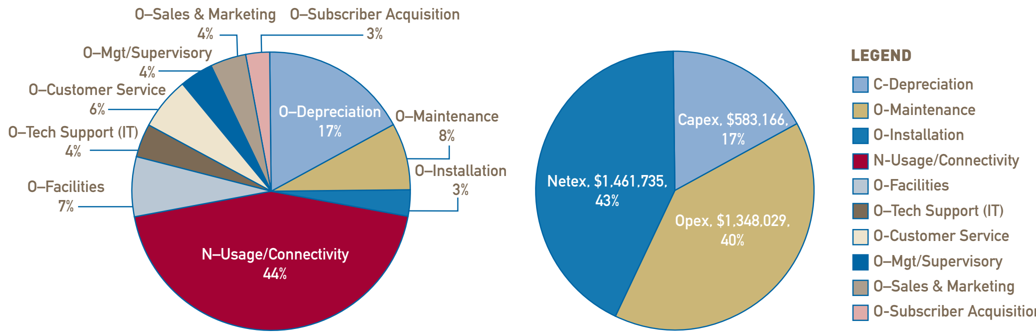

Figure 9. *Scenario A: Changes in the Cost Breakouts*

The chart in Figure 10 calculates the service gross margin for each year by combining data from both the revenue and costs for each year.

Based on the all of the inputs and calculations, scenario A yields a payback in 20 months.

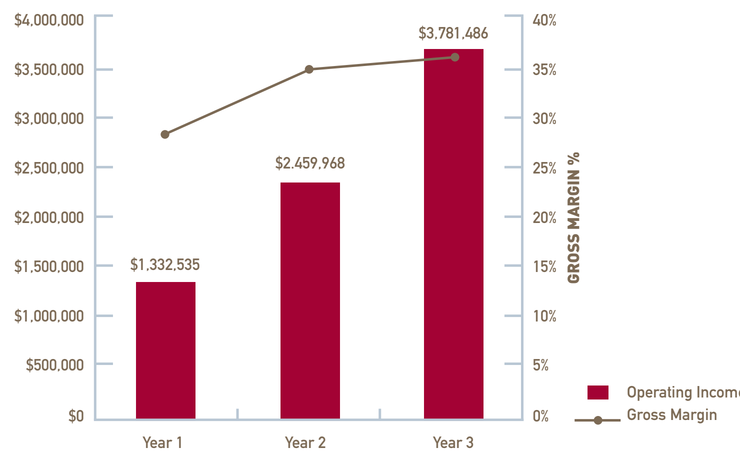

Figure 10. *Scenario A: Gross Margin and Payback*

Service Deployment Scenario B

While the majority of the discussion in this white paper is around NGN components, you'll notice that a large percentage of the costs associated with our audio conferencing business case Scenario A continues to be the network expenses. Fortunately, using a next generation architecture also allows us to leverage an economical SIP peering approach to further reduce Netex and to some extent Capex expenses. So in service deployment Scenario B the CSP is using VoIP connection to SIP Carriers to facility inbound and outbound PSTN access. From a network infrastructure standpoint this means replacing the Media Gateways with Session Border Controllers in the CSP network (Figure 11).

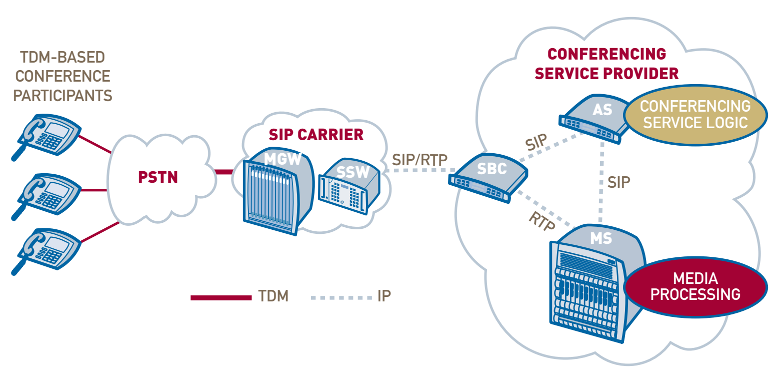

Figure 11. *Service Deployment: Scenario B*

Assumptions for Scenario B

In order to be consistent with Scenario A, Scenario B revenue assumptions are kept the same. Table 2 is a summary of key inputs for Scenario B.

Assumptions that have changed are highlighted in yellow. The changes included lower initial and ongoing component costs, slightly higher call costs per minute, lower facilities costs, and lower personnel costs.

Scenario B Business Case Analysis

The business case analysis tracks the changes as seen in the two pie charts in Figure 12. Even though the SIP Peering cost per minute is higher, the lower initial and ongoing Capex, along with lower facilities and personnel costs lower the Netex from 43% in Scenario A to 37% in Scenario B. This results in an improved payback period for Scenario B of less than 12 months (Figure 13). So SIP Peering does improve this business case model overall.

And there is one more key consideration concerning these two scenarios. Both involve deploying only one service application—audio conferencing. As we mentioned earlier, one of the key benefits of an IP-based services architecture is the reuse of common functions across multiple services. Other service revenue streams could be added to this IP communications infrastructure. And if these other services consume Media Gateway and IP Media Server resources in primarily non-concurrent usage patterns, then the business case becomes even more attractive.

	Startup (T0)	Year 1	Year 2	Year 3
Call Traffic Inputs				
% Busy Hour Calls		45%	45%	45%
Average Conference Call Hold Time (minutes)		53	53	
Revenue Inputs				
Audio Conference Calling Per Minute Price (reservation-less, on demand)		$0.050	$0.045	$0.040
Audio Conference Service Minutes Forecast				
Capex Inputs				
Initial Capex for AS + MS + SBC Components		$1,081,006		
AS + MS + SBC per Port Pricing		$425	$400	$375
Depreciation Rate per Year		33%	33%	33%
Netex Inputs				
T1/DS3 Call Cost per Minute		$0.005	$0.005	$0.005
SIP Peering Call Cost per Minute		$0.010	$0.010	$0.010
Opex Inputs				
Facilities		$115,200	$115,200	$115,200
Personel Annual Salaries		$360,000	$360,000	$360,000
Other Assumptions				
Conferencing App Customization	$200,000			
Billing & OSS System Integration	$100,000			
Network Management Integration	$100,000			

Table 2. *Scenario B: Key Inputs*

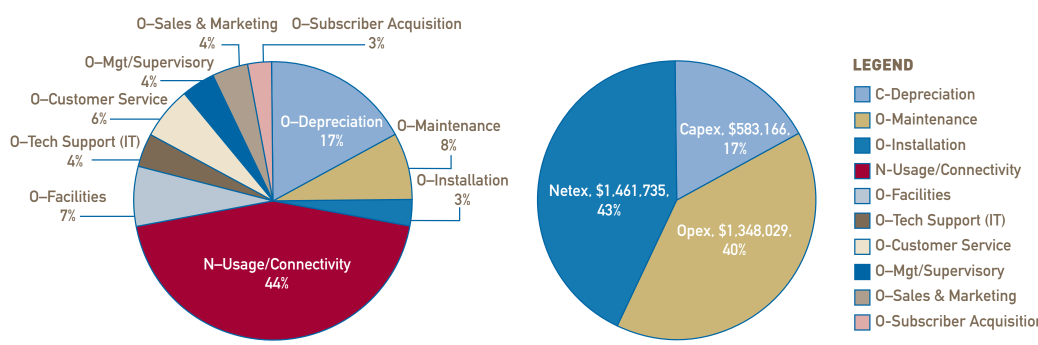

Figure 12. *Scenario B: Changes in the Cost Breakouts*

Conclusions

The separation of service logic from network resources is what enables NGN networks to achieve service flexibility that was never possible in a TDM environment. This will continue to support a revolution in how new multi-media services are brought to market by service providers. The IP Media Server is at the core of this revolution with the media processing horsepower needed to support the latest multi-media services, such as video ringback tones, video messaging applications with text-overlay, or IP contact center applications with integrated voice, video, fax, web, and speech recognition capabilities.

Key reasons to deploy an IP Media Server in an NGN services architecture are:

- Decomposition of network functions into separate components improves IP Media Server resource utilization and access
- Reduces Capex costs
  - Less equipment supporting a broader range of services
- Reduces Opex
  - Less real estate and power with very high port densities
  - Fewer specialized staff with common equipment across multiple services
  - Less training
- Reduces Netex
  - SIP Peering can be used to get the best per minute call rates
- Acceleration of new business services i.e. faster time-to-market
  - Once you have your Softswitch, Media gateways, and IP Media Servers in place, new service introductions are often isolated modifications to the application layer only, which gets new services to market faster.
  - Market trials can be quickly prototyped on top of existing resources

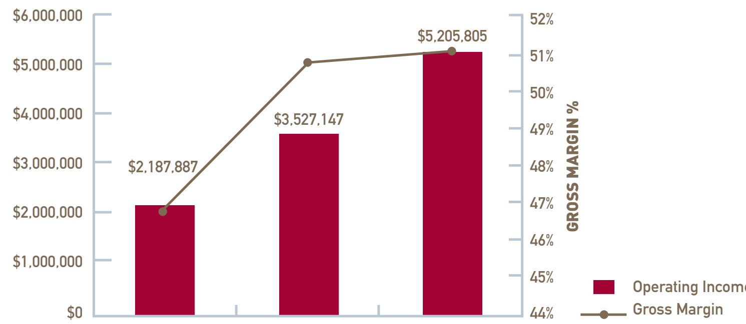

Figure 13. *Scenario B: Gross Margin and Payback*

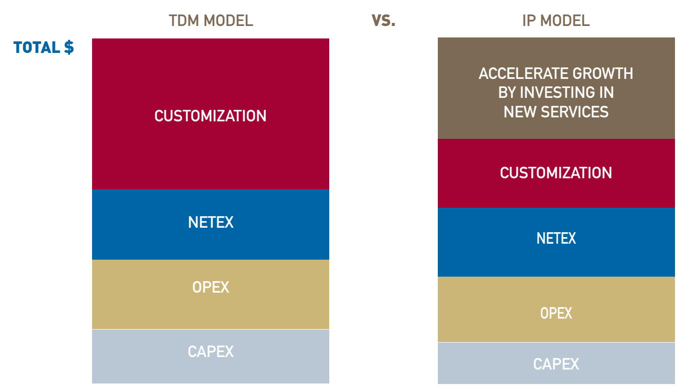

Figure 14. *Service Flexibility in an IP Model*

- Future-proof approach
  - IP Media Server multi-protocol support ensures a fit and longevity in any network environment—VoIP, wireline, wireless, and IMS

A pure IP approach to implementing voice services at the core of next generation, feature rich, multi-media services is the key value of NGN networks. This gives all service providers the basis for future proofing their networks and position for emerging core and access network technologies. The money saved versus TDM networks and service architectures is the icing on the cake that helps to fund the future of NGN services (Figure 14). This rebalanced investment approach gives service providers the ability to DO MORE WITH LESS!

Glossary

AIN: Advanced Intelligent Network

ARPU: Average Revenue Per User

ASR: Automatic Speech Recognition

CDR: Call Detail Record

CSCF: Call State Control Function

CSP: Conferencing Service Provider

DID: Direct Inward Dial

DS3: Digital Signal Level 3

DSL: Digital Subscriber Line

IMS: IP Multimedia Subsystems

IVR: Interactive Voice Response

MRF: Media Resource Function

MSML: Media Server Markup Language

NAT: Network Address Translation

NGN: Next Generation Network

OEM: Original Equipment Manufacturer

OSS: Operation Support System

PSTN: Public Switched Telephone Network

VoIP: Voice over IP

RTP: Real Time Protocol

SBC: Session Border Controller

SIP: Session Initiation Protocol

SNMP: Simple Network Management Protocol

TDM: Time Division Multiplexing

T1: T-carrier 1

TTS: Text-to-Speech

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