

Migration of Enterprise VoIP/SIP Solutions towards IMS

Ram Kumar¹, Frank Reichert¹, Andreas Häber¹,
Anders Aasgard², Lian Wu²

Abstract— Voice-over-IP (VoIP) solutions are now widely spread and accepted by end-users. For an enterprise customer it is increasingly difficult to choose one of the numerous VoIP solutions which are available today. What options exist for an enterprise VOIP/Session Initiation Protocol (SIP) solution to migrate from today’s solutions towards interoperability with future IP Multimedia Subsystem (IMS) solutions, and what would motivate such an upgrade? We visualize four such solutions and position them along a possible migration roadmap. A common VoIP/SIP/IMS research platform is necessary as a test-bed for establishing an environment for development of future “All IP” products and services.

Index Terms—VoIP, SIP, Telephony, IMS, interoperability

I. INTRODUCTION

A. Background

For fixed and mobile operators new solutions based on IMS are on the horizon, and investment strategies for VoIP solutions are difficult. Enterprises will need to select the right strategy for cost-efficient and flexible voice and application services as well as being prepared for future business. Operators need to address the enterprise market with new services while manufacturers have to invest in the right product portfolio. The VoIP implementation and testing dictates that the key element with in the test-bed should be a soft switch or SIP PBX. This element can be a combination of several SIP entities such as SIP registrar, proxy server, redirect server, forking server, back-to-back user agent (B2BUA) etc depending on the different session control requirements.

IMS will play a key role in the future All-IP infrastructure, but it is still in the development stage. It should be pointed out that it will take time for all 3G mobile networks to upgrade to 3GPP Release 5 network and for fixed network to migrate from Public Switched Telephone Network (PSTN) to IMS based Next Generation Networking (NGN). It will still take several years before the full IMS functionality is realized.

However, enterprises cannot wait for IMS. Some enterprises have already started using SIP VoIP soft-switch solutions. With the help of a common test-bed, this article examines how we can implement a VoIP solution that works now, and that can easily and efficiently interoperate with IMS in the future.

B. Motivation and Reference Scenario

For our discussion we assume a company that has a fully working VOIP/SIP solution with, e.g., telephony, PSTN interconnection, call centre applications, voice mail, presence, office application integration using Telephony Application Programming Interface (TAPI), and more. An implementation of the test-bed incorporating the above mentioned scenario is represented in Fig. 1.

The company has an own domain named “enterprise.com” that is linked via broadband access to Internet. Users can be called by a number belonging to a block of numbers linked to the enterprise, or by SIP URLs such as user@enterprise.com.

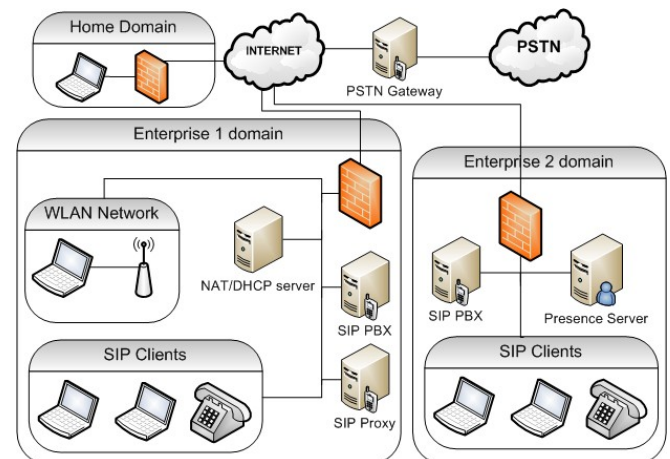


Fig. 1. The test-bed

The company is interested to integrate mobile phones into the enterprise environment. However, these mobile phones are linked to one or several mobile operators and therefore have own mobile numbers. Employees can redirect calls manually to their mobile phones but very often they would rather like to pick up a call on the fixed phone. When on business trips employees sometimes forget to redirect their calls, and some important customer calls might be lost.

Desirable would be a solution (see fig.2) where customers refer to the initially fixed enterprise phone numbers and enterprise SIP URLs, and whilst mobile phone numbers and SIP URLs belonging to the mobile operator would be hidden. Employees are able to pick up calls on their fixed phone, PC or mobile phone.

A second feature would be that the calls are automatically redirected to mobile phones if employees have not registered with the enterprise domain, e.g., because they are on a business trip or not currently in their office (“away

¹ Høgskolen i Agder, Norway

² Teleca Wireless Solutions AS, Norway

from keyboard status”).

If possible the solution should allow an enterprise to “kick” their mobile operator, if price or performance is not satisfying. Further on it should be possible to have business with many operators in parallel to support global business and market presence.

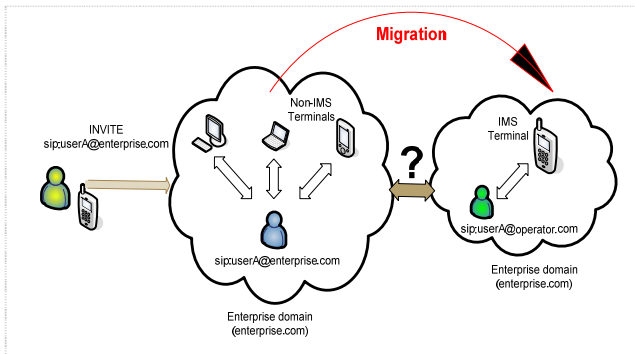


Fig. 2. Reference Scenario

C. A VoIP/SIP Enterprise Solution

A basic enterprise VoIP/SIP solution is illustrated in Fig.3. The key element is a soft switch (SIP PBX) which might be implemented as a combination of several SIP entities [9], such as SIP registrar, proxy server, redirect server, forking server, Back-To-Back User Agent (B2BUA) [1] etc.

SIP clients can be SIP hard-phones or soft-phones on PCs, PDAs etc. A PSTN gateway links the enterprise SIP PBX to the public PSTN.

Enterprise applications, media servers, presence servers, and the VoIP/SIP PBX are interconnected through a company intranet.

For the discussion we want to mention that many of these entities can be operated by the companies themselves or externally through managed service providers and operators.

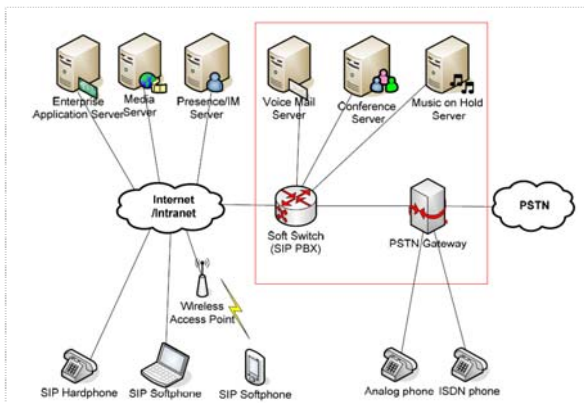


Fig. 3. Logical architecture of a typical VoIP/SIP scenario

The enterprise solution indicated here can be enhanced with value added services like multiple servers to link sub-domains when large enterprises have sub departments in different locations. Administrators may want to use dedicated VoIP servers at each location to handle the communication, avoiding delays and achieving better VoIP performance. This implementation also facilitates load balancing of the SIP PBX, by dividing the user database over several servers. Multiple server issues should be taken into consideration while implementing such VoIP/SIP

solutions for enterprises. VoIP may not be sufficient for enterprises as a complete communication layout. TAPI, presence information and instant messaging are important tools for availability and exchange of information. Furthermore, audio/video conferencing offers flexibility for enterprise users and reduces travelling costs.

A call establishment requirement between two SIP clients can vary based on the location of the SIP clients within the network. Clients located within the same LAN have no restrictions like firewall or Network Address Translation (NAT), hence avoiding the need for traversal technologies. The test setup provides us with real world implementation of such protected networks and enables the testing of transversal techniques without compromising the production/stable network elements.

Simple Traversal of UDP (User Datagram Protocol) through NATs (STUN) has major difficulties with the most common enterprise NAT systems and requires support from the clients. Application Layer Gateways (ALG) and secure tunnels are solutions when it comes to allowing incoming calls, but they are also complex. The final selection of a firewall traversal solution depends on the network structure and security policies of that particular enterprise.

D. Why IMS?

However, almost all IMS services can be accomplished without IMS. So why would anyone care about IMS?

IMS is a new All-IP architecture that provides establishment of end-to-end multimedia services, and a collection of key enablers based on well defined standards. Applications will be able to establish sessions across different access networks, with guaranteed QoS, flexible charging & AAA support. By taking advantage of enablers such as presence and push-to-talk, service developers will be able to create and deploy innovative services faster and more reliably.

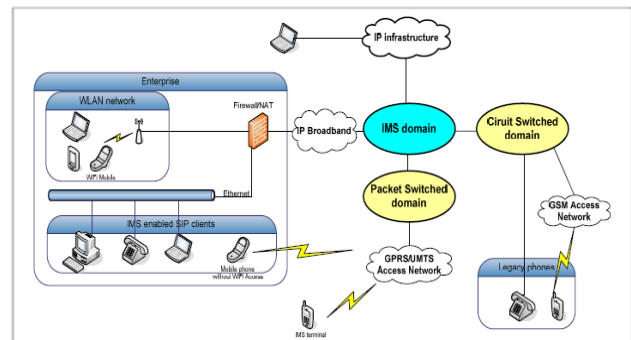


Fig. 4. IMS enterprise solution scenario

An enterprise that wants to integrate mobile devices into their intranet, and that wants world-wide reachability for their employees will need to cooperate with mobile operators. Interconnecting the enterprise domain across B2B interfaces with different operators will be a major integration effort.

Assuming the requirement for an efficient integration strategy, either an operator offering a proprietary B2B interfaces with the lowest service charges or an operator following a global standard with higher service charges at the beginning can be selected.

System integration houses are likely to ask for a high price as they have to write additional customization software. Later, a maintenance contract may turn out to be quite costly for this customized solution.

Even at the same cost, the innovation drive is now dependent on the mobile operator and the system integrators. An IMS solution will always grow in functionality, performance, and better price-performance ratio as manufacturers get a grip on the technology while fighting each other on global market dominance [12].

Therefore, once IMS is well-tested and well-performing, it will be a critical part of future cellular and fixed networks.

E. Possible IMS migration solutions

For the integration of the enterprise network (“enterprise.com”) and the operator domain (“operator.com”) we have now several possibilities.

1) Solution A: Forking

In this solution, the enterprise domain always forks incoming calls to operator.com. The enterprises SIP PBX works as a forking proxy during call setup.

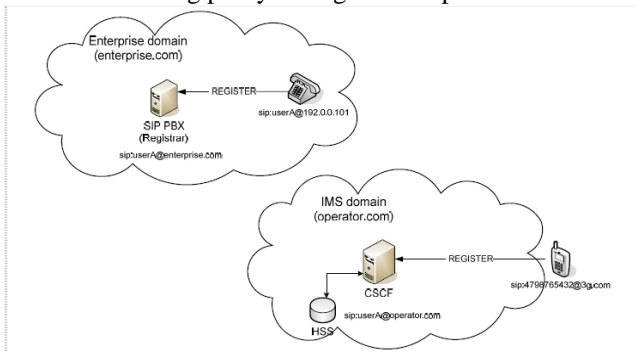


Fig. 5. Registration in forking solution

Users use either a legacy, SIP or IMS mobile phone. Once users switch on the device they are present in the operator domain. Users may or may not register in parallel with their enterprise VoIP/SIP solution.

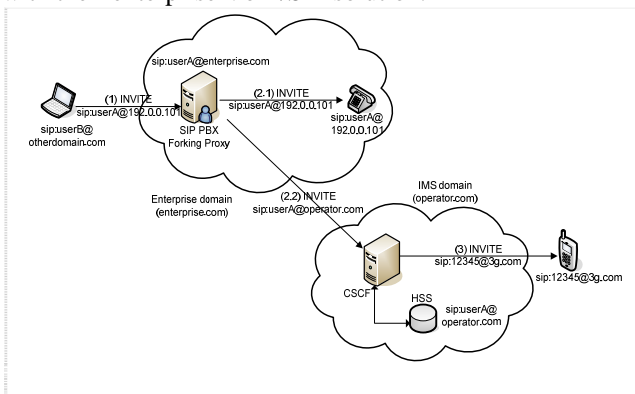


Fig. 6. Basic call setup using forking

Incoming calls to sip:user@enterprise.com are handled by the SIP PBX consulting a forking script. The script indirectly provides information that indicates which of the active devices should ring and when, e.g., the mobile phone may always ring by default, even though enterprise device like PCs or fixed phones are active. The incoming call request is then forwarded to the appropriate and external domain. Calls are forwarded using SIP or other means.

This is the simplest solution. No extra application servers

are needed. Simplicity is an advantage, and it may be used in a modified way even before an operator has introduced SIP/IMS.

As for disadvantage, maintenance of forking lists manually is time-consuming. Furthermore, if the forking list becomes large, network signalling traffic will be increased for parallel forking while calling delay time would be very long for sequential forking.

2) Solution B: Client based

This solution requires an advanced client to inform the enterprise SIP PBX that its location has moved to another domain and that all incoming call to should be redirected to sip:userA@operator.com.

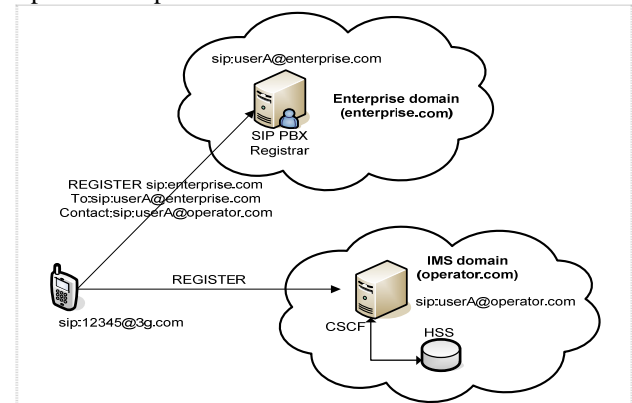


Fig. 7. Registration in client based solution

When a user turns on a mobile terminal, it will register with the operator domain for access. World-wide access is supported thanks to operator-operator roaming agreements. Once able to send IP-packets, the client will contact the enterprise.com domain to register under its new location and thereby inform SIP PBX about the location change.

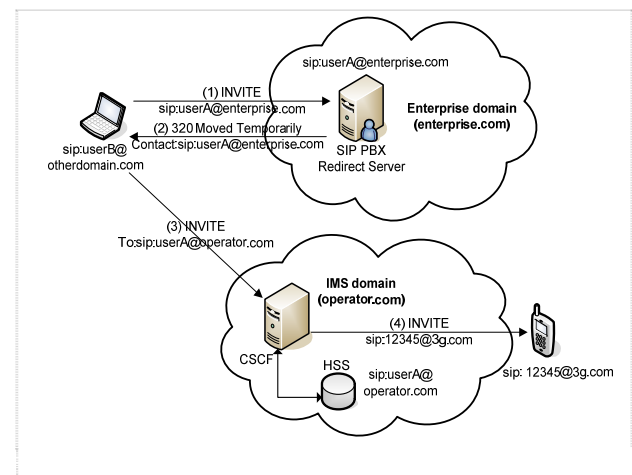


Fig. 8. Basic call setup in client based solution

An incoming call originated by caller UserB to mobile sip:userA@enterprise.com will first reach the enterprise SIP PBX. The SIP PBX will act as a redirect server and, e.g., send a “302 Moved Temporarily” SIP message to caller userB and instruct userB to try userA’s new location sip:userA@operator.com. UserB will initiate a new call to sip:userA@operator.com which is directly sent to IMS domain, and userA’s terminal.

Though a simple solution, it requires significantly more functionality from the client as it must be SIP enabled, and

be able to tunnel securely into the enterprise.com domain for registration.

Legacy mobiles are not supported by this solution as they cannot send SIP messages to SIP PBX to inform location changing.

Possibly an operator could install servers which imitate requests for legacy terminals based on the migration of the link access grant via AAA servers to such a special-purpose SIP server. This will however not be necessary once IMS is deployed (see option D).

3) Solution C: Presence based solution

In this solution, both domains interconnect their presence servers. These two domains have a business agreement so that they can watch the presence status off each other's domain for certain clients.

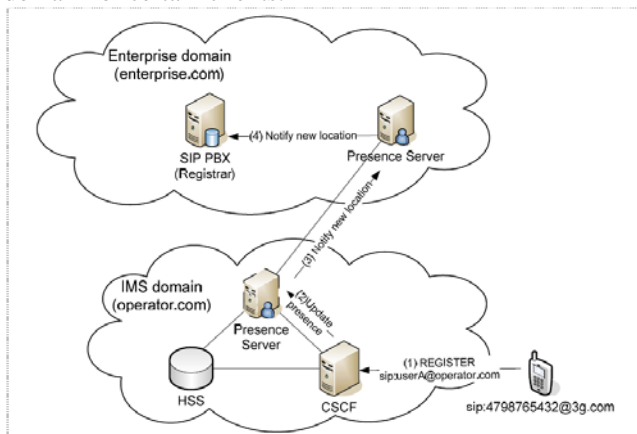


Fig. 9. Registration in presence solution

When an enterprise mobile device is turned on, the presence server in the IMS domain updates the user's presence status to be "online". It will update the presence server at the enterprise with the current status and location: sip:userA@operator.com. When the enterprise SIP PBX gets an incoming call to sip:userA@enterprise.com, the call will be redirected to operator.com.

According to the SIP presence architecture [3] [8] [10] [11], the SIP PBX plays the role as watcher of its Presence Agent (PA), which is the presence server in the enterprise domain. The presence server in enterprise domain works as a watcher of the presence server in IMS domain, and IMS terminal works as a Presence User Agent (PUA).

When userA starts using the IMS domain, he will be registered as sip:user@operator.com. At the same time, the IMS domain updates userA's presence status to online (according to filter criteria for presence service in user profile which is the IMS node Home Subscription Server (HSS)). It will also update his new location in the IMS presence server.

Then the IMS presence server will notify PUAs, i.e. enterprise presence server about the new presence location. Furthermore, the enterprise presence server will notify the new location to its PUA, i.e. enterprise the IP PBX which works as a corresponding SIP Registrar. To setup a call, the SIP PBX will work as a redirect server.

The advantage of this solution is that it is a network based solution which has no requirements for the client, i.e., it can support legacy telephone. Compared to forking, it avoids unnecessary signalling traffic in network. However, it is a

more complex solution. Enterprise and IMS operator should have a presence agreement so that they know each peer's presence status.

4) Solution D: Link Registration

Alternative to presence servers, this solution links registration procedure in both domains.

This solution requires an initial setup as the operator needs to store information that a certain user belongs to a certain enterprise domain uses a certain pre-defined enterprise based SIP URI.

In enterprise domain there is an application server that handles the link registration using subscribe-notify mechanism to do link-register service subscription.

The user profile which is stored in HSS contains a service profile with filters criteria for link registrations. The filter criteria help Serving Call State Control Function (S-CSCF) to decide when to involve a particular AS to provide a service. More information about filter criteria can be found at 3GPP TS 23.218 [13].

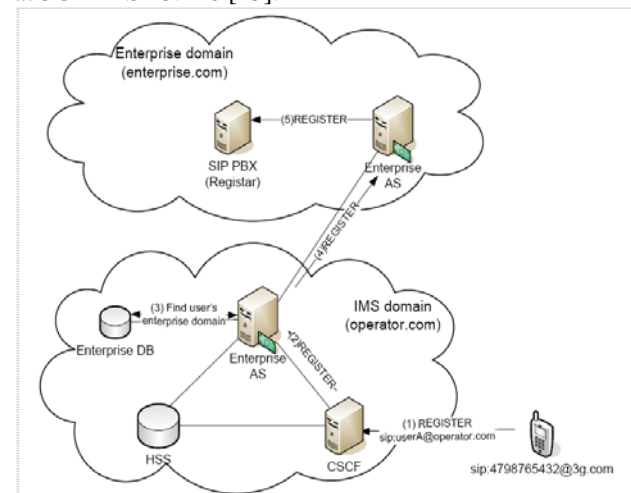


Fig. 10. Registration in link presence solution

When the phone is turned on, it registers with IMS domain. Based on the filter criteria in the HSS User Profile, the S-CSCF will send register information to the Enterprise AS. It will then perform service control procedures, i.e. the S-CSCF will send REGISTER message to Enterprise AS. It will check the database to find user's enterprise domain and send a REGISTER message to Enterprise AS. Now the SIP PBX in the enterprise domain knows that the user is currently registered with IMS domain as sip:userA@operator.com.

The advantage of this solution is that it has no requirement on the client and can avoid unnecessary signalling traffic. But the solution is a more complex and instead of a presence agreement, it needs registration agreement between two domains.

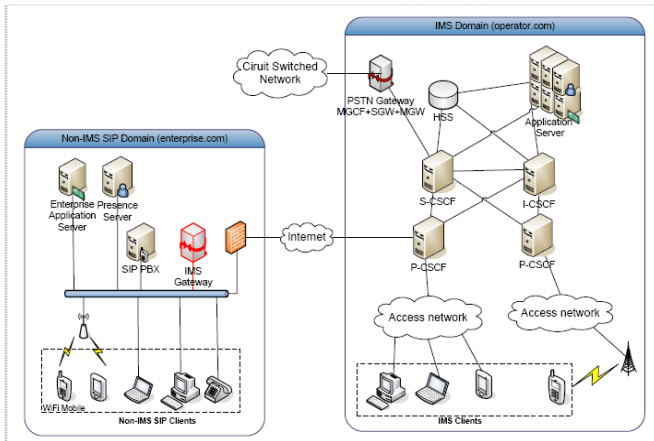


Fig. 11. Network infrastructure in an IMS migration stage.

IMS uses the 3GPP variant of SIP, which needs to interoperate with the IETF SIP because IETF SIP entities on the Internet do not support some of the extensions used in the IMS (e.g. preconditions). To use special IMS services, like “push to talk”, a SIP client needs to support IMS functionality.

An IMS Gateway will play a key role of interconnecting the IMS network and the non-IMS SIP based network. Depending on the scenario such a gateway can be implemented on the enterprise or the operator side.

II. SUMMARY

Considering the large number of future IMS-enabled operators and their well standardized services and enablers, we outlined migration scenarios that motivate preference for IMS operators.

Enterprises that want to integrate mobile devices into their networks need to buy services from mobile operators. If they want to gain freedom of choice for their mobile operator, lower initial integration investment and lower operating costs, a world-wide standard is always preferable. The implementation and testing of such robust systems would require state-of-the-art modular test-beds which are able to simulate and test real implementation scenarios.

With reference to our test-bed implementation, we outlined four solutions for legacy, SIP, and IMS phones. The forking solution may appear first since it is the simplest solution. It requires nothing special from an operator and mobile phone. However this solution is costly in user configuration and puts the entire burden on the enterprise, and is therefore not affordable for all enterprises.

Presence or client based solutions will be suitable for an intermediate phase before IMS is fully deployed. The client based solutions needs an advanced SIP client and secure tunnelling into the enterprise domain.

The presence solution links presence servers in the enterprise and operator domain, and supports legacy phones.

Link Registration is a relative complex solution. Like the presence solution this is a network based solution. It requires an IMS-enabled operator to support registration updates across business borders. This kind of support may not be added by many operators in an early phase of SIP/IMS introduction. So it seems that the link registration solution will appear in the later phase of migration towards

IMS.



Fig. 12. Possible Migration Roadmap towards IMS

This article illustrated how we can implement a VoIP solution that works now and can be easily and efficiently migrated to a complete IMS solution in the future.

According to complexity, functionality and dependency of IMS deployment, these four solutions might appear in different IMS migration phases. Our solutions verified on the test-bed discussed that enterprise users can smoothly migrate towards a future IMS infrastructure.

III. FUTURE WORK

We have now established a test-bed for working with VoIP solutions. Our next step will be to extend the test-bed described above with an IMS sub-domain based on the “Open Source IMS Core”[14]. Here we will focus on the strategies to interconnect SIP and IMS solutions as described above and possibly integrate Wi-Fi clients with it.

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