Multimedia Service Platform

Turnkey solution for SIP based communications

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Current telecommunications landscape

- From the old PSTN only the E.164 numbering plan remains
- SS7 and ISDN (circuit switched) networks reached their end-of-life
- In the end it will be one “all IP” global network, the Internet
- Replacement for signaling protocols has been proposed by IETF and ITU, first one was H323 (ITU) and SIP came second (IETF)
- ENUM complements SIP and allows IP devices to be reached from the PSTN without programming switches with numbering plans
- Alternatives are H323, MGCP/Megaco, Skype, Skinny or IAX.

All vendors support SIP today, it is a common denominator in the vendor landscape
SIP - the Session Initiation Protocol

- SIP is a horizontal, it allows end-points to find each other and initiate sessions over the Internet (any type of sessions)
- Intelligence is distributed among participating nodes, new applications can be rolled out without upgrades in the network
- SIP makes use of existing proven protocols for addressing, security and scalability, thus it did not re-invent the wheel (DNS, HTTP, TLS)
- NAT traversal has been addressed in SIP (not H323)
- SIP coexists happy with all other protocols (all vendor support)
- SIP replaces telephone numbers with e-mail addresses

SIP enabled Convergence between fixed and mobile networks, you can change access media, device or provider and it still works
Convergence and NGN

- Convergence means from Telecom + Internet we end up with one network alone, NGN is the Internet
- Mobile fixed or cable converge into the services at the end-user
- Internet is the dumb network the service is at the edge and not in the center
- The end-user is the center and not the carrier/provider
- You stay in business as long as your service is interesting enough
- You cannot build walled gardens anymore

This means new business models need to be developed
Open garden versus Walled garden

- Walled gardens so far have been motivated by poor implementations, lack of clarity over new business models, the push of Session Border Controllers

- PSTN and E164 numbering plan were a success because one was able to connect to everybody else (universal service)

- Is important to allow dialing of both E164 numbers and SIP URIs

“By the time you finish building up your walled garden, the customers will be safely outside” - Adrian Georgescu

“If you can dial a SIP URI and you can be called by a SIP URI than you have access to VoIP otherwise is just an emulation of classic PSTN” - Henry Sinnreich
Exiting business models based on SIP

SIP enables several selling points today:

- PSTN termination service with per minute charging
- ADSL service complement
- SIP devices
- Telephone numbers sold separately
- Company IP PBXs interconnected via IP
- SIP service subscription

But PSTN services will be short lived services, you must change a bit the mentality
Exiting business models based on SIP

The current approach of VoIP service providers is wrong:

- They compete to reach to bottom, till price per minute equals zero
- Incorrectly marketing VoIP as free, while somebody has to pay for it
- Applying telco logic to Internet does not work (see IMS design)
- Build walled gardens while the end-user may opt-out from its device
- Few telco comes out with innovative services
- They do not provide end-user with a SIP address, but only a classic telephone number

When everybody left the PSTN and VoIP is free where is the cake?
New business models based on SIP

SIP is more than Voice over IP. New services will emerge some of them you will have never heard before of:

- Mobility
- Presence and contextual communications
- Notifications based on business triggers
- Identify and certificate management, privacy services
- Synchronize data among devices
- Address translation (ENUM and domain names)

The cake is there you just need an enabler and that is SIP
Ingredients

Operators deploying SIP must address the following:

- SIP signaling - SIP Proxy / Registrar / Redirect
- E164 numbering resources and interconnect with the PSTN
- Far-end NAT traversal capability for devices behind NAT
- CDR mediation, accounting and tracing
- Provisioning interfaces
- Number portability and peering with other SIP providers
- Emulation of existing telephony services (class 5)
- Applications (Voicemail, Conferencing, IM and Presence)
- End-user devices (soft phones, hard phones)
Implementation scenarios

Operators may chose between:

1. Collocated: Implement infrastructure from scratch, either through an integrator or direct vendor selection (2 to 6 months)

2. Hosted: Find a white-label technology vendor and use a hosted service, this approach is good to gain time to market while implementing 1

3. Developed: Hire skilled engineers and develop systems based on Open Source components (6-12 months), this approach has unpredictable results
Requirements

When selecting the vendors there are key issues that must be addressed by the service provider:

- Compliancy with standards
- Support industry best practices (Trials, interoperability)
- Knowledge base
- Solution scalability
- Integration with other systems
- Availability of source code and debugging tools
The proposition

One solution for implementing SIP services using collocated, hosted or a combined model

• Complete solution directly from the vendor (predictable end-result)
• Based on proven components developed in Open Source model (large knowledge base)
• Trials during 2003, enjoys production status since 2004
• Low maintenance cost (one FTE can run and maintain the platform)

Commercial SIP services are now available based on it:
@Call (ARCOR), Talkin2YA (Budget Phone Company), Eurovoice (Euroweb International), SIP2GO (Sentiro)
Components

Multiple components work together smoothly to achieve the goal

- SIP signaling: OpenSER
- NAT traversal: MediaProxy
- E.164 number translation and number portability: ENUM
- Mediation and accounting: CDRTool
- Accounting: FreeRadius
- Database: MySQL
- Provisioning system: NGN-Pro (SOAP/XML API)
- Voicemail and voice to email: Asterisk

The goal is subscriber in, invoice out
Deployment track

During installation the Operator can learn how to build the platform, which is useful for training or disaster recovery purposes.

The following items are delivered with a standard installation:

- Installation log
- Source code
- Documentation for individual components
- As-built schematic (platform blueprint)
- Sample client for remote SOAP/XML provisioning
Managed services

AG Projects hosted facility provide fast deployment of SIP services

- SIP audio, video and IM
- White-label VoIP platform - all included (SIP signaling, NAT traversal, ENUM, provisioning and accounting), time to go to market is minimum (rollout in 2-4 weeks)
- Work with external PSTN gateways (like Cisco) or SIP termination carriers (like MCI or Level 3) or your SIP PBX (Asterisk)
- All you need is to print your own invoices
Provisioning interfaces

The following functions are available out-of-the-box on the hosted platform:

- Add/modify/delete customers
- Add/modify/delete SIP accounts
- Add/modify/delete DNS zones
- Add/modify/delete ENUM mappings
- Customizable SIP end-user control panel (SIP settings page) where end-users can login and access their own account
- Access to Call Detail Records, SIP traces and rating engine

SOAP/XML provisioning engine allows the development of web sites hosted outside the platform
Scalability

Multimedia Service Platform is able to grow with your business needs:

- Multimedia Service Platform is designed for operators with maximum 50,000 subscribers

- SIP Thor is the upgrade path to scale to millions of subscribers

SIP Thor is AG Projects solution for survivability and scalability of SIP communications to serve a base of multi-million subscribers.

SIP Thor provides a more flexible and lower cost alternative to the IMS concept proposed by 3GPP and enables next generation SIP services like user mobility, video, IM and Presence.
Supported SIP devices

All SIP compliant software and hardware phones are supported

ATA adaptors

Software phones

Video Phones

WiFi Phones

Desktop phones

Asterisk PSTN gateways
SIP settings page

Call forwarding
Do not disturb
Go to meeting
Time based forwarding
Short dial codes
Privacy control
Selective call accept
Selective call reject
Show online devices
Show last calls
Phonebook
Voicemail settings
### Access to Call Detail Records, traces and rating

#### CDRTool Express Router (Global Switch)

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<th>Number</th>
<th>Time</th>
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<th>In SIP destination</th>
<th>Out</th>
<th>Date</th>
<th>Price</th>
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#### Signing information

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- **SIP Method**: INVITE
- **SIP Status**: Ok (200)
- **SIP From**: sip:31171130390@budgetphone.nl
- **SIP To**: sip:311653770120@budgetphone.nl
- **SIP Call-ID**: 711189240 (Privacy disabled)
- **SIP from (callset):** 2005-08-10 19:06.06 Europe/Amsterdam
- **SIP to (callset):** 2005-08-10 19:06.06 Europe/Amsterdam
- **Caller ID**: Ok (200)
- **Called ID**: +316523770120 (Nederlands mobile 31653)
- **Called User**: Asterisk PDK
- **Signalling**: H.248
- **Destination Name**: Nederlands mobile

#### Media information

- **Application**: Audio
- **Codec**: GSM
- **Color**: Telen 2 Yearbook 1193m
- **Duration**: 52 s
- **Duration**: 53 s
- **Duration**: 54 s

#### Rating information

- **Rating**: 5.6 / 5.0
- **Rating**: 5.0 / 5.0
- **Rating**: 5.0 / 5.0

#### Billing Party

- **Billing Party**: 31171130390@budgetphone.nl
- **Billing Party**: 31123110322@budgetphone.nl
- **Billing Party**: 31123110322@budgetphone.nl
- **Billing Party**: 31171130390@budgetphone.nl
- **Billing Party**: 31123110322@budgetphone.nl

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Logged in as adriang (Adrian Georgescu) | Logout
### Management of subscribers

One central place to manage any type of record (DNS, SIP, ENUM, Email)

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<thead>
<tr>
<th>Type</th>
<th>SIP accounts</th>
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<th>All record types</th>
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</table>
Platform blueprint
This presentation is available at:

http://ag-projects.com
Questions?

Thank you,
Adrian Georgescu
sip:ag@ag-projects.com