Mobile Voice Off-Load

An AdvOSS Solution White Paper

Latest version of this white paper can always be found at:

http://advoss.com/resources/whitepapers/mobile-voice-offload.pdf

For more information, contact sales@advoss.com

Mobile Voice Off-load over Wi-Fi Networks

Similar to data off-load, an important emerging use case for mobile operators is the offloading of voice traffic over a companion Wi-Fi network. This service, if provided by mobile operators enables subscribers to seamlessly use a Wi-Fi network in the vicinity of the mobile device for sending and receiving voice calls using Voice over IP (VoIP) protocols. The proliferation of Smart-Phones has made this type of service possible on a mass scale since Smart-Phones are able to run the client side software for VoIP such as SIP based dialers and Soft-Phones. Millions of mobile consumers are already using many Over The Top (OTT) applications such as Skype, Tango, viber etc.

The Wi-Fi network in this scenario may be provided by the mobile operator itself, or through a Wi-Fi service providing partner. This later case is more common since Wi-Fi hot-spot operators specializing in this access technology are emerging that provide complete metropolitan level coverage via their mesh of hot-spot networks for data connectivity at wholesale rates.

Key Benefits of using Wi-Fi over other offload technologies

Although, voice traffic can be offloaded using Femtocell technology and others such as Wi-Max that are used for data traffic offload, there are significant advantages of using Wi-Fi over these technologies. Some of these include:

- Wi-Fi technology is not restricted by the use of licensed bandwidth spectrum
- Femtocells are only deployed in private, indoor environments whereas Wi-Fi hotspots can be implemented in public or private networks and both indoors and outdoors.
- The ubiquitous nature of Wi-Fi today enables operators to use existing residential and business networks with no need for subscribers to invest in new customer premises equipment
- WiMax technology has not proven to be as ubiquitous and cheaper to deploy as Wi-Fi

Using SIP based VoIP for Voice offload

VoIP technologies have matured tremendously over the past several years for providing sophisticated, mass scale voice services. These technologies have moved way past the simple vanilla voice service that was touted as a cheap alternative to Public Switched Telephone Network (PSTN) voice. Now, advanced enterprise and consumer value added services and applications have been made possible due to extensive standardization efforts by 3GPP, IETF and other bodies over SIP based technologies. SIP has been standardized as the protocol of choice in the next generation all IP based fixed and mobile networks such as IMS, LTE/EPC and NGN.

Value addition of SIP has been enhanced further by development of technologies and standards for Presence, Instant Messaging, Conferencing, Push-to-Talk to name a few. Furthermore, capability to integrate SIP based applications with web-based applications and services has opened up a whole new and largely un-tapped world of sophisticated services for both consumers and enterprise markets.

SIP based voice offload must be seamless

To reap the real benefits of SIP based voice offload, the service must provide as seamless a user experience as possible for making and receiving voice calls. The same phone number as the subscriber's cellular number and the phone's native address must be used for sending and receiving calls. The subscriber must not have to do any unnecessarily complicated or even extra steps for making VoIP calls other than may be giving subscribers a choice dialog whether they want to make the call over WiFi. This can also be achieved via one time setting in the mobile phone application that subscriber should be able to change, so that this dialog based choice is not necessary for every call.

Key Benefits of SIP based voice offload

Keeping in view these significant advancement in SIP based VoIP technology, there are major benefits for mobile operators to use SIP based VoIP for voice offload:

- Use of Wi-Fi connections for SIP calls enables operators to offload voice calls from the already congested cellular network

- By offering attractive rates and better Value Added Service (VAS) bundles, operators can encourage subscribers to use their cell-phone instead of their regular landline for calling from home and residential areas

- Operators can motivate the subscribers to use their native mobile service on VoIP instead of OTT alternative VoIP services e.g. Skype, Viber etc. by offering reduced rate calls and better Quality of Service (QoS) and Quality of Experience (QoE). This reduces customer churn and improves customer loyalty

- On-Net calls can be made at a fraction of the cost of cellular On-Net calls between two SIP based registered clients embedded in Mobile Phones. This is because On-Net calls only consume bandwidth associated with the data center where SIP Servers have been hosted.

- The offload service can be further enhanced by enabling sending/receiving of text messages (SMS) over the same Wi-Fi connection.

- Subscribers can make and receive calls on or from their cell-phone, even in areas with poor cellular reception if Wi-Fi network is available in the vicinity.

- Roaming can be provided at much cheaper rates, even the same rates as regular VoIP calls in many cases. This would significantly improve customer loyalty, reduce churn and increase call volumes.

- Since SIP based voice offload allows cellular subscribers to initiate and receive voice calls using the same phone number as embedded in their SIM module over a Wi-Fi based IP network, the subscribers find this to be much less hassle since they don't have to distribute a special number or user identifier for VoIP which they have to do in case of OTT VoIP applications such as Skype.

- SIP based Soft-Switches and application servers can provide much more sophisticated VAS services for consumers and enterprises. Mobile operators have the potential to enhance their service offerings to enterprises and consumers with advanced features and complete end to end applications and monetize them for new and un-tapped revenue streams. This is one of the major benefits for mobile operators who are desperately looking for new services and tap new markets for mobile consumers instead of engaging in call rate wars with other operators in commodity voice.

- Mobile operators who are planning to migrate to 3G/4G technologies could make themselves fully ready to embrace them since their main source of revenue stream i.e. voice shall be already shifted and tested to a large extent with new all IP based technology and services, that is the main promise of 3G/4G technologies. The only thing they would need is to replace the existing cellular access network based on GSM, UMTS or CDMA etc. to technologies like LTE. The application level core voice infrastructure would already be in place.

Media Routes Voice offload Solution

Media Routes voice offload solution is a comprehensive core network solution for Mobile Voice Offload. It provides all the core network side components to realize a successful offload strategy over any pre-deployed Wi-Fi network. For client side SIP based mobile applications, Media routes partners with several Mobile VoIP Soft-Phone suppliers. Mobile Operators are free to choose their own supplier for client applications as well, in which case Media Routes core network side components would fully integrate with the client application over SIP and related protocols.

The end to end solution consists of the following components:

- A SIP based Soft-Phone Application for on multiple Mobile platforms such as Android, IOS and Windows based Smart-Phones, supplied by third party vendors and Media Routes partners

- A SIP based, combined Soft-Switch and Session Border Controller
- A SIP based Application server built upon Media Routes Service Delivery Platform.
- Integrations with Mobile Operator's Core Network elements

We give a brief description of each solution component below:

A SIP based Soft-Phone Application

These are SIP user agents that are embedded in the Smart-Phone as an application. This application is invoked when the user tries to make a call and the mobile phone is configured to use the SIP client for placing calls, or receives a call over a Wi-Fi data network. The soft-phone application is supplied by Media Routes partners or any other third party vendors of the Mobile Operator's choice. They should however, be available on mainstream platforms such as Android, IOS and Windows 8 Mobiles.

SIP based Soft-Switch and Session Border Controller

This is a highly scalable, high performance SIP Server with complete SIP Registrar, Proxy, Redirect and Routing capabilities. It also provides Access Control, NAT traversal, SIP trunking and peering control features. It interacts transparently with its companion SIP Application Server to realize complete Core network functionality for Voice Service Delivery. More information on SIP Server can be found at:

http://www.mediaroutes.com/Session-Border-Controller.html

SIP based Application Server

This is a feature rich Application Server that provides several fully functional, out of the box SIP applications for realizing a core voice service with value added class 5 features. The Application Server is available in many flavors and versions that can handle advanced Value Added voice services for consumer as well as Enterprise markets.

The Application Server is built upon Media Routes Service Delivery Platform (SDP). As a consequence, it is fully programmable and extensible via simple scripting language exposed by SDP. This makes it a very powerful offering for Mobile Operators as they can enhance the basic voice service by programming the SDP itself and offer further value addition to their offloaded voice customers at a brisk pace to create differentiation from the competition.

More information on Media Routes Service Delivery Platform and SIP Applications Suite can be found at:

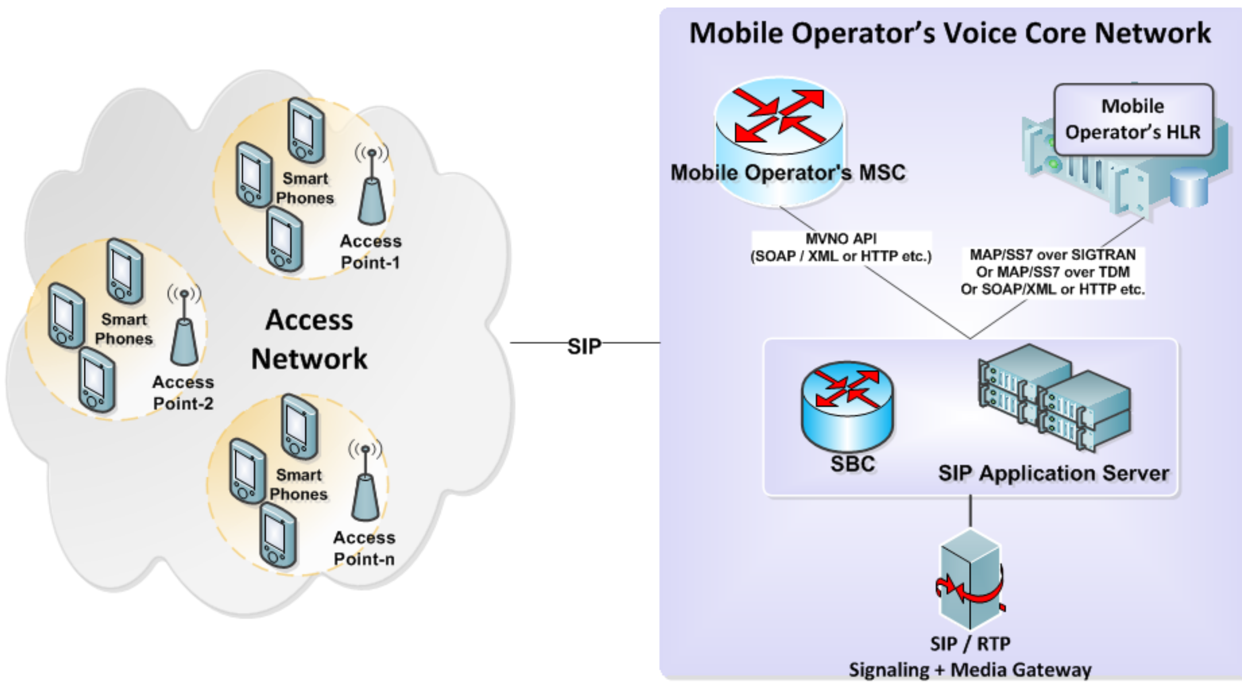
http://www.mediaroutes.com/Service-Delivery-platform.html http://www.mediaroutes.com/Applications.html

Integrations with Mobile Operator's Core Network elements

There are two options for deployment model for voice offload solution:

Option 1—Use the existing Mobile Switching Center (MSC) for handling calls

This is the simplest model of voice offload deployment. In this model, calls are actually handled by the MSC by shifting them back to it on a SIP trunk from the VoIP network. In this case, all prepaid, post-paid and other services scenarios are as before for the customers with no additional services available or provided. The rating, charging and Value Added Service Delivery scenarios are exactly the same as before.



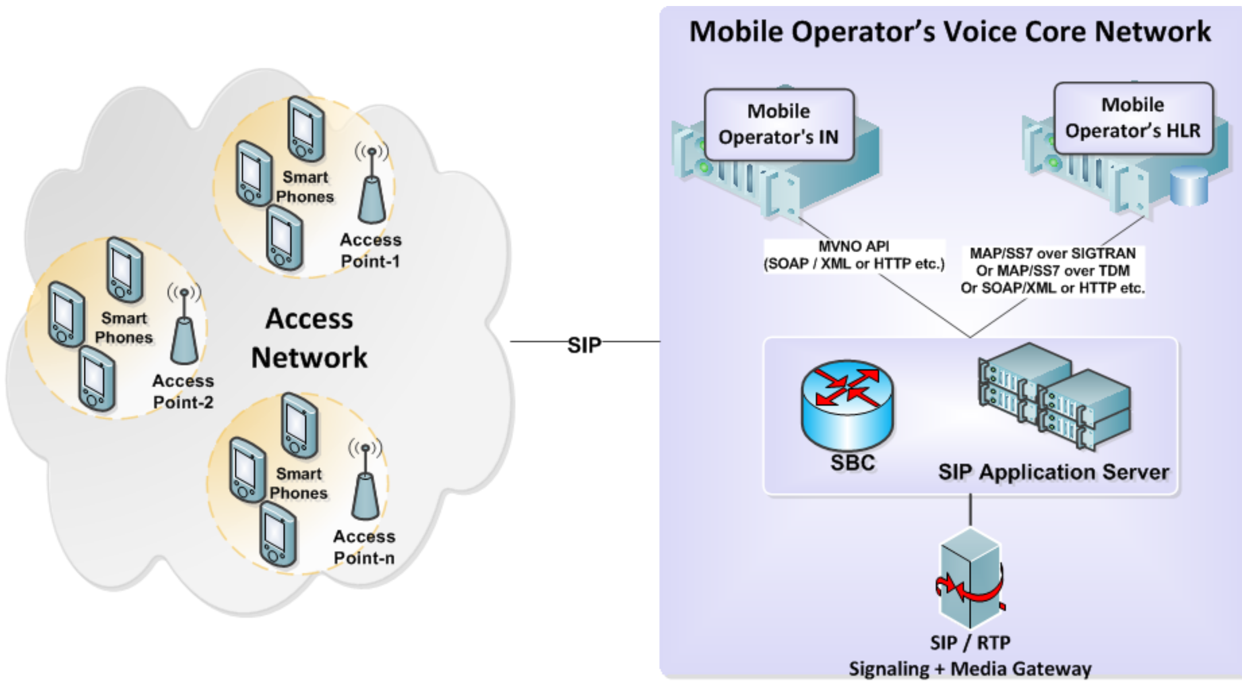
An outgoing call, in this scenario is routed directly to the MSC as soon as it hits the SBC. To the MSC it appears as an outgoing call from the calling subscriber to the desired destination. The only integration required in this case is for the SBC to interoperate with MSC on a SIP trunk.

An incoming call, in this scenario is handled by the MSC first as usual. The MSC consults HLR for subscribed services for the user. If the HLR has a flag set indicating that this user is registered at that point in time on the VoIP network, and has a preference set for receiving calls on VoIP whenever possible, it routes the call on a SIP trunk towards the SBC. The call is handled by SBC from there on as an incoming voice call and is routed after looking up the user's location in the location database.

This requires integration with HLR and SBC sets and resets the relevant flags to indicate registration and de-registration events on the VoIP network.

Option 2—Using the VoIP network to provide Enhanced and Value Added Services

This option is useful when the mobile operator intends to provide additional VAS services and enhanced capabilities through VoIP network such as HD Voice, Video Calling, IP Centrex for Enterprises etc. to consumer or enterprise users. In this case, the operators may want to harness the full power of VoIP technologies and provide differentiation and value addition as additional benefits of offload.



This scenario requires the following integrations:

Customized integration with Mobile Operator's OSS/BSS systems

The solution needs to be integrated with Mobile operator's OSS/BSS systems. These mainly include the following possible integrations:

Home Location Register (HLR)

This is required to meet the following use cases:

- Authentication of outgoing voice calls or Authentication of user at the time of Registration on the SBC requires integration with HLR. The Application Server or SBC retrieves user credentials from HLR e.g. secret key for authentication against the mobile number. Authentication works in SIP through a Digest based authentication method that requires the same secret key to be used at both ends without exchanging it on the wire (or on the air).

- For incoming calls, the requirement for HLR integration is the same as for Option 1 described above i.e. mobile operator's MSC needs to send the call to SBC/SoftSwitch if the user's phone is registered via SIP on the SBC, and the user has set the preference of receiving calls on VoIP whenever possible. In this case, the SBC would set a service flag in HLR whenever the SIP soft-phone registers and reset that flag when the user de-registers (goes offline).

Prepaid IN

Media Routes SIP Applications support complete prepaid charging model via quota reservation, balance authorization based on destination etc. To support these use cases however, the solution needs to integrate with mobile operator's IN system to meet the requirements for prepaid charging. The IN may need to expose a few APIs to meet the following use cases:

- Balance update (increase or decrease)
- Balance reserve (quota management)
- Balance retrieval

Customized, Service Level CDR generation

Generating voice offload Call Data Records (CDRs) is a critical requirement for Mobile operators. These CDRs may need to be mediated and fed to the billing system for rating A Lightweight built-in Mediation Engine for generating customized and fine grained service usage level CDRs for Mobile Operator's billing system.

Note that for option 1, although VAS services may not be provided by the solution, CDRs still need to be generated s that they can be reconciled, and extensive reporting can be provided about the offloaded traffic in terms of usage statistics and user behavior patterns.

Subscriber Provisioning and CRM

Subscribers need to be provisioned in the system for providing VAS and other enhanced services. This requires integration with mobile operator's CRM and provisioning systems and workflows. Media routes solution exposes service and subscriber provisioning APIs for this type of integrations over well known HTTP based interfaces such as SOAP/XML and REST.

Mobility in Voice Offload

Mobility is managed at two levels in the Voice Offload Solution:

1. Wi-Fi Mobility
   In this type of mobility scenario, SIP client application detects that the IP address of the Wi-Fi interface has changed on the mobile phone. It immediately sends a SIP Re-INVITE message to the other end-point of an ongoing session and the SIP session, along with RTP streams shift to the new IP address. This type of mobility takes a few seconds to complete and the user may experience a short interruption in voice communication. Another alternative for the client application might be to shift the voice call to the regular cellular network i.e. take it off the Wi-Fi. This may reduce time of media interruption and improve the user experience. However, both these cases require intelligence in the SIP client application embedded in the mobile device.
2. Multi-device Mobility
   This case deals with providing mobility across multiple user devices, registered under the same phone number. In this scenario, the SIP Server or Soft-Switch needs to support the mobility use cases. First the Soft-Switch needs to support simultaneous registration of multiple devices. On receiving an incoming call, the Switch must be able to reach all devices via SIP forking techniques or in serial fashion in some priority according to user preferences. Similarly, the soft-Switch must be able to transfer calls dynamically between different devices on user's input via some hot-keys.
   Media Routes solution supports both the above mobility scenarios in the server SIP Application server. It handles client generated re-invites properly passing it on and reconfiguring media streams on its end. Similarly, it supports forking incoming calls to multiple user devices as well as user initiated transfer on on-going calls between devices.