HIERARCHICAL INTER-CMS ARCHITECTURE USING STANDALONE SIP ROUTE PROXY

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Abstract

Cable multiple service operators (MSOs) are rapidly ramping up their VoIP subscriber base. As they deploy regional Call Management Servers (CMSs), they also deploy costly interconnects with the PSTN for routing calls either to another CMS or to the PSTN. A hierarchical inter-CMS architecture using a standalone SIP route proxy enables MSOs to avoid the unnecessary PSTN interconnection costs and gives them flexibility to deploy CMSs regionally in a way that best utilizes their network resources. This paper will discuss the role of a SIP route proxy in interconnecting CMSs, media gateway controllers (MGCs) and peer VoIP providers and illustrate the business case behind deploying one such architecture.

INTRODUCTION

Voice has become an integral part of triple play offered by the Cable multiple service operators (MSOs). According to the latest data, North American MSOs have announced over 3M VoIP subscribers and have plans to add many more. With this blistering growth comes the complexity of scaling the network. Also, Cable MSOs are looking for ways to keep voice calls on their IP network which not only improves voice quality but also significantly reduces call-termination costs. Additionally, offering multimedia services to enhance subscriber experience is also a priority. Hierarchical Inter-CMS architecture can not only be used to effectively scale the network but also to provide policy-driven PSTN interconnect strategy which can significantly reduce operating costs and help evolve MSOs network to PacketCable 2.0 / IMS architecture.

TODAY'S ARCHITECTURE

In order to fully understand the potential of the Inter-CMS architecture, let's have a look at the current architecture deployed today. Cable MSOs have deployed VoIP using the integrated softswitch architecture. This architecture is composed of CMS and MGC functionality, as defined by CableLabs™, into one single element. The key benefits of the current architecture are: scalability (compare to TDM switches), significant reduction in capital expenditure and the ability to introduce features at a reasonable cost. Most of the VoIP deployments by MSOs today are based on the PacketCable compliant multimedia terminal adapters (MTA) which can communicate with softswitches via network call signaling (NCS) protocol. The generic high-level diagram below shows the current VoIP architecture.

As shown in Figure1, the CMS and MGC are combined in a single element known as softswitch. When a call is received by a softswitch, it determines, from its internal database, the type of call. If the call type is long-distance, then it routes the call out to PSTN interconnection trunks. If the call is local, then it terminates the call to itself. In this scenario all off-net calls are routed out using PSTN interconnection. Even the calls between some subscribers served by the same MSOs have to traverse through PSTN.

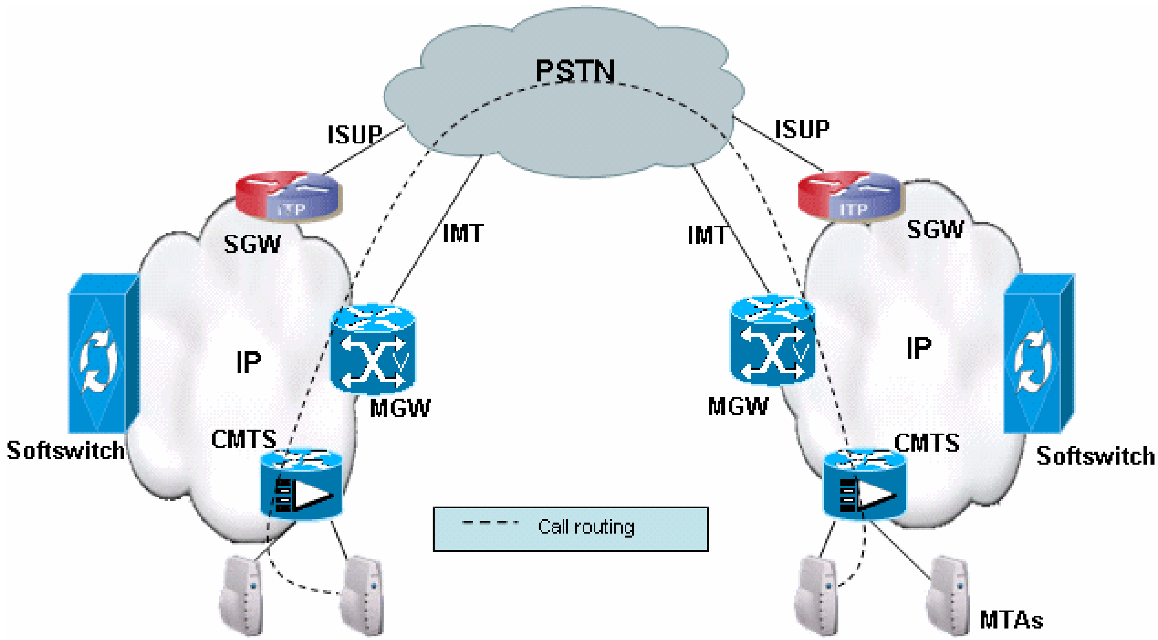

Figure 1. PacketCable™ architecture without SIP routing

This architecture has allowed MSOs to quickly and profitably deploy VoIP service to tens of thousands of subscribers in different regions. Multiple softswitches have been deployed in multiple points of presence (PoPs). To connect the VoIP subscribers in different PoPs and to the PSTN, most MSOs use inter-machine trunks (IMTs) for bearer connectivity and SS7 network for call signaling. This approach is reliable and widely deployed today but it has few shortcomings.

1) High cost associated with terminating calls to PSTN. The inter and intra lata calls must traverse the PSTN before reaching their destination. Also, MSOs have to pay a huge sum of money to buy the port capacity on media gateways and connecting signaling links and linksets to the SS7 network.

2) During interconnect, packet-to-TDM and TDM-to-packet conversion adds processing delays and hence affects the voice quality.

3) TDM and SS7 facilities are dedicated and can't be used for anything else but Voice traffic.

4) No sharing of ILEC interconnection between softswitches is allowed.

One way to potentially overcome the above mentioned shortcomings is to deploy SIP trunk signaling between softswitches. Most of the softswitches today support standardized version of SIP protocol – RFC 3261. By connecting softswitches with SIP trunks, it eliminates the need to interconnect to PSTN for "on-net" calls which not only improves the voice quality but also drives down the call terminating cost since fewer PSTN interconnections are required. The diagram below shows the SIP trunk signaling between softswitches.

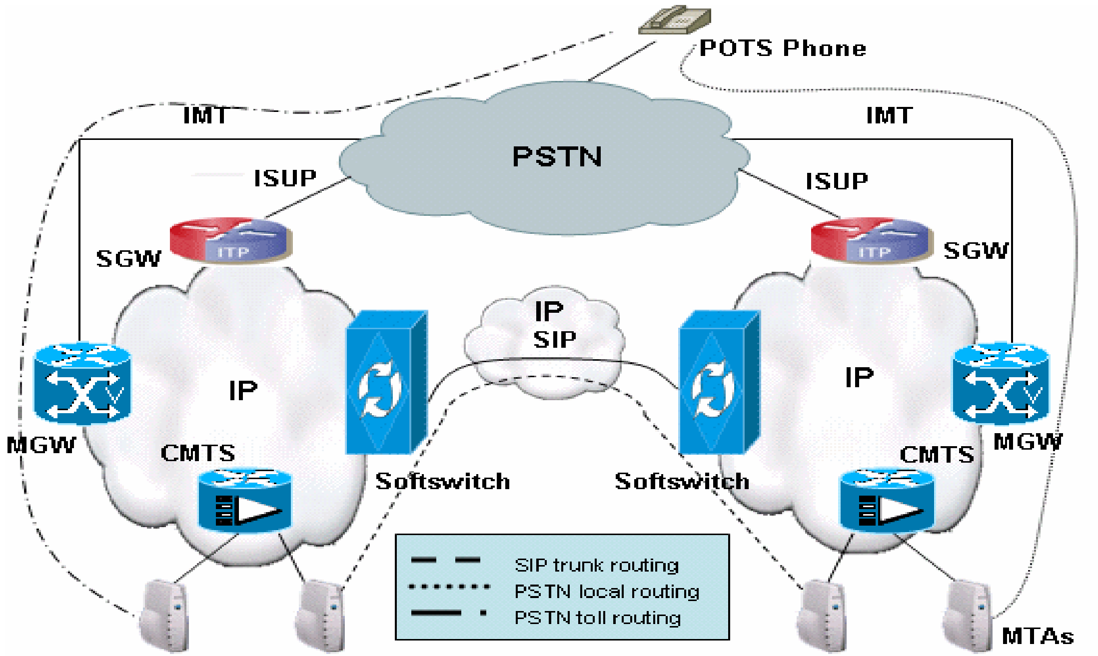

Figure 2. PacketCable™ architecture with SIP trunk signaling

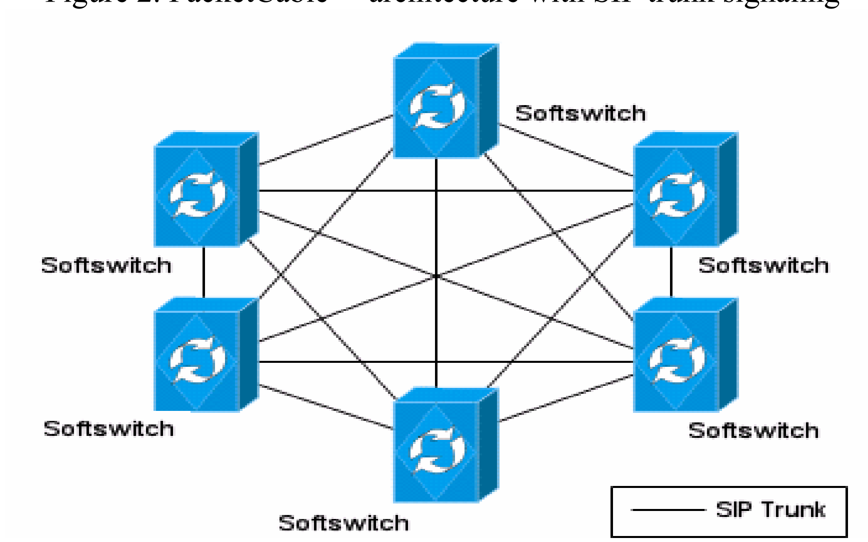

Figure 3. Full Mesh routing using SIP trunk between softswitches

As shown in the Figure 2, when a call is received by a softswitch, it determines, from its internal database the type of the call. If the call type is long-distance, then it routes the call out to PSTN interconnection trunks only if the call is terminating to a foreign subscriber (PSTN toll routing in Figure 2). But, if the call is terminating to MSO-owned subscriber then call is routed out to SIP trunk connected to terminating softswitch (SIP trunk routing in Figure 2). In this scenario not all "on-net" calls have to use PSTN interconnection for routing which results in operational cost savings.

Many MSOs have deployed SIP trunk signaling between softswitches. As this solution reduces the need for PSTN interconnections and improves voice quality, it raises other operational and technical dilemmas. MSOs have deployed multiple softswitches in a PoP and multiple PoPs across different geographic locations to support the growing demand of VoIP. This approach requires a fully meshed network – connecting every softswitch not only to every other softswitch in a PoP and to softswitches in other PoPs as well. Figure 3 below shows a fully meshed network.

As it is apparent from Figure 3, fully meshed networks are not scalable due to the complexity involved in configuring and maintaining large networks. For every n CMSs in an MSO's network, they'll need n times (n-1) – i.e. n-squared – number of trunks. Provisioning these trunks as the network grows to 10 CMSs and beyond can result in an unduly burdensome number of man-months or even man-years spent provisioning trunks on each new CMS added. This is to say nothing of continually managing and modify translation tables etc. as numbers or number ranges served by each CMS are continually modified.

Additionally, and more importantly, some of the formerly co-operative inter-exchange carriers have been acquired by the cable operators' competitors. Cable operators are therefore searching for ways to define new network architectures that provide operationally viable and cost-effective alternatives for both CMS deployment and PSTN interconnect to deliver high quality VoIP services to their subscribers. Introducing tandem-like hierarchical means of routing calls to and from NCS endpoints using stand-alone SIP route proxy could very well solve all the operational, technical and economical issues that MSOs face today.

INTER-CMS ARCHITECTURE USING STANDALONE SIP ROUTE PROXY

Hierarchical inter-CMS call routing using standalone SIP route proxy allows MSOs to more intelligently and economically route calls on their network. As we will see in later sections, inter-CMS architecture, despite its incremental capital cost, offers steep OpEx savings. But, let's first take a look at how a SIP route proxy can be utilized to offer highly-available, scalable, centralized and intelligent solution to enhance MSOs PSTN strategy.

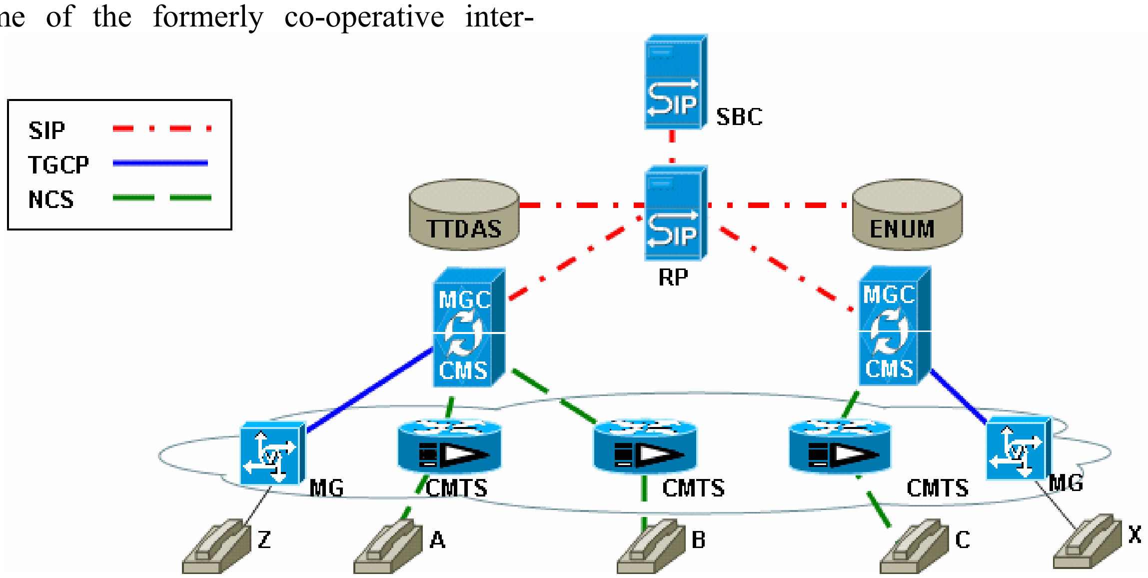

Figure 4. Inter-CMS architecture using SIP route proxy

The figure below shows a generic Inter-CMS architecture. It consists of a standalone SIP route proxy, external database, for e.g. TTDAS, and an ENUM server along with all the required components of PacketCable architecture.

With the help of SIP route proxy, call routing can be accomplished in variety of ways as described later in this section. This gives traffic engineers greater flexibility as to how different types of calls can be routed in the network. By gaining control over call-routing, MSOs can achieve two exceedingly important goals of keeping calls on-net and avoid costly PSTN interconnections. For example, in the diagram above, call routing can be accomplished in following ways:

On-net routing:

Local: User A calls B. Since there is IP connectivity between A and B and both are on the same CMS, the CMS route from A to B is NCS to NCS.

Long-distance: User A calls C. There is IP connectivity between A and C but they are on different CMSs. The CMS route table for A routes the call to SIP Route Proxy (RP). The SIP RP routes the call to the CMS that C is attached to.

Off-net Couting:

Local Origination: User A calls POTS Z. CMS determines that call is local and hands the call off to MGC serving the local market. If all the PSTN trunks are currently in use then a route advance is performed and the call is forwarded to closest MGC using the SIP RP.

Long-distance Origination: User C calls POTS Z. CMS determines that call is long-distance and routes the call to SIP RP. SIP RP determines the nearest MGC and hands the call off to that MGC for call termination.

Local Termination: PSTN subscriber Z calls user A. MGC determines that call is terminating to directly-connected CMS and thus hands the call off to that CMS.

Long-distance Termination: PSTN subscriber Z calls user C. MGC determines that call is not local so, routes the call to SIP RP. SIP RP determines which CMS to route the call and hands the call off to that CMS.

As it is evident from the examples above, the SIP route proxy facilitates the routing between CMSs and supports various routing options for calls destined to/from PSTN. All different call types such as local, toll, and long-distance can be routed by SIP route proxy such that calls can be retained on-net for better voice quality and reduced operating expense.

IMPLEMENTING INTER-CMS ARCHITECTURE

The first phase in implementing Inter-CMS architecture is logically and physically separating softswitch into CMS and MGC as shown in Figure 5. CMS is used for line side features such as controlling MTAs via NCS protocol, providing subscriber features and accessing subscriber database. MGC is used for trunk-side features such as outbound route selection, PSTN interconnection and providing PSTN features such as LNP.

Once the CMS and MGC are separated, the next step is to introduce a SIP route proxy in the middle. Figure 6 shows a standalone SIP route proxy that can route calls directly between CMSs or between CMSs and multiple MGCs in the network.

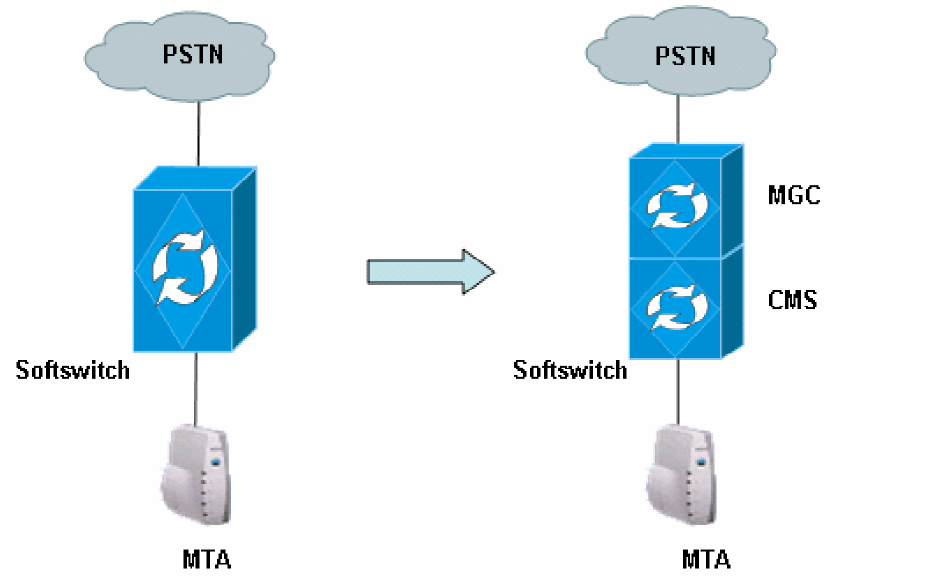

Figure 5. Step 1: Separate Softswitch in CMS and MGC

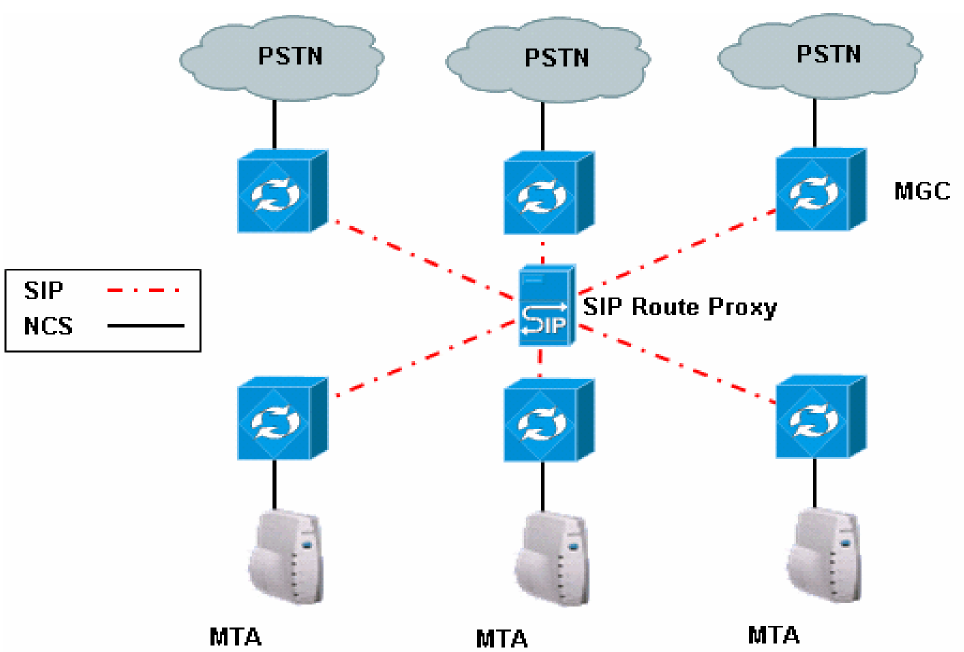

Figure 6. Step 2: Deploy SIP Route Proxy

The SIP route proxy performs multiple roles in the network –

1) It allows PSTN interconnections to be shared by multiple CMSs such that CMS on the left, in the figure above, can route calls to PSTN by interconnecting with MGC on the right or on the left depending upon the least cost routing logic set in the SIP route proxy.

2) Since CMS and MGC are separated in two different physical

components, they can be scaled independent of each other. You can have difference number of MGCs, depending on number of PSTN interconnections required, then CMSs which depends on growth of VoIP subscribers.

3) The SIP route proxy can be used to provide policy-driven IP interconnection strategy to Peer MSOs' or partner IXCs' networks. The IP interconnections can be managed by connecting SIP route proxy with the session border controllers (SBC).

Routing Logic and Capabilities

The SIP route proxy is typically a centralized function in the network. It can be shared by multiple PoPs and is usually deployed as a cluster of servers for high availability. Its primary role is to handle the routing of calls such that calls between MSO-owned subscribers can be kept on-net and for all other calls needing PSTN interconnection, can be routed based on least cost option available. Routing requests can be received from any element in the network which either do not provide this resolution capability, or in cases where operating policy dictates this logic be centrally located and administered.

SIP route proxy can be configured to do ENUM lookup on every request it receives. When a request is received by SIP route proxy, it queries the ENUM database to determine if the call is to a MSO-owned subscriber. If that is the case then it routes the call to the CMS responsible for the subscriber. Otherwise it can route the call either to a session border controller (SBC) or to a local media gateway controller (MGC) depending on the type of desired PSTN interconnection. For terminating calls in the network, SIP route proxy can query the external database such as TTDAS. When a request is received, SIP route proxy can query the database on NPA-NXX of the called number to determine and to route the call to CMS attached to that subscriber.

SIP route proxy can also be used to route advance in case of congestion or downstream failure. When a failure response is received, SIP route proxy can attempt to route the call over to the alternate PSTN interconnect point such that the call can be successfully completed.

Enhanced Routing

As the subscriber base of VoIP service grows, MSOs are not only witnessing increase in "on-net" traffic but the inter-MSO traffic is growing as well. Hence, MSOs are seeking for ways to directly route calls between their networks to reduce OpEx. SIP route proxy combined with ENUM server can be effectively used to route calls between providers' networks. All "off-net" calls can be routed to SIP route proxy which can determine, after an ENUM query, where the call needs to be routed – either to a TDM interconnect point or to a directly connected peer-MSO network.

Besides reducing the OpEx, SIP route proxy can be used to effectively manage and in some cases deploy new revenue generating services. For e.g. Virtual Number service which provides its subscriber the freedom to choose the area code of their choice in return for a fixed subscription fee. SIP route proxy, upon receiving a request to terminate a call to the Virtual Number, performs ENUM query and determines the actual number and routes the call to the CMS the subscriber is attached to.

Evolution to PacketCable 2.0 Architecture

Cable operators today are not only thinking about reducing OpEx but also are looking to evolve towards next generation architecture to offer their subscribers revenue-generating SIP based multimedia services.

For today's MSOs, it is not the question of "if" but "when" and "how" they will move from PacketCable 1.X architecture to tomorrow's PacketCable 2.0 / IMS architecture. It is important for MSOs to achieve this evolution through addition of individual capabilities that solve a business need, such as inter-CMS call routing, and also add components of the target architecture. Gracefully introducing individual pieces of next-generation architecture is a key to successful network evolution. Inter-CMS architecture, if you will, is the first step in that evolution process. By adding SIP route proxy, which can function as interrogating call session control function (I-CSCF) or border gateway control function (BGCF), and separating CMS and MGC, where MGC can perform media gateway control function (MGCF), completes the first step in evolving towards the next generation architecture.

Adding an I-CSCF function to a cable company's VoIP network becomes increasingly important as MSOs look to evolve and deploy parallel SIP-based voice (e.g. business line services to SMB customers) and SIP-based multimedia (e.g. video phone services to consumer customers) service networks leveraging their current deployed PacketCable infrastructure for PSTN interconnect, 911 calling, etc. These services need a common point of attachment – or the full meshing problem described above becomes infinitely worse – and all services need a way of routing calls or "sessions" that are addressed to a SIP URI (e.g. bob@anycableco.net) as well as to e.164 telephone numbers. Cisco believes these should be capabilities of any Route Proxy added to MSO's networks from Day One.

BUSINESS CASE

To demonstrate the benefits of pursuing inter-CMS routing via a standalone SIP route proxy, a business case has been developed. This business case considers alternative deployment scenarios for routing traffic between CMSs, thereby keeping more calls on-net. This business case captures the investment (both CapEx and OpEx) required to implement each scenario and details the resulting economic savings for each.

In total, three scenarios were evaluated

- Case A – No Inter-CMS Routing

- Case B – Inter-CMS Routing via Meshed SIP Trunk Network

- Case C – Inter-CMS Routing via Standalone SIP Route Proxy

The economic analysis for the scenarios is comprised of the following factors:

- CapEx: Investment in TDM gateways, session border controllers, SIP route proxy servers, router ports required for VoIP backhaul between PoPs

- OpEx: IP circuit expenses for backhauled traffic, PSTN interconnection charges, and deltas in operational staffing expenses. (Note: capturing full operational costs for running a VoIP network is not part of this exercise, only the

	Emb Base	Year 1	Year 2	Year 3	Year 4	Year 5
Subscribers						
Number of Subscribers	500,000	600,000	720,000	864,000	1,036,800	1,244,160
Network Layout						
Number of PoPs	5	5	5	5	5	5
Total Number of CMS	9	10	12	15	18	21
Call Volume						
Total Minutes of Use (M)	5,475	6,570	7,884	9,461	11,353	13,624
Calls per Second (BH)	278	333	400	480	576	691
Simultaneous Calls (BH)	50,000	60,000	72,000	86,400	103,680	124,416

Table 1. Network Topology

relative differences between scenarios)

- Revenue: Incremental revenue associated with enhanced services.

Network Topology

The sample network modeled has an existing base of 500,000 voice subscribers. Over the next 5 years, this network is expected to grow with a CAGR of 20%. An existing network was selected versus a greenfield network in order to reflect the current state of the industry and to recognize that inter-CMS routing often becomes a problem after initial launch. Further statistics about the network topology and traffic are provided in Table 1.

Traffic Distribution

A critical factor in determining PSTN interconnection expense is characterizing traffic as “on-net”, where both originating and terminating points of the call are on the cable network vs. “off-net”, where the call either originates or terminates on the PSTN. Also, whether the call is local or long distance has a significant effect on interconnection fees.

Table 2 illustrates the percentage of calls by type over the life of the model. It is apparent that as more subscribers use the cable voice network, the probability of ‘on-net’ calls increases. This model assumes symmetric patterns of originating and terminating calls.

PSTN Termination Fees

A cable MSO typically has agreements in place with a competitive or inter-exchange carrier that handles PSTN interconnection. The MSO pays a per-minute termination to this interconnect partner for off-net calls. PSTN termination fees assumed in this model are:

Long Distance (including Toll) Termination Fee: $0.02 per call minute

Local Termination Fee: $0.003 per call minute.

A ten percent reduction in termination fees was assumed in the case where the cable operator interconnects to the carrier network directly via IP trunks.

Scenario Descriptions and Call Handling

Case A (No Inter-CMS Routing) is illustrated in Figure 7. In this case each softswitch is isolated from the others. The only calls that are kept “on-net” are intra-CMS calls, calls that originate and terminate on the same CMS. All other calls are routed to the interconnect partner via MGC and MG.

	Year 1	Year 2	Year 3	Year 4	Year 5
On-Net Calls					
Intra-CMS Calls	2%	3%	3%	4%	4%
Intra-PoP Calls	2%	4%	5%	7%	8%
Inter-PoP Calls	3%	4%	4%	5%	6%
MSO On-Net Calls					
MSO Peer Calls	0%	0%	2%	4%	6%
Off-Net Calls					
Local	46%	44%	42%	40%	38%
Long-Distance + Toll	47%	47%	44%	41%	38%

Table 2. Traffic Distribution

Case B (Inter-CMS Routing via Meshed SIP Trunk Network) is illustrated in Figure 8. In this case, all of the CMSs are interconnected to one another via a meshed network of SIP trunks. This offers the benefit of keeping significantly more calls on-net: intra-CMS calls, inter-CMS calls within the same PoP, inter-CMS between PoPs. All remaining calls are handed off to the interconnect partner either via TDM trunk (MG) or IP trunk (SBC).

Case C (Inter-CMS Routing via Standalone SIP Route Proxy) is illustrated in Figure 9. In this case, each of the CMS is interconnected to a set of SIP route proxies that provide the routing logic for all calls. This not only enables the carrier to keep the following calls on-net: intra-CMS, inter-CMS calls within the same PoP, and inter-CMS calls between PoPs, but it also retains MSO peer traffic on-net. This latter category is enabled by providing a single network point for all ENUM queries. Interconnection is handled via a combination of SBCs (MSO peers and PSTN) and MGs (PSTN).

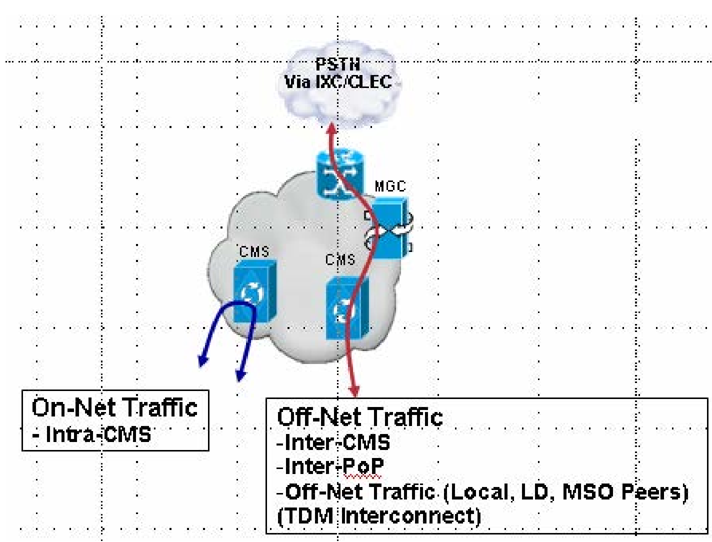

Figure 7. Case A - No Inter-CMS Routing

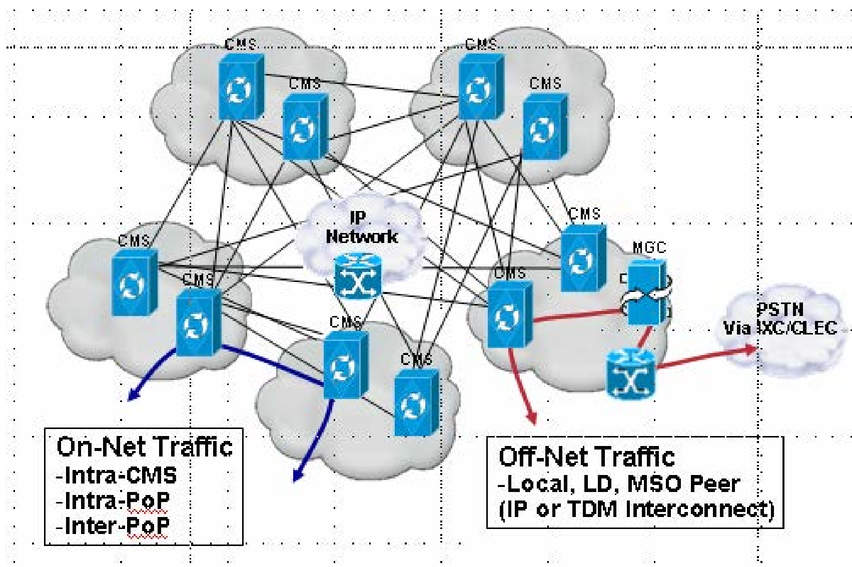

Figure 8. Case B – Inter-CMS Routing via a Meshed SIP Trunk Network

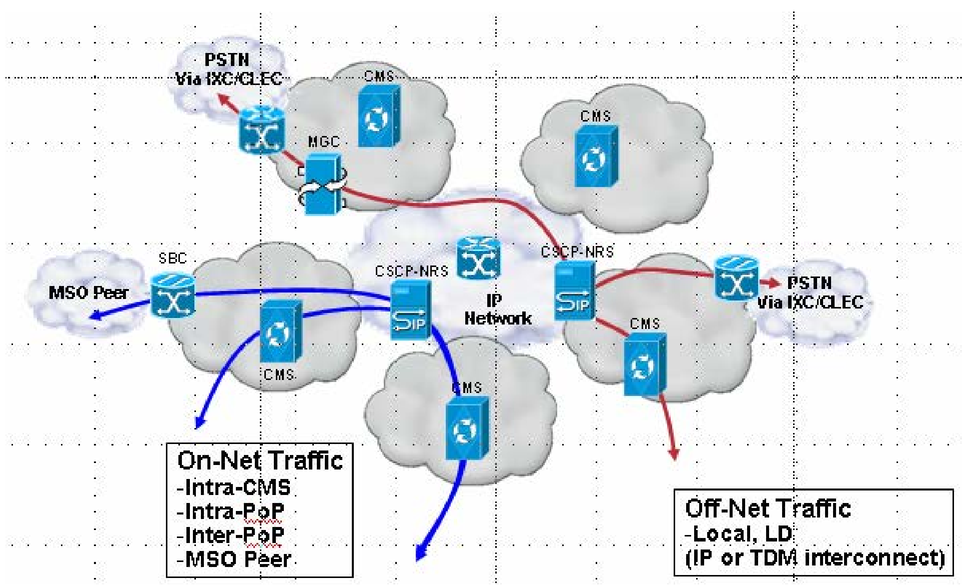

Figure 9. Case C – Inter-CMS Routing via SIP Route Proxy

Financial Investment

The amount and nature of investment required for each scenario differs.

Case A requires relatively little investment. Since most of the traffic is handed off to the PSTN carrier at the originating PoP; the only CapEx investment required is the purchase of incremental TDM gateway ports as traffic increases. A modest staffing expense is incurred to establish and maintain the routing tables in each softswitch.

Case B requires a modest investment. While more calls are staying on-net, driving a reduction in gateway capital expenses, there is incremental expense associated with backhauling IP traffic (routers and IP circuits). Also, there is a significant expense associated with setting up and maintaining a meshed network between CMSs (N-squared problem), and maintaining unique routing tables in each CMS.

Case C also requires a modest investment. Operationally, this scenario is easier to implement than Case B because fewer SIP trunks are required and there are fewer routing databases to provision and maintain. Consequently, there is a noticeable reduction in operational expense. But there is an incremental investment in SIP route proxies in order to support this routing function.

A summary comparing the investment required (excluding PSTN interconnection fees) for each of the three scenarios is illustrated in Figure 10.

Financial Return

The return on investment for each scenario can be articulated by calculating how much cost avoidance was achieved thru the reduction in PSTN interconnection fees. Additional value is captured by accounting for incremental revenue achieved by selling virtual number services.

In Case A, the cable operator pays a total of $495M in PSTN interconnection expenses over 5 years. This figure understandably increases in each of the years, as the total number of subscribers and traffic increases. This fee, $495M, represents the largest expense item for the cable operators rolling out cable voice service.

Case B reduces that expense by a considerable amount. By keeping more calls on-net and beginning to employ SBCs for IP interconnect, the total PSTN interconnection expense for the 5 years is $425M, a reduction of $69M or 14%.

Case C, however, offers the optimal return. Keeping even more of the traffic on-net allows the cable operator to reduce the

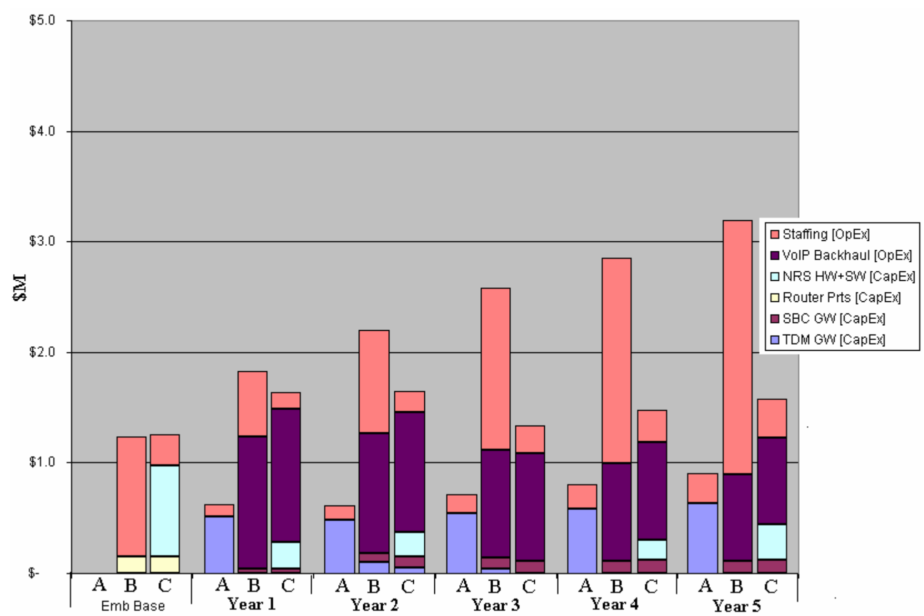

Figure 10. Comparative Financial Investment
(Excluding PSTN Interconnection Expense)

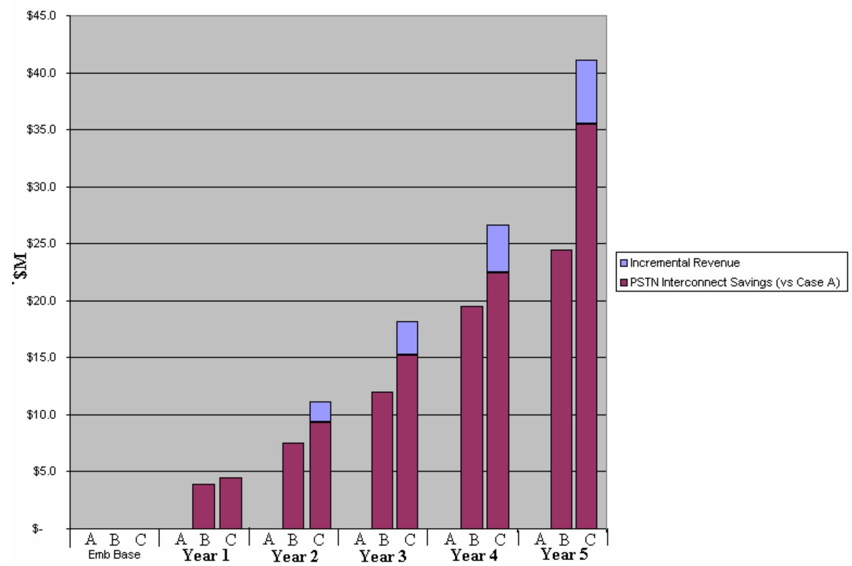

Figure 11. Comparative Financial Return

	5 Year Investment	5 Year Return	NPV	IRR
Case A	$ (3.65)	$ -	$ (1.87)	NA
Case B	$ (13.89)	$ 69.12	$ 35.08	392%
Case C	$ (8.93)	$ 121.34	$ 70.62	475%

Table 3. Financial Metric Summary

PSTN interconnection expenses for the 5 years to $388M, a $37M improvement over Case B (9%), and a $107M improvement over case A (22%). The case is further enhanced by incremental revenue associated with virtual number service adding an incremental $14M to the top line over 5 years.

The summary comparing the financial return for the three scenarios is shown in Figure 11.

Financial Metrics

Table 3 summarizes the investment and return for each of the scenarios. Case C, which uses the standalone SIP route proxy for inter-CMS routing offers the best return on investment.

SUMMARY

The hierarchical inter-CMS architecture using a standalone SIP route proxy helps MSOs scale their cable voice networks, while effectively managing their operating costs.

The benefits achieved are:

Operational Benefits:

- Maintaining more traffic 'on-net' improving overall service quality

- Avoiding operational complexity associated with growing cable voice networks

- Scaling subscriber counts independently of PSTN interconnection

Financial Benefits:

- Dramatically reducing PSTN interconnection expenses
- Optimizing CapEx investment
- Reducing ongoing staffing expense
- Facilitating new revenue streams

Strategic Benefits:

- Reducing dependency on PSTN interconnect partner who is often a competitor
- Gracefully evolve towards NGN based on PacketCable 2.0 / IMS architecture.

The inter-CMS architecture is based on highly-available and intelligent technology which can immensely help cable operators increase their profitability, efficiency and level of control over call routing both from, and within, their network.