SIP SESSION ROUTING

A METASWITCH WHITE PAPER

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EXECUTIVE SUMMARY

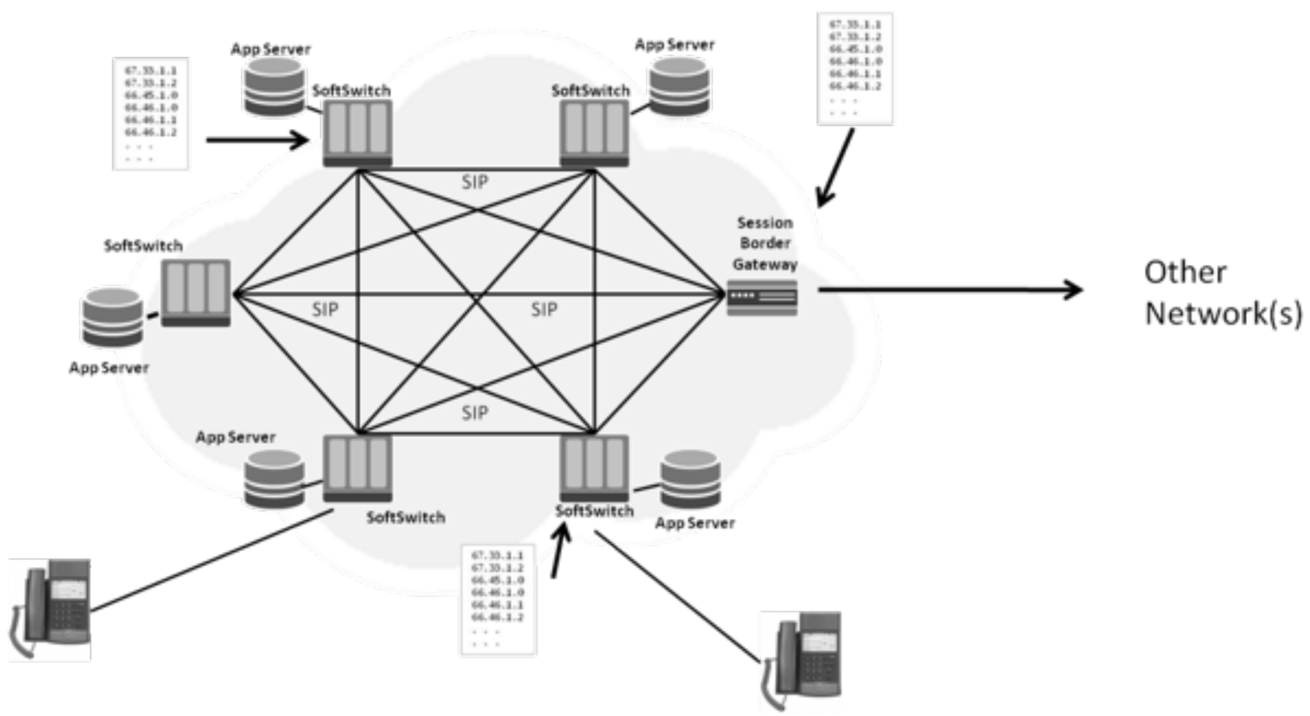
Service providers are now accelerating the move to NGN as an interim step towards a full IMS architecture. As the number of SIP-based switches and other SIP elements grow, the maintenance of the SIP network becomes a limiting factor, as SIP lacks dynamic routing capabilities and therefore routes must be manually provisioned on each NGN network element. SIP Session Routing addresses this shortcoming and provides additional benefits to streamline NGN.

INTRODUCTION

The global deployment of Next Generation Network (NGN) is accelerating as service providers take advantage of VoIP / SIP inherent benefits in lower costs, improved capabilities and greatly expanded possibilities for new services and features. This growth is seen in both subscriber access as part of the move of the PSTN to NGN, as well as in the enterprise arena as PBXs are swapped for IP-based PBXs.

As the number of SIP-based switches, proxy servers, feature servers, etc. grows, service providers are finding that there are some very significant limitations on how SIP message routing is implemented. Namely, unlike IP data networks that rely on dedicated routing protocols, SIP elements on NGN must be individually provisioned to contain routing tables describing how to reach peers and beyond. Effectively this means that adds / deletes on the NGN affect each and every SIP element, and each one of these elements must be re-provisioned when there is a topology change, which places a great deal of stress on support and deployment teams and limits when maintenance can take place. Of course this also translates into higher operational costs.

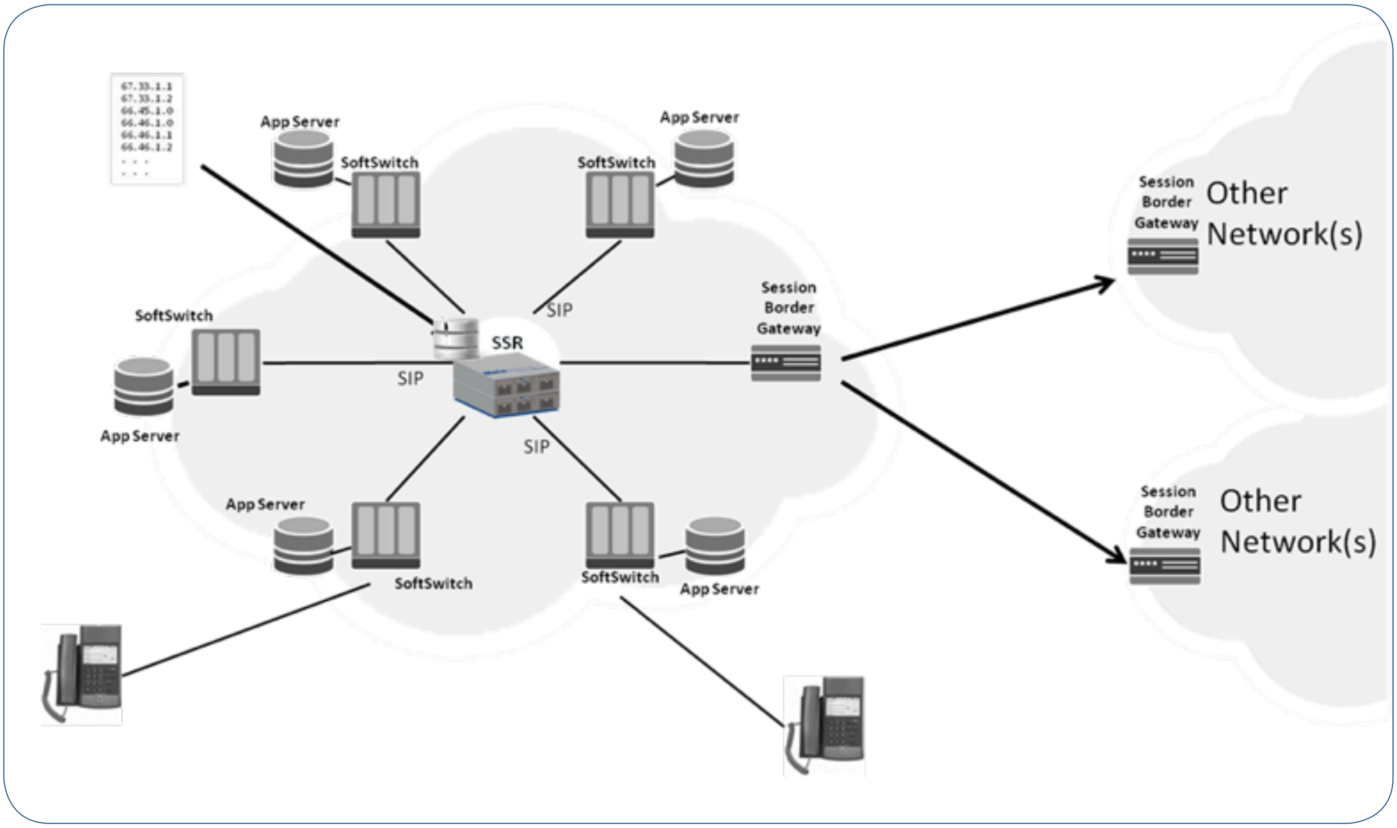
SS7 networks inherently have similar limitations but those are addressed by creating hierarchical architectures with Class 5, Class 4, and STPs. However, in NGN, no such hierarchical architectures exist and therefore NGN networks are effectively mesh networks as shown in the figure below.



CREATING A HIERARCHY

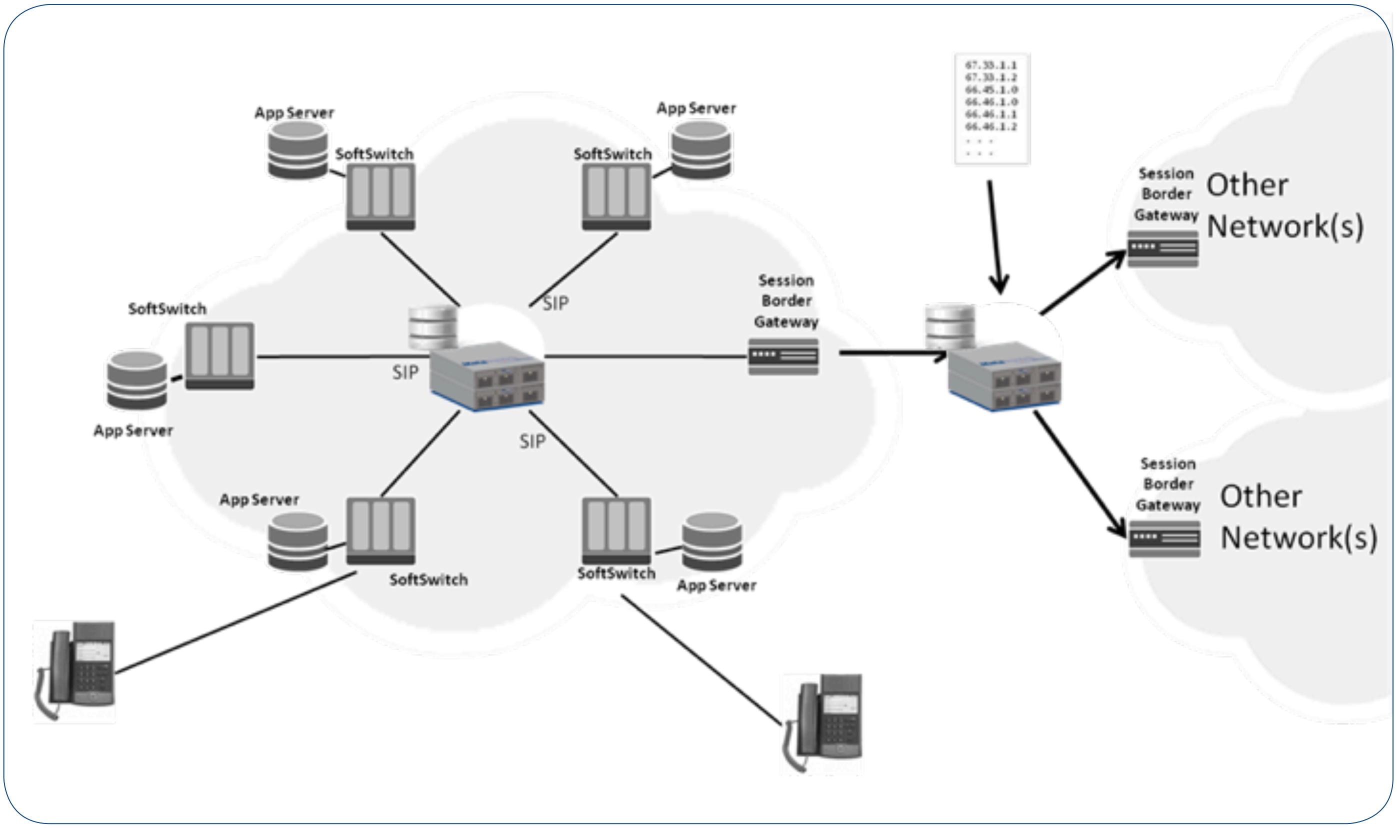
The solution to this problem is to bring the concept of hierarchical networks to NGN by introducing a SIP element known as the SIP Session Router (SSR). The SSR is designed to host all the routing tables and make the routing decisions that were previously delegated and distributed throughout the NGN mesh network. The SSR may be viewed as a specialized SIP proxy with the capacity to process the call load of the NGN switches, and potentially, with additional capabilities designed to optimize the NGN.

When the SSR is introduced into the NGN, the existing soft switches are re-provisioned with a new upstream peer, the SSR. The following diagram shows the concept:



The network then becomes much more manageable and adds / deletes to the topology have minimal impact to the rest of the SIP elements. In some cases, the soft switches may gain call processing capacity in that less CPU cycles are spent making routing and forwarding decisions.

In deployments where session border gateways provide access to the service provider network, the provider may actually have many SBCs that connect to it from large enterprises and smaller rural providers. SBCs face the same challenge and may also benefit from the introduction of the SSR as a centralized routing point. This concept is shown below, where an SSR has been added by the service provider between different logical networks.



OTHER APPLICATIONS OF THE SIP SESSION ROUTER

The SIP Session Router is able to provide a number of solutions on the NGN network, all based on the fact that as a central SIP routing platform, a great deal of SIP traffic is inherently visible to it. While the initial goal of the SSR is to centralize SIP routing functions, the following applications are additionally (and concurrently supported):

- SIP Trunking
- Application Server Load Balancing and Fault Detection
- Media Server Load Balancing and Fault Detection (Media Resource Brokering)
- Intelligent Networks (IN) database look up / IN “dips”

SIP TRUNKING

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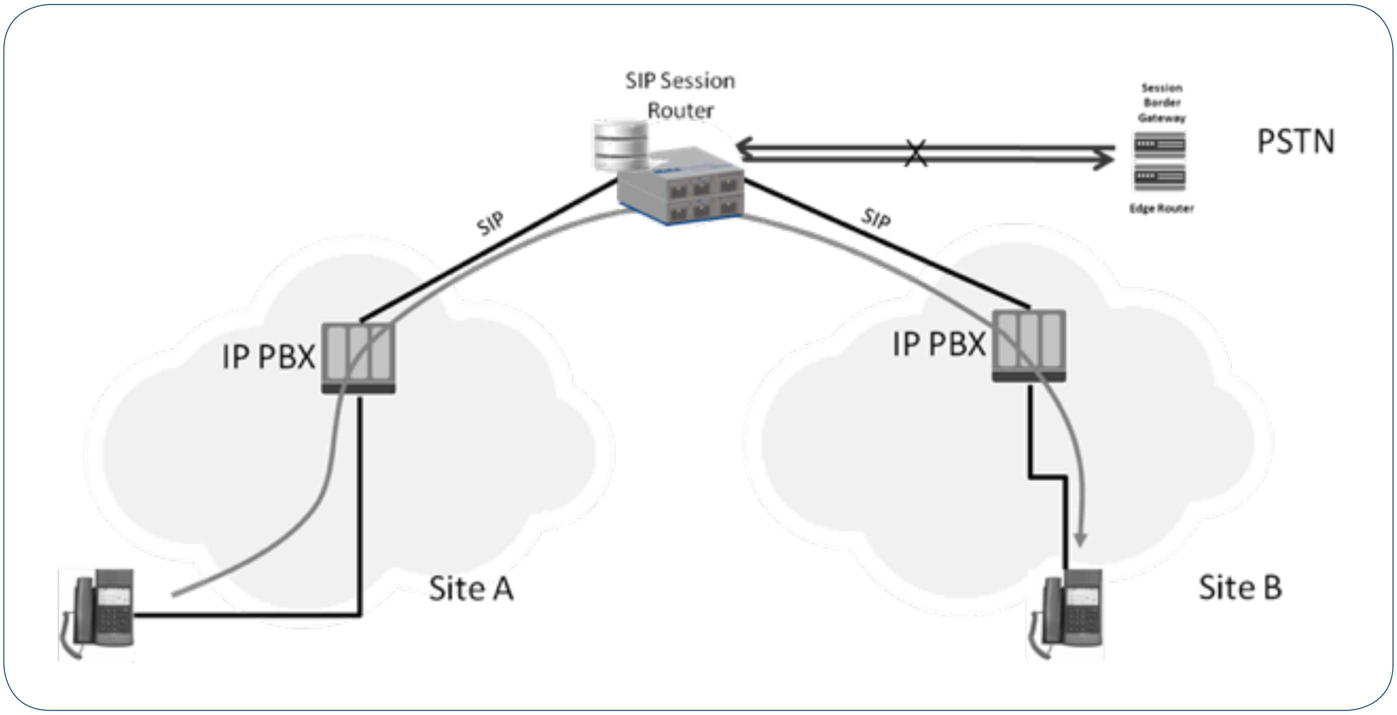
1 SIP Normalization

Looking at the SIP trunking example, large organizations typically insist on utilizing their own approved gear, perhaps an IP PBX or a session border gateway, which is likely to introduce SIP message interoperability issues. The SSR can provide the SIP interworking to ensure large heterogeneous networks can communicate.

2 OnNet / OffNet Routing

Continuing with the enterprise trunking example, let’s assume the customer has several large campuses across the country and desires to have each IP PBX communicate with each other. The service provider may currently simply send each IP PBX back towards the PSTN via session border gateways, meaning that calls between two IP PBXs effectively trombone via the PSTN consuming valuable circuits and TDM switching equipment.

By utilizing the SSR, the service provider may configure SIP routes that keep the traffic “OnNet” of the SIP enterprise customer as illustrated below. This solution reduces latency, lowers operational and equipment costs, and delivers better service quality.



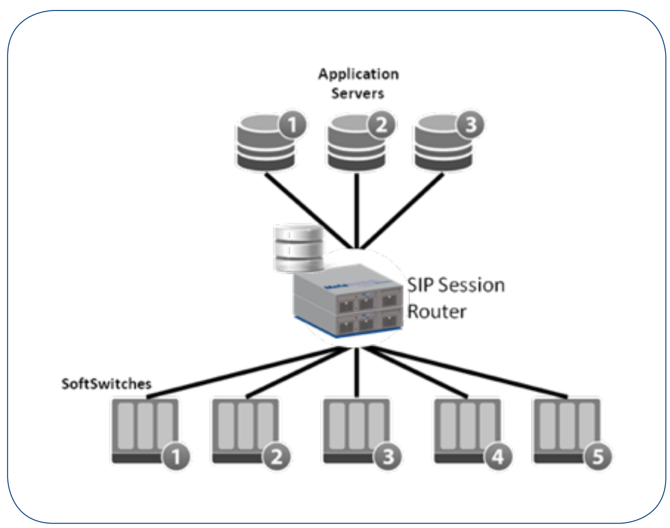
APPLICATION SERVER LOAD BALANCING

The typical deployment model of early NGN has been to couple a feature server with media gateway control in what is effectively the soft switch entity. Typically, there is a one-to-one relationship between the switching and feature server components. However, evolution is taking NGN towards the IMS model where there are a number of application servers in the network that provide the subscriber experience. In addition, larger networks inherently require larger call capacity and resiliency, not well served by this one-to-one architecture.

What is needed is the ability to expand the deployment to include a larger number of application servers, while at the same time ensuring that those resources are load balanced and collectively present a resilient resource. The SSR provides the ability to proxy the application/feature servers towards the switching infrastructure by appearing as a single application server entity towards any number of switches. The switches in turn are simply provisioned to forward SIP towards a single entity that has the shared capacity of all application servers.

The SSR provides the capability for identifying and correctly forwarding traffic towards the correct application server and provides a mechanism to load balance and diminish the impact of application server failures when multiple servers are pooled.

Depending on the service provider needs, the growth of application servers and switches can now happen asynchronously, adding capacity only where is needed. Furthermore, overall network uptime and service availability is greatly enhanced as server failures can be non-service affecting.



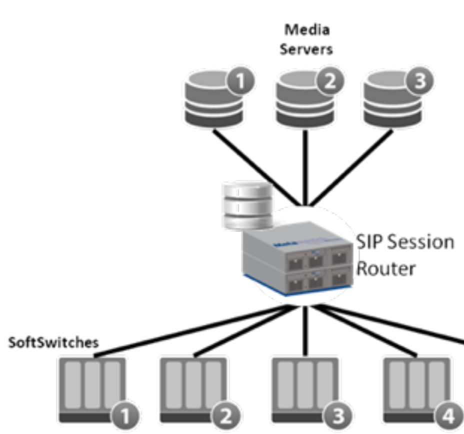
MEDIA SERVER BROKERING

A common goal across many network architecture areas has been to remove the silo effect where resources are tied to specific services. This goal extends to the deployment of media servers used by enhanced applications on the network, such as voice mail, televoting, etc. The most common deployment model is for application servers to have dedicated media servers to deliver bearer (voice) services, such as recording messages, playing prompts, etc. As multiple applications are deployed, the service provider often finds that there is excess media server capacity. Much of this capacity is there to deal with peak traffic, which depending on the application, may not coincide with any other peak load of any other application. Ideally, a single pool of media resources would exist in the network for everyone to utilize.

IMS addresses some of this issue by specifying the media resource function or MRF. In order to provide additional resiliency and fault tolerance, 3GPP added a new functional entity, the Media Resource Broker, or MRB, as an application layer capability.

The SSR optionally implements this MRF/MRB functionality to allow the pooling of media servers into a common resource to be used by new and existing application servers. From a SIP application server point of view, the SSR appears to be a media server accepting SIP INVITEs and negotiating media streams as needed. The SSR is then responsible for maintaining a view of available media servers and load balancing traffic across all available resources. In addition, media server failures are hidden from view from the rest of the network, enhancing network uptime and minimizing any service downtime.

Logically, the MRB functionality within the SSR looks identical to the server load balancing, as shown below.

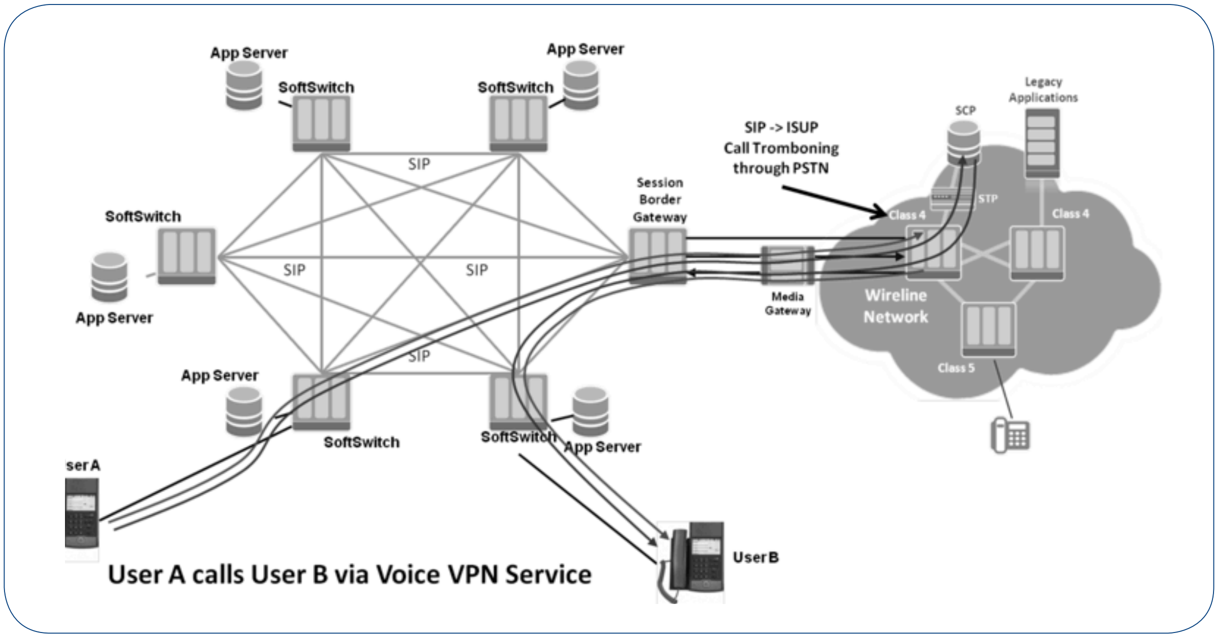


INTELLIGENT NETWORKS INTERWORKING OPTIMIZATION

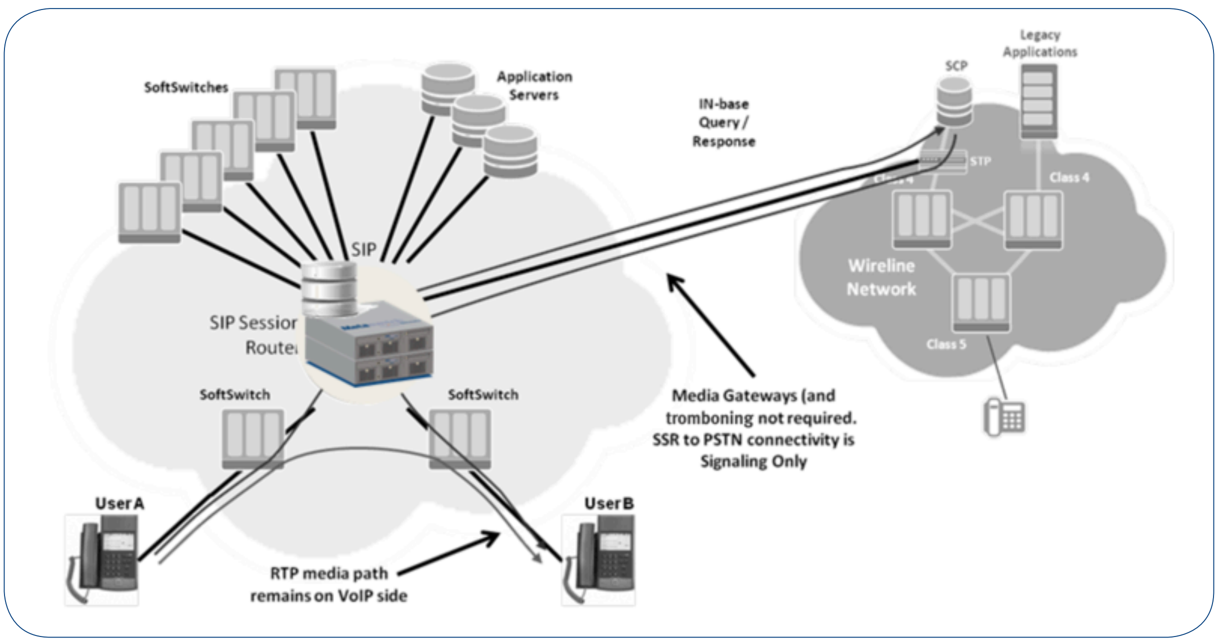
NGN networks are growing, with current subscribers moving from the circuit-switched TDM network. Of course, all current applications and services need to be provided to those migrated customers, which often mean reutilizing existing IN SCPs and servers. Services such as CNAME, LNP, 800 / freephone, etc. are usually hosted by IN-based servers that have enough capacity to support PSTN and NGN subscribers and are therefore not targets for transition to SIP.

A solution often implemented is to simply forward calls that require such services back towards the PSTN to be serviced, as described in the diagram below. In this example, User A tries to reach User B, who is subscribed to a voice VPN service. As the local soft switch does not know how to reach that user, the call is forwarded out towards the PSTN and terminated at a Class 4 switch which triggers an SCP for the number (VPN) lookup via INAP or TCAP.

The entire call is forwarded, which means signaling and bearer is delivered to PSTN and anchored on the Class 4. The Class 4 in turn sends the call back out towards the appropriate soft switch on the NGN, thereby consuming two voice circuits for the duration of the call. By the time the call ends, it would have consumed 2x resources on the session border gateway, media gateway and Class 4



A better way of delivering such legacy services on the NGN is to intelligently perform only the INAP or TCAP transaction, without the bearer and tromboning. This requires that some element on the NGN identifies this need and makes the IN lookup on the behalf of the soft switch. In the diagram below, the SSR has been deployed to provide SIP routing between soft switches, and hence is a natural transversal point for SIP traffic outside the local soft switch. By providing IN support on the SSR, it is able to perform this address lookup natively during call setup. As User A makes the outbound call to User B, the SSR performs the IN lookup after the initial SIP INVITE from User A, but obviously before an SDP has been negotiated and without having to send the call towards the PSTN.



The SSR may be configured to support either SS7 links or SIGTRAN associations natively and as such only needs to send signaling towards the PSTN, which means that valuable circuits are not tied and tromboned across the PSTN network

SUMMARY

NGN lacks the SIP routing and hierarchal design required to scale at the pace of SS7 networks. SIP Session Routing allows the service provider to greatly expand their current NGN networks without incurring tremendous operation expense maintaining SIP routing tables on soft switches. Additional capacity may also be gained by liberating the soft switches from performing extensive routing table management. Once deployed, the SIP Session Router may optionally provide additional benefits in the form of application server and media server load balancing and fault management, and may also be used to lower capital and operational costs related to delivering legacy applications and services via call tromboning to the PSTN.

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