Multimedia
Service Platform

Turnkey
SIP infrastructure

Document assembled by
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Executive summary

Multimedia Service Platform and its SIP Thor variant are scalable turnkey solutions for delivery of SIP services like Voice and Video over IP, Instant Messaging and Presence to residential and SME markets.

The platform is based on IETF standards and integrates the best-of-breed Open Source software components available on the market. The platform is being used for more than five years in production environments and has an excellent track record. The platform can scale to serve up to 3 million end-points.

Features

- Voice and video calling based on IETF SIP protocol (RFC 3261)
- NAT traversal support for both signaling and media
- Presence based on SIP SIMPLE (PUBLISH/XCAP/XDM)
- Session based IM (MSRP protocol including relay extension)
- Transparent for SIP applications in the end-points
- Provisioning server (SOAP/XML)
- DNS based addressing, multiple domain enabled
- Multiple SIP devices per SIP account (parallel forking)
- Multiple telephone numbers and aliases per SIP account
- Rich class 5 telephony features (call forwarding, voicemail)
- Rating engine for postpaid with anti-fraud system or prepaid
- Rating engine with tariffs per SIP account, SIP domain or IP address
- Real-time web based SIP and RTP tracing
- Least cost routing (LCR) and ENUM peering engine
- Multimedia Ad-hoc Conferencing (Audio, Chat, File Transfers)
- XMPP protocol gateway

The beneficiary of Multimedia Service Platform is referred in this document as the “Operator”.

This proposal outlines the components of the platform and delivery schedule.

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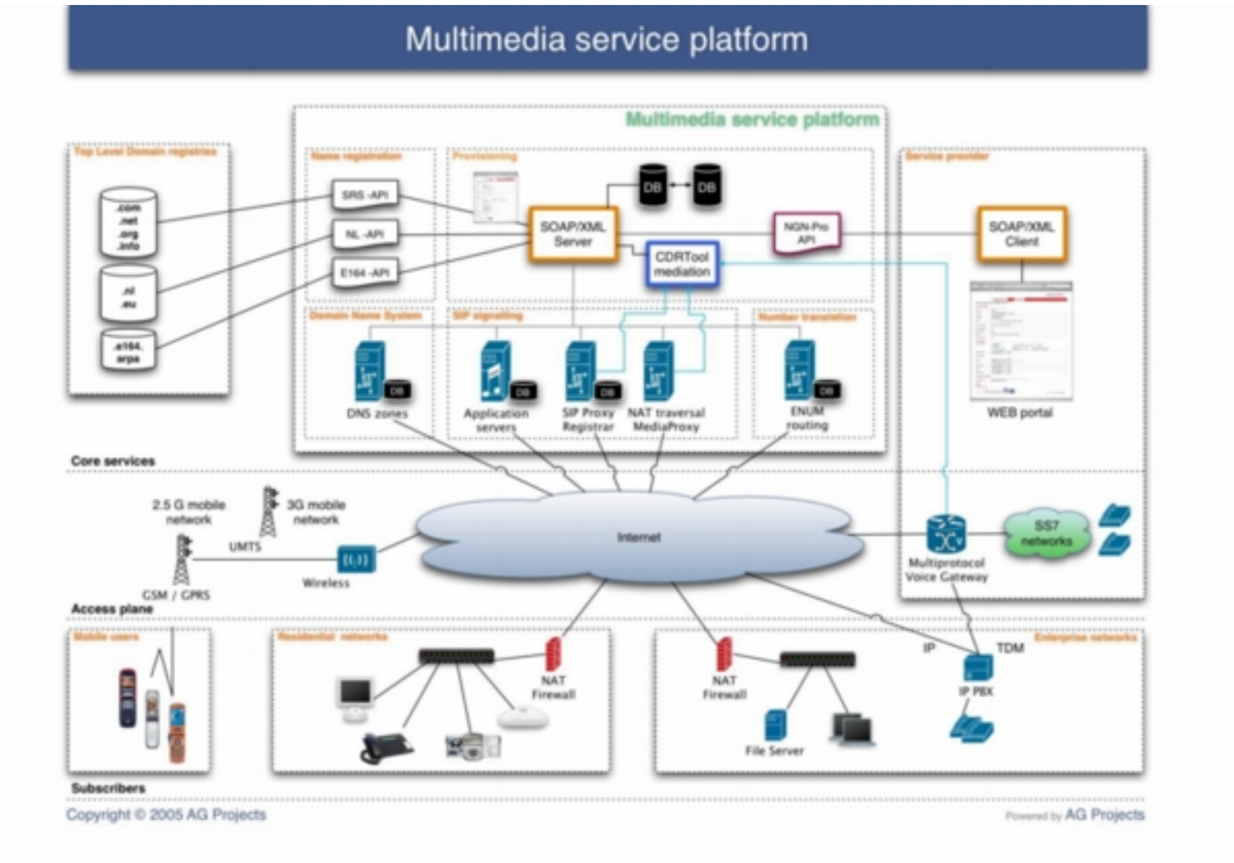
Platform components

	Function	Solution	Vendor
1	SIP Proxy/Registrar/Presence agent	OpenSIPS	AG Projects
2	RTP media relay	MediaProxy	AG Projects
3	CDR generation	FreeRadius	FreeRadius.org
4	CDR mediation and accounting	CDRTool	AG Projects
5	E.164 number translation	ENUM system	AG Projects
6	DNS server	Power DNS	PowerDNS.org
7	Database backend	MySQL	MySQL.org
8	Presence policy	OpenXCAP	AG Projects
9	SOAP/XML Provisioning	NGNPro	AG Projects
10	Voicemail	Asterisk	Digium.com
11	Operating system	Linux	Debian.org
12	IM/File transfer relay	MSRPRelay	AG Projects
13	Scalability/ high availability	SIP Thor	AG Projects
14	Conferencing	SylkServer	AG Projects

Platform diagram



Multimedia Service Platform requires little to no maintenance. Upgrades of software components can be planned and performed on stand-by machines with minimal service disruption[1].

When complemented with an alarm system for all critical services, the platform can be safely operated by one FTE[2].

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[1] Downtime depends on the take-over time among cluster members
[2] Full time equivalent with good knowledge of administration of Linux operation system

SIP Proxy / Registrar / Presence agent

The SIP signaling component used by Multimedia Service Platform is based on OpenSIPS, a SIP Proxy/Registrar/Presence agent server.

OpenSIPS, is a variant of the open source project SIP Express Router (SER) started back in 2001 by Fraunhofer- Fokus Institute in Berlin, changed to OpenSER in 2005 and finally into OpenSIPS in 2008.

AG Projects is an active developer of OpenSIPS project since its inception. For more information see http://OpenSIPS.org

Features

- Multiple domains
- SIP aliases
- Network asserted identity
- ENUM routing support
- Support for NAPTR and SRV records
- Parallel and serial forking
- CDR generation
- SIP protocol tracing
- Lawful intercept (signaling)
- Emergency calling
- Presence and IM
- PSTN Lease cost routing
- NAT traversal support

OpenSIPS configuration provided by AG Projects handles properly all supported SIP call signaling flows while eliminating possible routing loops and providing accurate accounting. More information is provided in the Call control engine section.

Numbering translation is realized using a DNS system based on ENUM protocol. More information is provided in the ENUM section.

AG Projects provides integrated CDR mediation and accounting for OpenSIPS. CDRTool does the CDR mediation including real-time normalization, rating and provide accounting information for all components used (SIP Proxy, inbound and outbound gateways). The process of CDR collection, normalization, rating and export into external applications has been fully automated. More information is provided in the CDR mediation solution section.

A central Provisioning engine based on SOAP/XML is able to both push and pull information to and from the individual systems of the platform including the SIP Proxy/Registrar.

CDR mediation and accounting

For CDR mediation and accounting, AG Projects provides CDRTool.

CDRTool consists of a rating engine and an easy to use web application, which can be used with minimal training of the helpdesk and operations staff. It can also be used for troubleshooting purposes to access in real-time SIP signaling information, provide statistics grouped by criteria like release cause, destination or billing parties.

The real-time SIP tracing facility can help pin-point SIP call flows failure reason. Using CDRTool, the response time for customer support center for SIP related problems could be lowered dramatically. End-users may have access to own Call Details Records in a web page.

Rating

CDRTool has on the fly rating capabilities that may complement or replace external rating solutions available in the operator network. By using it, Multimedia Service Platform delivers rated CDRs.

Rating plans based on time of day and day of week can be specified per SIP account, SIP domain or source IP address basis.

For rating calls, which span multiple time intervals, the right rate is selected and applied for the call duration within each time interval. Each customer may be assigned its own dedicated rating plans destination id and names.

CDRTool rating engine is available via the network so that it can be used also by third-party applications.

CVRTool rating engine is used by the call control prepaid application, both prepaid and postpaid calls are rated and displayed in the same way.

CDRTool provides accurate accounting for call detail records generated by OpenSIPS used in combination with MediaProxy far-end NAT traversal regardless of the presence of BYE messages.

Anti-fraud

CDRTool has anti-fraud mechanisms to block OpenSIPS accounts that exceed a predefined quota. User quota can be based on cost, number of calls, number of minutes or network traffic. Several mechanisms are available for detecting and mitigating denial-of-service attacks.

Data export

CDRs can be exported in CSV (Comma Separated Values) format. Export can be scheduled or executed on demand in CDRTool web interface. Direct access to MySQL data storage is possible. Data like last missed/received/placed calls may also be queried via SOAP-XML[3].

Screen samples

[3] Using NGNPro

CDRTool Powered by AG Projects

SIP Express Router (Global Switch)

Call search | Rating tables| Log| Accounts | 2005-04-14 09:28:03 (Europe/Amsterdam) | CDRTool 3.0 Logged in as **adriang (Adrian Georgescu)** Logout

Refine search | Refresh | Export results to file | Save a description for this query: [ ] Save

51 records found.

Found 2 CDRs for normalization.

From 2005-04-14 08:17 to 2005-04-14 23:55

	Session start time	SIP caller	In	SIP destination	Out	Dur	Price	KBIn	KBOut	Status	Codecs
1	2005-04-14 09:27:56	31307110326@budgetphone.nl	In	+31184493030 (Nederland 31)	Out	00:00		0.00	0.00	Ok (200)	
2N	2005-04-14 09:26:13	31307110326@budgetphone.nl	In	+31306572935 (Nederland 31)	Out	00:59	0.0348	1,331.64	703.83	Ok (200)	G711u

Signalling information
SIP Session: 639b1c156be197df3cc9037a0fbabb32@budgetphone.nl
SIP Method: Invite from *80.60.170.108:5060*
SIP Status: Ok (200) DelayTime: 11(s)
SIP From: sip:31307110326@budgetphone.nl
SIP To: sip:0031306572935@budgetphone.nl
PSTN Caller ID: 307110326 Privacy disabled
Start time (caller): 2005-04-14 09:26:13 Europe/Amsterdam
Start time (proxy): 2005-04-14 09:26:13 Europe/Amsterdam
Stop time (proxy): 2005-04-14 09:27:12 Europe/Amsterdam
Session duration: 00:59
SIP Proxy: 81.23.228.139
SIP Canonical URI: sip:0031306572935@voipgw02.budgetphone.nl
Next SIP hop: sip:0031306572935@voipgw02.budgetphone.nl
Destination name: Nederland
Billing Party: 31307110326@budgetphone.nl

Media information
Application: Audio
Codecs: G711u
Caller UA: Talkin 2 Ya release 1103m
Called UA: Cisco-SIPGateway/IOS-12.x

Rating information
ConnectFee: 0.0200
--
Span: 1
Duration: 59 s
Appl: PSTN voice (130270 KB)
Destination: 31 (Nederland)
Customer: domain=budgetphone.nl
StartTime : 2005-04-14 09:26:13 (Europe/Amsterdam)
ProfileId: 441 for weekday
RateId: 441 for 8-19h
Rate: 0.0150 / 60 s
Price: 0.0148

	Session start time	SIP caller	In	SIP destination	Out	Dur	Price	KBIn	KBOut	Status	Codecs
3N	2005-04-14 09:25:31	31307110326@budgetphone.nl	In	+31167560509 (Nederland 31)	Out	00:04	0.0210	73.83	74.92	Ok (200)	G711u
4N	2005-04-14 09:24:49	654271273@voipgw01.budgetphone.nl	In	31135802460@budgetphone.nl		00:12		65.16	64.92	Ok (200)	G729
5N	2005-04-14 09:23:20	31307110326@budgetphone.nl	In	+31497681754 (Nederland 31)	Out	00:00		0.00	0.00	Timeout (408)	
6N	2005-04-14 09:23:14	251659353@voipgw01.budgetphone.nl	In	31237110301@budgetphone.nl		00:32		184.57	184.45	Ok (200)	G729
7N	2005-04-14 09:22:10	713315247@voipgw01.budgetphone.nl	In	31717110310@budgetphone.nl		00:00		0.00	0.00	Canceled (487)	
8N	2005-04-14 09:21:16	31337110304@budgetphone.nl	In	+31334570606 (Nederland 31)	Out	01:49	0.0473	2,137.50	1,128.05	Ok (200)	G711u
9N	2005-04-14 09:20:57	412629571@voipgw01.budgetphone.nl	In	31737110337@budgetphone.nl	Out	00:02		13.26	13.71	Ok (200)	GSM
10N	2005-04-14 09:20:51	654271273@voipgw01.budgetphone.nl	In	31135802460@budgetphone.nl		02:02		712.97	706.95	Ok (200)	G729
11N	2005-04-14 09:20:28	31717110310@budgetphone.nl	In	+31713315247 (Nederland 31)	Out	00:14	0.0235	243.82	102.07	Ok (200)	G711u

SIP packet trace

```
a=rtpmap:31 H261/90000
```

Incoming packet 2/9: received from 82.92.39.72:13186 at 2005-04-09 12:35:46

```
INVITE sip:248@ag-projects.com SIP/2.0
Via: SIP/2.0/UDP 10.0.0.161:5060;branch=z9hG4bK4006693386
From: <sip:3268@ag-projects.com>;tag=1265054047
To: <sip:248@ag-projects.com>
Call-ID: 1450651508@10.0.0.161
CSeq: 21 INVITE
Contact: <sip:3268@10.0.0.161>
Authorization: Digest username="3268", realm="ag-projects.com", nonce="4257b0a6c918f2574d639beb0ddf324fa2
Proxy-Authorization: Digest username="3268", realm="ag-projects.com", nonce="4257b12eb9d40ac0bd266ca029e2
max-forwards: 69
user-agent: oSIP/Linphone-1.6.7-IMVP
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO
Content-Type: application/sdp
Content-Length: 352

v=0
o=3268 123456 654321 IN IP4 10.0.0.161
s=A conversation
c=IN IP4 10.0.0.161
t=0 0
m=audio 7078 RTP/AVP 18 0 8 101
a=rtpmap:18 g729/8000/1
a=rtpmap:0 pcmu/8000/1
a=rtpmap:8 pcma/8000/1
a=rtpmap:101 telephone-event/8000/1
m=video 7080 RTP/AVP 34 96 31
b=AS:192
a=rtpmap:34 H263/90000
a=rtpmap:96 H263-1998/90000
a=rtpmap:31 H261/90000
```

Outgoing packet 3/9: sent by the SIP Proxy at 2005-04-09 12:35:46

SIP Proxy

```
INVITE sip:248@213.84.95.205:5060 SIP/2.0
Record-Route: <sip:248@81.23.228.150;ftag=1265054047;lr=on>
Via: SIP/2.0/UDP 81.23.228.150;branch=z9hG4bK18d5.5212ddb6.0
Via: SIP/2.0/UDP 10.0.0.161:5060;rport=13186;received=82.92.39.72;branch=z9hG4bK4006693386
From: <sip:3268@ag-projects.com>;tag=1265054047
To: <sip:248@ag-projects.com>
Call-ID: 1450651508@10.0.0.161
CSeq: 21 INVITE
Contact: <sip:3268@82.92.39.72:13186>
Authorization: Digest username="3268", realm="ag-projects.com", nonce="4257b0a6c918f2574d639beb0ddf324fa2
max-forwards: 69
user-agent: oSIP/Linphone-1.6.7-IMVP
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO
Content-Type: application/sdp
Content-Length: 357

v=0
o=3268 123456 654321 IN IP4 10.0.0.161
s=A conversation
c=IN IP4 81.23.228.144
t=0 0
m=audio 47312 RTP/AVP 18 0 8 101
a=rtpmap:18 g729/8000/1
a=rtpmap:0 pcmu/8000/1
a=rtpmap:8 pcma/8000/1
a=rtpmap:101 telephone-event/8000/1
m=video 47314 RTP/AVP 34 96 31
b=AS:192
a=rtpmap:34 H263/90000
a=rtpmap:96 H263-1998/90000
a=rtpmap:31 H261/90000
```

CDR fields

CDR mediation and accounting system provides access to all information that makes SIP sessions[4]. The combined CDR contain information from the SIP signaling, the media streams and rating information. The CDRs are stored in a central database table with the following format:

Field	Type
RadAcctId	bigint(21)
AcctSessionId	varchar(255)
AcctUniqueId	varchar(255)
UserName	varchar(64)
Realm	varchar(64)
NASIPAddress	varchar(15)
CiscoNASPort	varchar(255)
NASPortId	varchar(50)
NASPortType	varchar(255)
AcctStartTime	datetime
AcctStopTime	datetime
AcctSessionTime	int(12)
AcctAuthentic	varchar(32)
ConnectInfo_start	varchar(32)
ConnectInfo_stop	varchar(32)
AcctInputOctets	bigint(12)
AcctOutputOctets	bigint(12)
CalledStationId	varchar(50)
CallingStationId	varchar(50)
AcctTerminateCause	varchar(32)
ServiceType	varchar(32)
FramedProtocol	varchar(32)
FramedIPAddress	varchar(15)
AcctStartDelay	int(12)
AcctStopDelay	int(12)
SipMethod	varchar(50)
SipResponseCode	smallint(5)
SipToTag	varchar(128)
SipFromTag	varchar(128)
SipTranslatedRequestURI	varchar(255)
SipUserAgents	varchar(255)
SipApplicationType	varchar(255)
SipCodecs	varchar(255)
SipRPID	varchar(255)
SipRPIDHeader	varchar(255)
SourceIP	varchar(255)
SourcePort	varchar(255)
CanonicalURI	varchar(255)
DelayTime	varchar(5)
Timestamp	bigint(20)
DestinationId	varchar(15)
Rate	text
Price	varchar(255)
Normalized	enum('0','1')
BillingId	varchar(255)

[4] Multimedia Service Platforms generates one CDR per session

RTP media relay

Media Proxy is a far-end NAT traversal solution for media streams. By using Traversal Using Relay NAT technique (TURN), MediaProxy acts as media relay allowing audio and video RTP media streams to bypass NAT routers without any setting at customer premises. MediaProxy can distribute calls to multiple servers to achieve fail-over and load balancing.

It assumes no knowledge about NAT environment for the SIP clients, it does not required additional STUN servers and it works with any type of NAT[5] or SIP client[6].

When installed at the border of enterprise customers it maintain the shortest path for corporate users when calls are either meant for inside the enterprise or for outside to Operator network.

MediaProxy supports multiple streams per session and on-hold for established sessions.

MediaProxy has an important accounting feature. Combined with CDRTool it helps to perform accurate accounting for SIP calls regardless of how calls have ended[7].

Monitoring

MediaProxy has a web interface for monitoring SIP sessions with visual representation of call status, user agents and codec types in use.

				Media Streams									
	Call	Phones		Caller address	Called address	Via address	Status	Codec	Type	Duration	Bytes Caller	Bytes Called	Bytes Relayed
1	**From:** 9762742@budgetphone.nl **To:** 8707010@budgetphone.nl			10.0.0.139:10036	10.0.0.139:10038	81.23.228.139:35902	active	G711u	Audio	0'14"	133.98k	133.79k	267.58k
2	**From:** 1438921@eurovoice.ro **To:** 9975129@ag-projects.com			10.0.0.139:10028	10.0.0.139:10030	81.23.228.139:35898	active	G711u	Audio	0'14"	133.98k	133.79k	267.58k
3	**From:** 9498818@ag-projects.com **To:** 5086459@ag-projects.com			10.0.0.139:10012	10.0.0.139:10014	81.23.228.139:35890	active	G711u	Audio	0'14"	133.98k	133.98k	267.77k
4	**From:** 5831731@budgetphone.nl **To:** 4167291@eurovoice.ro			10.0.0.139:10000	10.0.0.139:10002	81.23.228.139:35884	active	G711u	Audio	0'14"	133.98k	133.98k	267.77k
5	**From:** 4594279@ag-projects.com **To:** 1268770@ag-projects.com			10.0.0.139:10032	10.0.0.139:10034	81.23.228.139:35900	active	G711u	Audio	0'14"	133.98k	133.79k	267.58k
6	**From:** 8026847@sip2go.com **To:** 657957@ag-projects.com			10.0.0.139:10016	10.0.0.139:10018	81.23.228.139:35892	active	G711u	Audio	0'14"	133.98k	133.98k	267.77k
7	**From:** 907817@budgetphone.nl **To:** 9297287@eurovoice.ro			10.0.0.139:10004	10.0.0.139:10006	81.23.228.139:35886	active	G711u	Audio	0'14"	133.98k	133.98k	267.77k
8	**From:** 1565648@budgetphone.nl **To:** 6352643@sip2go.com		*	10.0.0.139:10020	10.0.0.139:10022	81.23.228.139:35894	active	G711u	Audio	0'14"	133.98k	133.98k	267.77k
9	**From:** 8297044@budgetphone.nl **To:** 5250148@sip2go.com			10.0.0.139:10008	10.0.0.139:10010	81.23.228.139:35888	active	G711u	Audio	0'14"	133.98k	133.98k	267.77k
10	**From:** 7764374@ag-projects.com **To:** 2573937@budgetphone.nl			10.0.0.139:10024	10.0.0.139:10026	81.23.228.139:35896	active	G711u	Audio	0'14"	133.98k	133.79k	267.58k
11	**From:** 8541841@budgetphone.nl **To:** 3686514@budgetphone.nl			10.0.0.139:10040	10.0.0.139:10042	81.23.228.139:35904	active	G711u	Audio	0'14"	133.98k	133.79k	267.58k
12	**From:** 208005299@ag-projects.com **To:** 326@ag-projects.com			213.84.95.205:17006	193.230.183.51:8002	81.23.228.139:35832	active	G729	Audio	3h09'08"	32.44M	32.44M	64.87M

	Server	Caller traffic	Called traffic	Relayed traffic	Sessions	Status
1	81.23.228.139	882.81kbps	882.81kbps	1.72Mbps	12	Ok
2	81.23.228.140					Ok
3	80.84.236.194					Ok
4	80.247.197.195					Ok
5	80.247.197.196					Ok

[5] Excluding default deny-all policy firewalls
[6] SIP clients that support symmetric signaling and media
[7] Missing BYE messages

Call control engine

The call control module enhances the standard capability of the OpenSIPS with rich class-5 telephony feature set, prepaid functionality and a flexible routing algorithm with loop detection among lookup of any combination of ENUM, Quick dials, SIP Aliases and Forwarding.

Prepaid functionality

- Terminate SIP sessions in progress based balance
- Multiple simultaneous calls based on common SIP account balance
- Forward call to voice prompt when balance is insufficient
- SIP accounts can be switched from prepaid to postpaid mode
- Real time display of ongoing sessions per SIP account
- Session termination and debit balance work when RTP media has time out
- Session termination and debit balance work when no BYE is received from the caller
- Add balance and Debit history available as SOAP/XML functions
- Provisioning and monitoring from CDRTool web interface
- Prepaid card generator with prepaid vouchers (card number with pincode)

Call forwarding

- Forward or Redirect calls Unconditional
- Forward or Redirect calls on Busy
- Forward or Redirect calls on No-Answer
- Forward or Redirect calls on Not-Online
- Forward or Redirect calls on Un-Available

Forwarding can be chained with automatic loop detection to several destination types:

- PSTN number
- SIP URI
- ENUM number
- Voicemail account

Operator has exclusive control over:

- Access to PSTN and network assigned Caller-Id
- Network assigned caller id (Remote-Party-ID with Privacy parameters for PSTN gateways)
- Voicemail account creation and assignment

Forwarding is realized based on Operator policy:

- SIP Redirect 302 Moved Temporarily
- SIP Proxy to destination – The called party covers the cost of the call, accounting records will have the Forwarder set as billing party

Capacity control

- Maximum call duration
- Maximum number of calls per user
- Limitation of call origin by IP networks

Call barring

Destinations can be blocked per users basis; call barring provides parental control and control abuse of SIP phone line (e.g. deny calling 0900 numbers).

Privacy

Hides display name in From and Contact SIP headers for IP to IP calls and set of call screening and CLI presentations on ISDN Q931 by using Remote-Party-Id header for calls via PSTN gateways.

Selective accept/reject

With the address-book integrated in the SIP Proxy, contacts can be categorized in VIP, Friends, Business or Family groups and the SIP account may allow incoming calls based on selected groups.

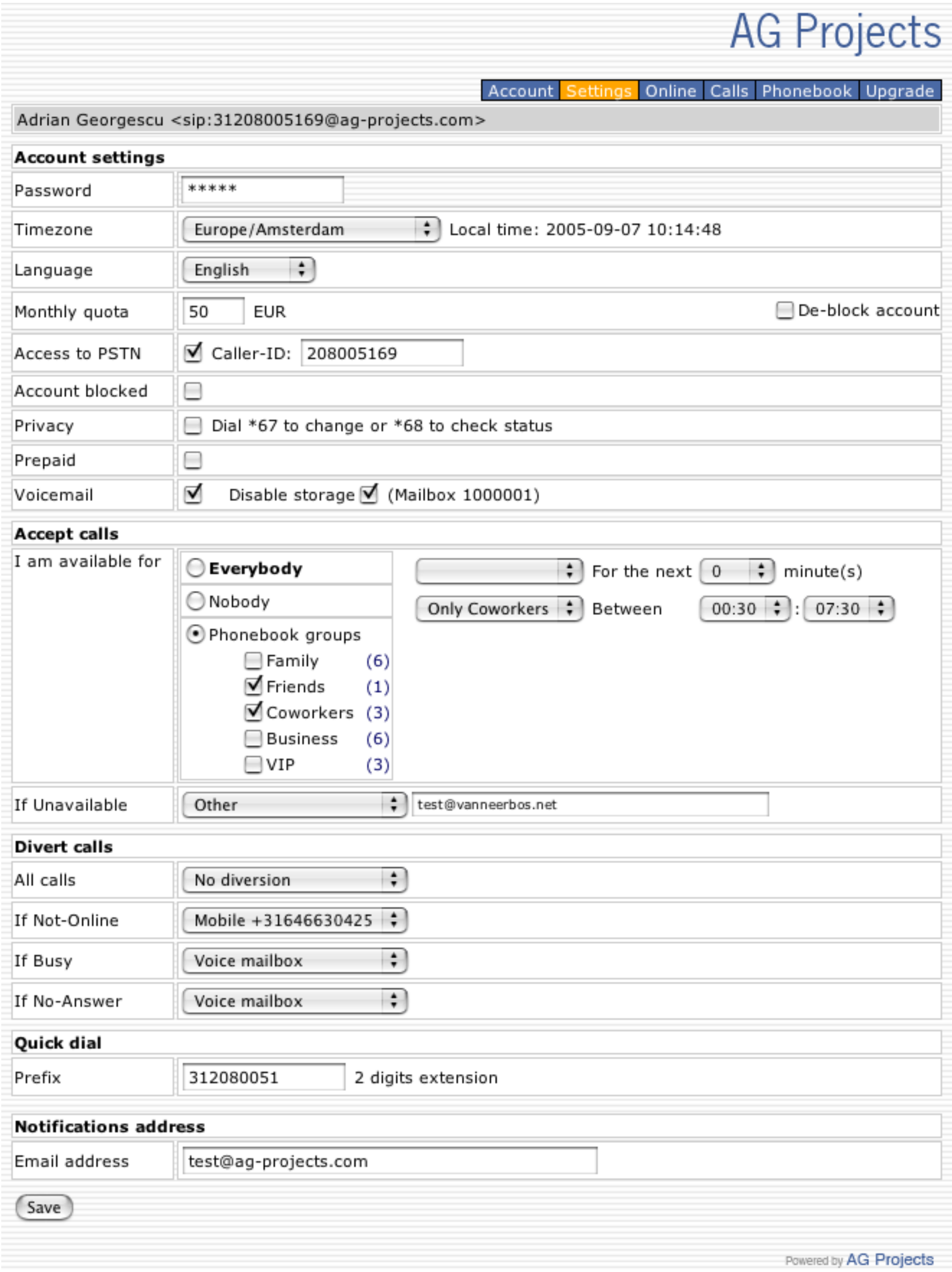
Above features can be combined to achieve Selective call forward, Selective call rejection and Anti-spam policy.

Time based accept

- Temporary redirection with count-down (e.g. enter meeting)
- Control of accepting calls during night or weekends (per day / interval matrix)
- Redirect of rejected calls to voicemail or other recipient (Forward on Un-Available)

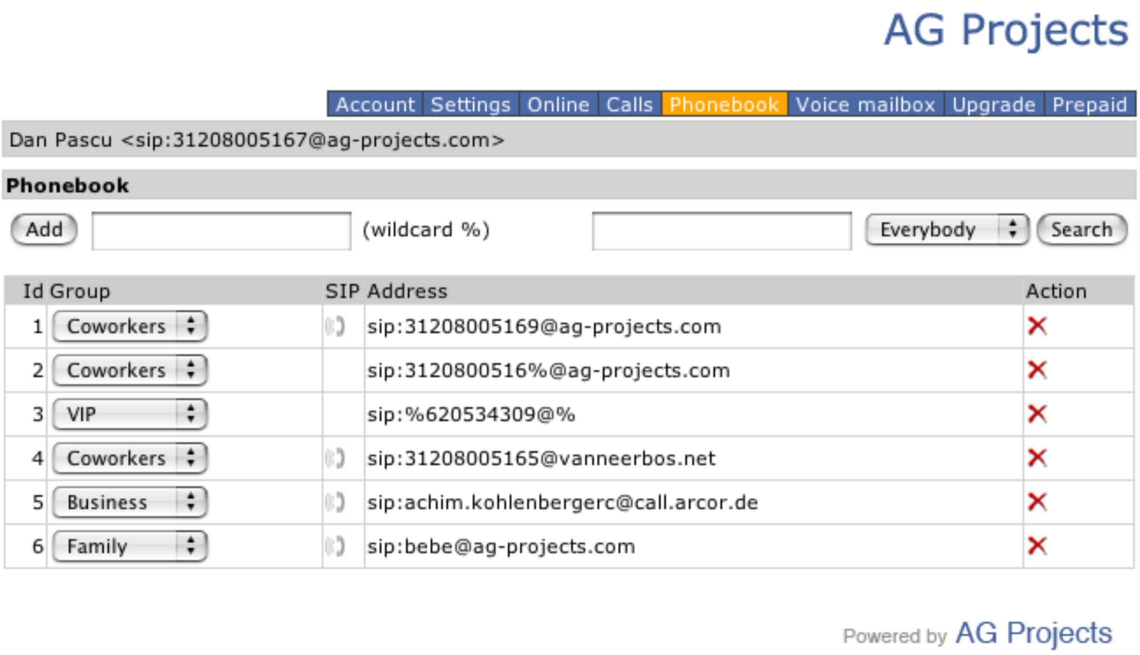
SIP account settings

Using the NGNPro API the Operator can develop customized WEB portals that can be easily upgraded or changed without affecting the platform functionality. Examples[8] of such interfaces:

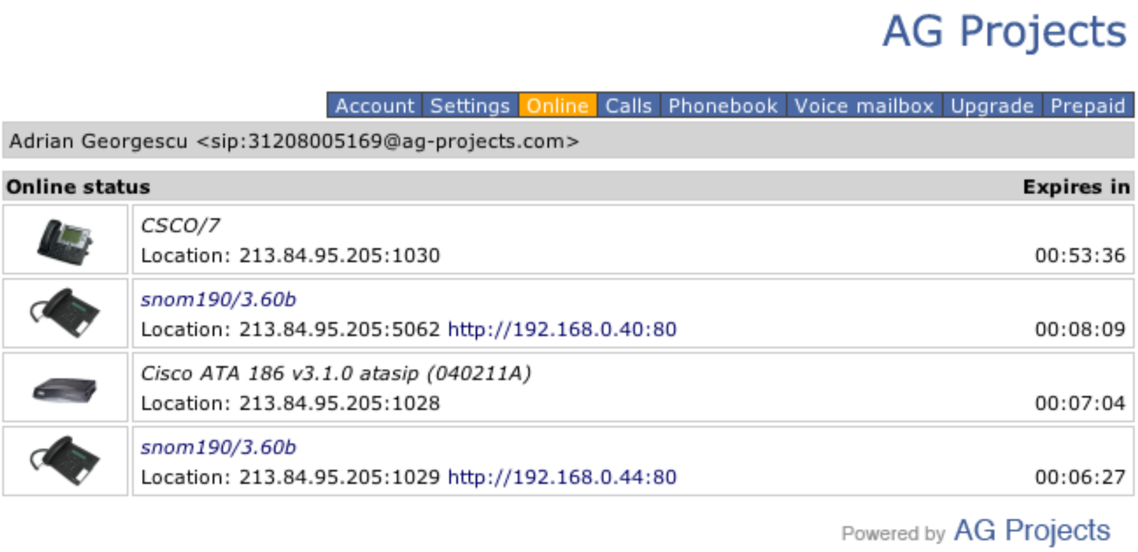


[8] It is the responsibility of the Provider to build any end-user interfaces. AG Projects provides an API for the provisioning functions.

Address book example



Online SIP user agents



Voicemail

The voicemail server is based on Asterisk. The voicemail software has the following features:

- Access from both SIP and PSTN networks
- Voice to E-mail using .wav attachments
- DTMF access to voicemail account
- Customizable welcome message
- Server storage for voice messages
- Remote provisioning engine using NGNPro
- Rich codec support (G711, G722, G729, GSM, Speex, iLBC)
- Session logging accessible from CDRTool
- Access from both Internet and PSTN
- Message waiting indicator (MWI) using SUBSCRIBE/NOTIFY
- Provisioning API integrated with SIP Registrar and ENUM directory

Provisioning of voicemail accounts is integrated within the SIP proxy account management; it is possible to automate the creation of voicemail boxes together with the SIP accounts by using NGNPro API.

Presence

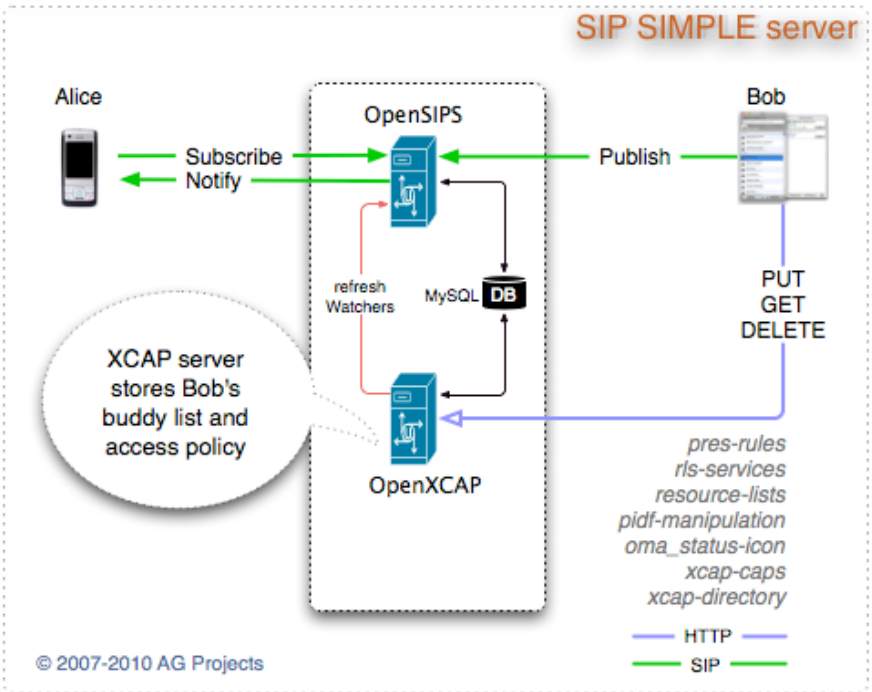
Presence based on SIMPLE IETF specifications can be handled in three configurable modes[9].

- Peer-to-Peer mode leaves the responsibility of notification and subscription to the end points. The platform provides authentication and authorization for SUBSCRIBE requests from User Agents and proxies the requests to the intended recipients.

- Proxy-mode allows server based presence by providing a bridge between User Agents and a Presence server located behind Multimedia Service Platform. In this mode the platform provides authentication and authorization for SUBSCRIBE/PUBLISH requests from User Agents and proxies the requests to the Presence server. Any SIMPLE compatible server can be used for this purpose.

- Native mode (built-in Presence Agent) allows the platform to behave like a presence agent that handle PUBLISH and NOTIFY with a centrally managed access policy and buddy lists (XCAP protocol) based on the following standards:

  - RFC 3265 (ietf-simple-event-notification)
  - RFC 3856 (ietf-simple-presence)
  - RFC 3903 (ietf-simple-publish)
  - RFC 4479 (ietf-simple-data-model)
  - RFC 4480 (ietf-simple-rpid)
  - RFC 4662 (ietf-simple-event-list)
  - RFC 4825 (ietf-simple-xcap)
  - RFC 4827 (ietf-simple-xcap-pidf-manipulation-usage)



[9] Only one mode can be active. The default is Native mode.

Presence policy

XCAP protocol allows a client to read, write, and modify application configuration data stored in XML format on a server. XCAP maps XML document sub-trees and element attributes to HTTP URIs, so that these components can be directly accessed by HTTP. An XCAP server is used by the XCAP clients to store data like Presence policy in combination with a SIP server that supports PUBLISH/SUBSCRIBE/NOTIFY methods to provide a complete SIP SIMPLE server solution.

The XCAP server is used as a policy decision maker by the SIP Presence server to manage the subscriptions and notifications of presence information based on end-user controlled preferences. An immediate advantage of using an XCAP server is to allow end-users to reliably syncronize presence configuration data like buddy lists and access lists between multiple SIP SIMPLE devices. The XCAP server can also be used to publish to SIP devices information retrieved from external sources and using different APIs acting like a bridge between real-time communications based on SIP protocol and other web technologies like SOAP/XML.

Example of Presence/XCAP enabled user agent Blink.

Interactive messaging

A series of related instant messages between two or more parties can be viewed as part of a "message session", that is, a conversational exchange of messages with a definite beginning and end. This is in contrast to individual messages each sent independently. Messaging schemes that track only individual messages can be described as "page-mode" messaging, whereas messaging that is part of a "session" with a definite start and end is called "session-mode" messaging.

Page-mode messaging is enabled in SIP via the SIP MESSAGE method, as defined in RFC 3428. Session-mode messaging has a number of benefits over page-mode messaging, however, such as explicit rendezvous, tighter integration with other media-types, direct client-to-client operation, and brokered privacy and security.

Message Session Relay Protocol (MSRP) is a protocol for transmitting a series of related instant messages in the context of a session. Message sessions are treated like any other media stream when set up via a rendezvous or session creation protocol such as the Session Initiation Protocol (SIP). MSRP Sessions are defined in RFC 4975.

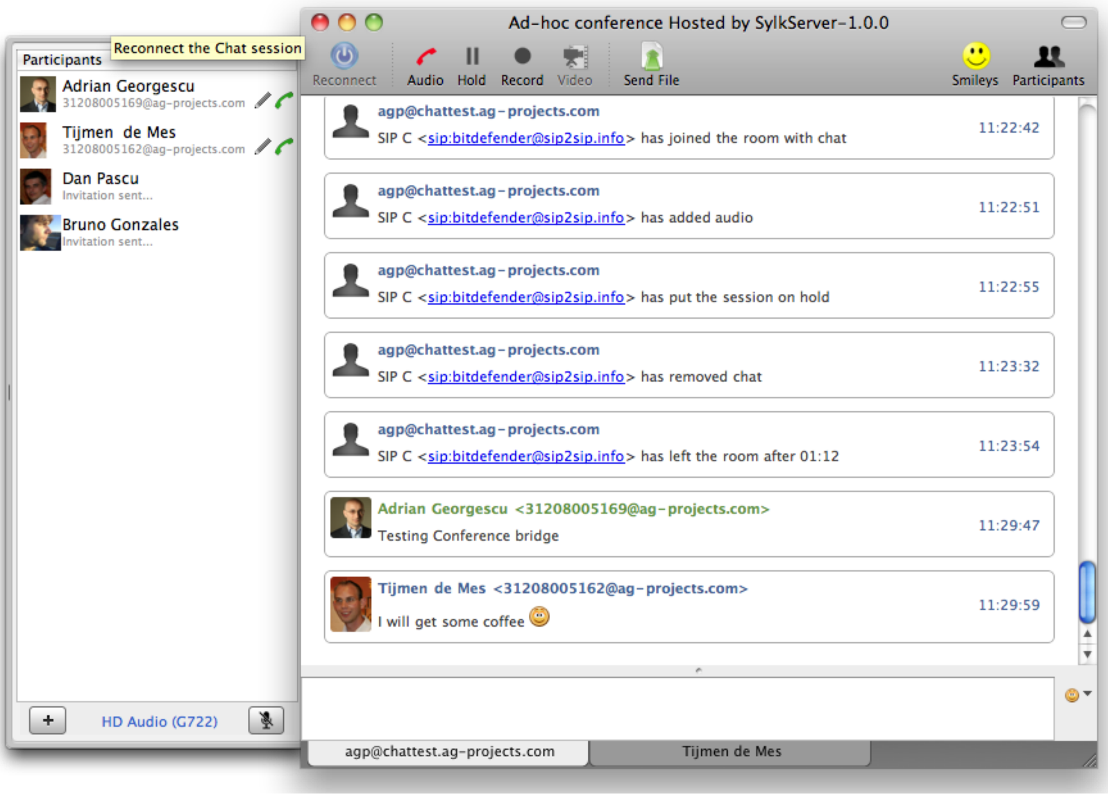
Interactive messaging is supported by MSP in both modes:

1. Paging mode using SIP MESSAGE method
2. Session mode using MSRP protocol.

IM/File transfer relay

For NAT traversal purposes, an MSRP relay component is provided.

- TLS encryption and digital certificates using GnuTLS library
- Digest or basic HTTP authentication with support for multiple realms
- Multiple relays can be used for load balancing purposes



ENUM routing engine

ENUM solution included with Multimedia Service Platform allows the routing of ITU E.164 telephone numbers to IP addressing schemes (like SIP addresses, IM, MMS and e-mail) to be managed in one place and outside of the switching equipment. ENUM can also be used to port numbers in and out operator network. Multimedia Service Platform is ENUM ready on both client and server side. SIP Proxy is able to perform ENUM queries and the built-in DNS ENUM directory service allows the provisioning of E164 telephone numbers and sharing of the directory with other parties.

AG Projects has a rich experience with development and deployment of ENUM systems; it participated to ENUM trials in several countries, participated to the ENUM plug test organized by ETSI and it is the first company to have deployed ENUM in commercial environment during 2004.

Multimedia Service Platform is able to perform ENUM queries in behalf of its end-points like PSTN gateways, SIP User Agents or IP PBX as long as they are SIP enabled. It is not necessary to have the end-points upgraded in order to support ENUM and interconnect with other SIP/ENUM enabled parties.

The system complies with the following specifications and recommendations:

- RFC 3401 Dynamic Delegation Discovery System (DDDS) Part One: The Comprehensive DDDS
- RFC 3402 Dynamic Delegation Discovery System (DDDS) Part Two: The Algorithm
- RFC 3403 Dynamic Delegation Discovery System (DDDS) Part Three: The Domain Name System (DNS Database)
- RFC 3404 Yes DNS Dynamic Delegation Discovery System (DDDS) Part Four: The Uniform Resource Identifiers
- ETSI TR 102 055 V1.1.1 - Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); ENUM scenarios for user and infrastructure ENUM)
- ETSI TS 102 172 V1.2.1 - Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Minimum requirements for interoperability of ENUM implementations

ENUM server features

- NAPTR syntax and logic checks
- Manipulate records in both DNS and E.164 format
- Delegation of records to end-users
- Multiple site setup with automatic replication
- Local number portability
- Fast changes available in real time to the SIP Proxy or other ENUM enabled clients
- Multiple TLD support (private trees and public tree)
- SOAP/XML provisioning API integrated with SIP Registrar accounts
- Multiple TLD support (private trees and public tree)

Supported NAPTR service types

Service	URI schemes	Normative references
E2U+sip	sip:,sips:	RFC 3764
E2U+h323	h323:	RFC 3762
E2U+voice:tel	tel:	RFC 4415
E2U+email:mailto	mailto:	RFC 4355
E2U+fax:tel	tel:	RFC 4355
E2U+mms:tel	tel:	RFC 4355
E2U+sms:tel	tel:	RFC 4355
E2U+ems:tel	tel:	RFC 4355
E2U+mms:mailto	mailto:	RFC 4355
E2U+sms:mailto	mailto:	RFC 4355
E2U+ems:mailto	mailto:	RFC 4355
E2U+web:http	http:	RFC 4002
E2U+web:https	https:	RFC 4002
E2U+ft:ftp	ftp:	RFC 4002
E2U+pres	pres:,sip:	RFC 3953
E2U+ifax:mailto	mailto:	RFC 4143
E2U+pstn:tel	tel:	RFC 4769
E2U+pstn:sip	sip:	RFC 4769
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E2U+void:http	http:	ID
E2U+void:https	https:	ID
E2U+iax	iax:	ID
E2U+iax2	iax2:	ID
E2U+im	im:	ID
E2U+loc:http	http:	ID
E2U+loc:https	https:	ID

AG Projects ENUM implementation is compliant with ETSI recommendations for ENUM interoperability. For more information about ENUM interoperability see:

http://www.etsi.org/plugtests/History/2005ENUM.htm

For more information about AG Projects ENUM activities visit:

http://ag-projects.com/ENUM

SOAP/XML Provisioning

NGNPro is an integrated remote provisioning engine for all platform components (SIP Proxy, Voicemail server, Call control, User rights, ENUM mappings) and offers a central point of control to the platform. The design of the SOAP control interface allows one to 1) provisioning and 2) obtain usage information to/from the platform components (SIP Proxy/Registrar, Voicemail, ENUM, and Accounting) by the use of RPC (Remote Procedure Calls).

NGNPro is an Application Programming Interface (API) based on SOAP/XML.

Architecture

NGNPro is an intermediate abstraction layer between the existing provisioning system of the enterprise deploying VoIP and the data stored in the platform components. This means that the Operator provisioning system or front-end does not need to know the low level details of how the user data is stored in the database and only concentrate on a few operations it needs to perform to manage the users, operations made available using SOAP/XML.

NGNPro provides a clear delimitation between the Operator infrastructure (gateways, web front-ends, billing or customer database) and Multimedia Service Platform. It is the responsibility of the Operator to build any end-user front-ends (like web portals) based on the supplied SOAP/XML interface.

To build a SOAP/XML client for NGNPro, a SOAP/XML description file (WSDL) is provided.

NGNPro web service description language (WSDL) is available at:

https://mdns.sipthor.net/ngnpro/wsdl

Provisioning functions

The provisioning part of the SOAP interface allows management SIP users along with all the auxiliary settings that accounts have (like group membership, voicemail accounts, SIP aliases, PSTN access, ENUM mappings, call diversions, etc). The SOAP provisioning interface provides the functions that operate on each individual component of the platform. In addition to this, there can be higher-level wrapper functions defined that can execute multiple such basic operations in a single request (like creating a SIP account together with a voicemail account and an ENUM mapping, as well as placing the user in certain groups that give him certain rights).

These wrapper functions allow not only the SOAP client to execute fewer requests to manage an account, but also guarantee the atomicity of the operation and the fact that the database will not end up in an inconsistent state. This is because the SOAP server can use commit/rollback techniques that are not available to the client when it executes the requests independently.

Obtaining usage information

The querying functions provided by the SOAP/XML interface allow one to request information about existing SIP accounts, like SIP account profile, voicemail info and call forwarding settings. It also allows querying for calls related to a user (missed, placed or received), given various conditions, like specifying a time interval when they took place.

API functions

1. SIP proxy domain management (adding, removing and querying domains)
2. SIP account management (adding, updating, deleting and getting a SIP account)
3. SIP aliases (adding, deleting and getting SIP aliases for a given SIP account)
4. Group membership management (granting, revoking or listing group membership)
5. Voicemail account (adding, updating, removing or getting a voicemail account)
6. Call diversions (setting or getting the diversions for a given user)
7. Retrieving call information for a certain user (missed, placed, received calls)
8. Retrieving the registered phones one user has (getting the user locations info)
9. ENUM management (adding, removing and getting the ENUM mappings)
10. SIP account prepaid balance manipulation and history

Web portals

The Operator is responsible to build its web portals to interface with Multimedia Service Platform. Using NGNPro interface the Operator can build new front-ends or adapt existing CRM, provisioning or support systems to the Multimedia Service Platform, making the integration easy and cost effective.

SOAP/XML is an industry standard for which there are implementations available in almost any programming language (like C, Java, Python, PHP) and can be used by web developers to create customized web portals for the Operator or its resellers without bothering with the complexity of the infrastructure sitting behind.

The development cycle for a web portal based on NGNPro is about 3-12 weeks depending on the portal complexity[10].

[10] Estimation based on previous deployments

Telephony features

Many classic telephony features have a direct equivalent in the SIP world and are available as standard features in the platform. Other features that are not possible with classic telephony but possible with SIP protocol are available (like IM, presence, parallel forking to multiple devices).

Platform built-in features
Call Rejection (DND)
Call Barring (Blocked destinations per caller)
Call Forward/Redirect (Unconditional, Busy, No-Answer, Not-Online, Not-Available)
Caller ID Display (CLIP)
Caller ID presentation
Permanent Caller ID Restriction (CLIR)
Call-By-Call Caller ID Restriction
Selective Call Accept (based on caller groups)
Selective Call Rejection (based on caller groups)
Voice Mail and Voice to Email (via e-mail or IVR)
Anonymous call handling
Quick dial (dial short numbers for team-working)
Call forwarding settings using DTMF
Configurable Timeout on no-answer (in seconds)
Emergency routing to the closest emergency access point in the region
Fraud prevention (daily monthly quota)
Time based call accept (do not disturb after midnight or week-end)
Multiple identities (SIP aliases e.g. john@example.com)
Fax support based on G711 pass-through[11]
Parallel ringing to multiple phones
Support for both telephone numbers and SIP URIs
MWI (Message waiting indicator)
Prepaid or postpaid billing
Call limit (total number, per IP restrictions)

SIP User Agent features
Caller waiting
Call Hold
Call redial
Call return
3-way calling
PBX media processing (Call queues, auto-attendant, IVR)
Playing of voice announcements in the middle of the call setup

SIP Applications supported by the platform
Presence based on SIP/SIMPLE
Voicemail and voice to email
IM based on SIP MESSAGE (paging mode) and MSRP (session mode)

Known limitations

[11] T38 fax support with NAT traversal is work in progress

Having a SIP proxy at the core, the platform does not act as a media end-point. As a result of a SIP Proxy design the platform cannot play voice prompts announcements in the middle of a call setup, which was a feature often available with classical telephone systems.

While this missing feature may seem a weakness when compared to a classical voice switch it actually helps the platform be much more scalable[12] then other telephony solutions and maintain a lower cost (because there is no media processing, no coding/decoding and no codec licenses involved). Voice prompts can still be played as a last resort by using the Asterisk voice mail server after a session setup has failed (for example when a number is incorrectly formatted or the prepaid balance has been exceeded).

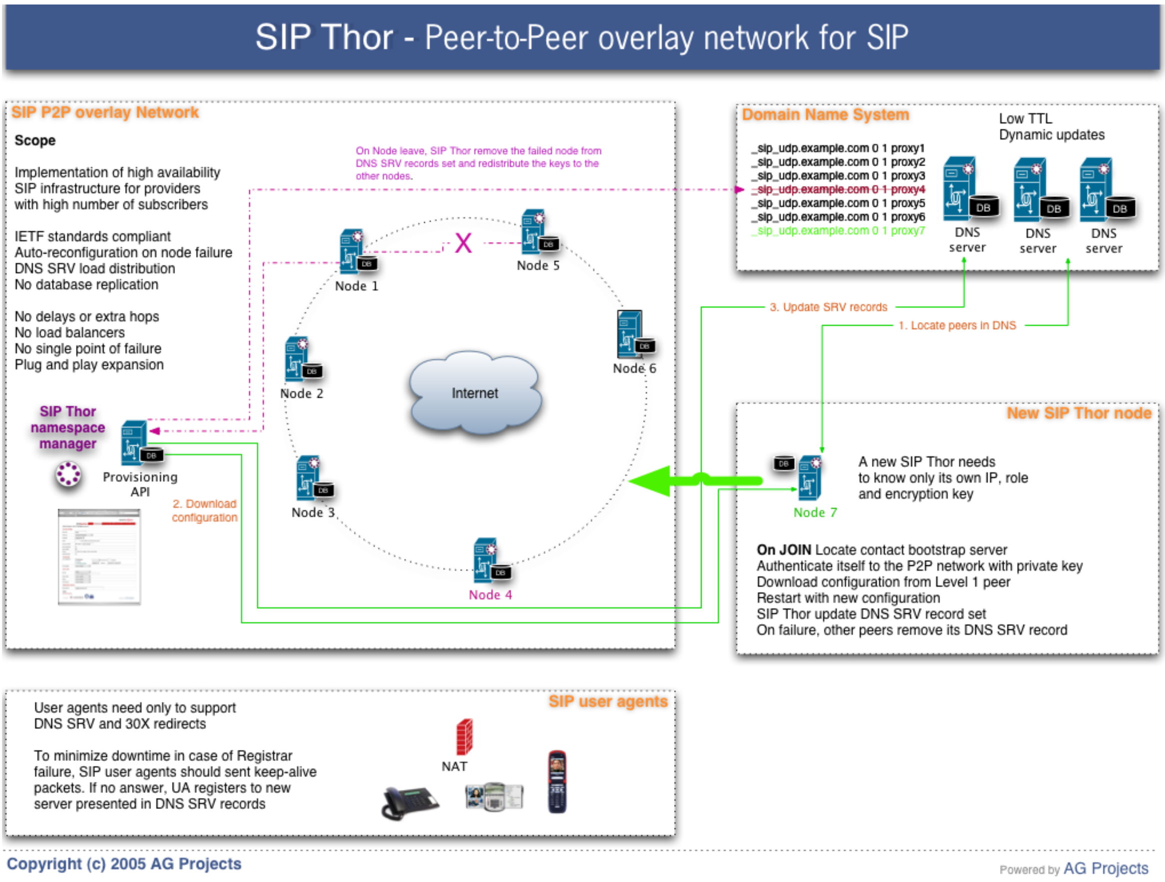
Some telephony features that deal with media processing do not have a SIP Proxy correspondent[13] and are not built in the platform. Applications not built in the platform itself are PBX media plane features (like IVR, N-party conferencing, ACD, call queues, call parking). These applications can be added as external application servers to which the SIP Proxy routes the calls accordingly.

[12] About two orders of magnitude higher per server that a PBX or voice switch
[13] Because a SIP Proxy does not keep state of sessions and does not interact with media streams

Scalability and high availability

For scalability and high availability beyond a single node, Multimedia Service Platform uses SIP Thor. Using SIP Thor, multiple servers can act like one big system by sharing the load and recovering from server or network failure without human intervention.

SIP Thor is a Peer-to-Peer overlay that allows for load balancing, self-organization and geographical distribution of SIP infrastructure elements. The Peer-to-Peer overlay provides Multimedia Service Platform with the primitives for self-organization, high availability and scalability.



Self-organizing technology

SIP Thor is based on P2PSIP, a set of technologies that combines exiting IETF standards like SIP, DNS and ENUM with Peer-To-Peer techniques like distributed hash tables (DHT). This combination allows scalability unmatched by traditional clustering or load balancer models while dramatically reducing the operational costs. SIP Thor implements an overlay network for SIP Proxy/Registrar and other network centric functions like DNS, presence agent, database storage, voicemail, xcap policy and provisioning.

The P2P layer provides the primitives for self-organization, routing and resource lookups. The self-organization property brings the operational costs to minimum by eliminating the need for monitoring and maintenance activities, which traditional infrastructures require.

High-availability

SIP Thor is designed to automatically recover from disasters like network connectivity loss, server failures or denial of service attacks. On node failure, all requests handled by the faulty node are automatically distributed to surviving nodes without any human intervention. When the failed node becomes available, it takes back its place in the network without any manual interaction.

Scalability

Traditional technology duplicates functionality and wastes resources by using expensive and inefficient hot/stand-by clusters. SIP Thor uses all available servers in active mode to their maximum capacity. With SIP Thor, there is no need for expensive hardware or stand-by machines to achieve high-availability distributed among different geographic sites. Capacity can be increased by simply adding extra nodes. Just by installing a standard software package, a new node is able to take its place in the network without any manual configuration.

Security

Authentication and authorization

The trust relationship between SIP accounts and SIP Proxy is based on DIGEST algorithm; both have a database with shared credentials.

The trust relation ship between XCAP clients and XCAP server is alos based on the same DIGEST algorithm. The same SIP credentials can be used for XCAP authorization.

The SIP Proxy challenges all incoming SIP requests that generate new sessions (authentication). Once authenticated, the session is established depending on the rights associated with the SIP account (authorization).

For interconnection with the PSTN, a transparent SIP trunking service from the SIP Proxy to a SIP gateway provider must be setup. The authorization of SIP requests is based on transitive trust. The SIP Proxy has a trust relationship with each SIP account and the SIP gateway has a trust relation with the SIP Proxy. The trust relation between the SIP Proxy and the SIP gateway can be realized either by IP addresses or by the use of TLS protocol combined with digital certificates signed by the same Certificate Authority. The SIP gateway may not use DIGEST authentication in the relation with the SIP Proxy because it does not have access to the SIP accounts database of the SIP Proxy.

Denial of Service Attacks (DOS)

The SIP Proxy code is mature and is designed to run on the public Internet like any web or e-mail server. A denial of service attack consisting of a packet rate higher than its handling capacity might render services to legitimate users unusable. While the SIP Proxy has mechanisms for throttling, traffic shaping and access lists based on IP addresses it is recommended that denial of service attacks are detected and blocked at layer 3 (IP) in the border router in front of the SIP Proxy.

There is a common misunderstanding that session border controllers can help against DOS attacks. They do not, a session border controller positioned in front of the SIP Proxy does not help mitigate this problem; it only moves the problem one hop further and also brings all the disadvantages of the session controllers (interoperability problems, costs, capacity limitations, lack of support for applications beyond voice like video, Presence and IM).

SIP Thor, the upgrade path for Multimedia Service Platform allows for distribution of SIP Proxy and other resources to multiple data centers, this makes the system much less likely to be affected by DOS or DDOS attacks.

Systems monitoring

All systems can be monitored for availability of networking and application stance by remote monitoring probes installed in different networks plus software running internally on each application server.

Designated network, system or component failures can trigger alarms, which are sent based on a predefined contact list to email or SMS recipients[14].

[14] It is the responsibility of the Operator to setup monitoring and alarming functions

Performance indicators

These indicators are based on previous experiences, may vary depending on hardware, network or software versions and are by no means guaranteed. The figures where not otherwise indicated are per server unit.

Per node component throughput /capacity:

- SIP Proxy: <2,000 transactions per second (UDP)
- Media relay[15]: ~2000 simultaneous streams (G711 RTP+RTCP)
- SOAP/XML Provisioning: ~20 requests/second (TLS)
- XCAP: ~20 requests/second (TLS)
- Rating engine: ~200 CDRs per second
- DNS response time: < 10ms
- SIP Registrar: 30,000 accounts/node
- Address book: 100 contacts per SIP account
- Accept rules: 20 rules per SIP account
- Reject rules: 20 rules per SIP account
- Barring rules: 20 rules per SIP account
- Voicemail server: ~20 calls per second
- Voicemail capacity: 100,000 mailboxes/node
- Voicemail messages: max 99 messages per mailbox
- Custom announcements: 1 message per mailbox
- ENUM resolution speed: ~5,000 queries/second (UDP)
- ENUM records: 3,000,000 NAPTR mappings
- Call Detail Records: MySQL dependent
- SIP tracing: MySQL dependent

Platform throughput

- 20 successful call attempts/second[16] per node with all features enabled (e.g. ENUM and DNS lookups, Call control, all DNS lookups, Media relay selection, Authentication, Authorization, and Accounting).

Notes

- All figures exhibited in this document are based on previous experiences with particular call flows, software, and hardware and network combinations. It is not guaranteed that the same figures can be reproduced.

- Storage capacity may greatly differ than the indicated figures. No guarantee can be given upon the overall platform performance should any particular component operational limit be exceeded.

- No guarantee can be given to fix any scalability issue caused by the exceeding of a particular component operational limit.

- The hardware used for testing is based on servers featuring two Intel Xeon 3GHZ CPU, 2 GB Memory, 36 GB SCSI discs and dual 100 GB Ethernet card.

[15] MediaProxy 2.0
[16] In 60 seconds there are 2400 active sessions in progress

Resilience

The Operator at multiple levels can achieve resilience:

Facility. Operator is responsible for contracting suitable hosting space including power in dual configuration protected by always-on UPS system with a minimum of 15 minutes run-time and backed-up by diesel-generators with at most 1-minute start-up time. Redundancy should be required for air conditioning (at least N+1).

Power distribution. All servers should n be fitted with dual-power supply fed by two different RPP/PDU. Power consumption per rack will be dimensioned to be below 1500 W. Servers without static switches must feed dual power supply.

Network level. All network paths can be meshed with layer II switches configured for layer II spanning tree protocol using managed Ethernet switches.

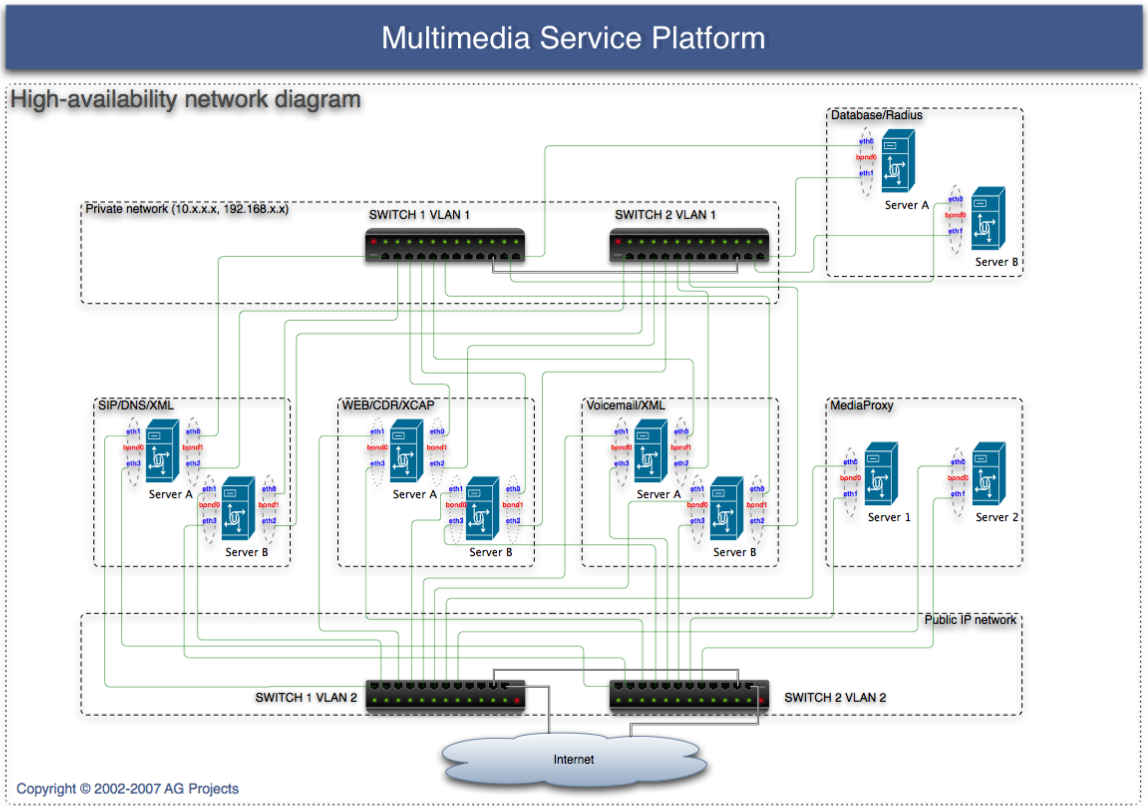
Server level. Application servers run in master/slave cluster mode controlled by a heartbeat mechanism.

Application level. Applications are set to achieve fall-over on system or network failures.

The design of the platform allows testing and upgrade to future software versions without major service disruption. All critical components can be upgraded one by one by setting the active server to the master/slave members of the designated clusters.

The failover occurs automatically when the heartbeat or monitoring servers detects a network or server failure.

High-availability network diagram



Hardware requirements

Production platform

Deployment of a standard Multimedia Service Platform with all components in resilient configuration requires 10 servers and two Ethernet switches with virtual LAN support. The servers have the following hardware specifications:

Profile	Specification	Qty	Example
Profile 1	Dual processor 3GHz 4 GB RAM 4x Ethernet ports RAID 1 Dual power supply	10	Dell PowerEdge 1950

Testing platform

Deployment of a staging Multimedia Service Platform with all components in non-resilient configuration requires 6 servers and one Ethernet switch. The servers have the following hardware specifications:

Profile	Specification	Qty	Example
Profile 1	Dual processor 3GHz 2 GB RAM 2 Ethernet ports	3	Dell PowerEdge SC1435

SIP end-points requirements

Multimedia Service Platform works with any SIPv2 User Agents, providing SIP industry best practices have been used in the development of the end-points (most SIP User Agents do this nowadays).

SIP User Agents requirements

1. Must send and receive signaling and media using the same port, described in RFC 4961
2. Must support DNS SRV record lookup[17] for locating the SIP Registrar and Outbound Proxy based on RFC3263 (Locating SIP Services).
3. Must honor the Time to Live (TTL) of DNS lookups

PSTN gateway requirements

4. Must support SIP extensions for caller id and privacy (P-Asserted-Identity / Remote-Party-Id headers)
5. Must work in active mode (send RTP data as soon as call setup is completed)
6. Must have publicly routable IP address for both signaling and media

[17] It is not possible to build scalable SIP networks without DNS support in the SIP User Agents

Post installation activities

Support activities

AG Projects provides a maintenance contract for all software solutions developed in house. Maintenance activities cover bug fixing of AG Projects software, upgrades to all AG Projects standard builds namely OpenSIPS, MediaProxy, CDRTool, NGNPro, MSRPRelay, OpenXCAP, SylkServer and SIP Thor components and their integration with third party back-end systems like Freeradius and MySQL servers.

The operator is responsible for operating the platform (e.g. provisioning, accounting, first and second level support to its customers, trouble-shooting of network problems, monitoring the servers' health (disc space, system load, CPU usage, process list and memory usage), back-up and restore of platform configuration and data, monitoring and fixing database replication problems).

Support interventions must be requested via AG Projects trouble-ticketing system[18]. For each problem reported by the Operator in the trouble tickets, AG Projects provides in a timely manner answers and send notifications to contact points specified in the ticket.

In order to provide quality support AG Projects requires:

1. Operator must be trained for the products and services provided by AG Projects
2. Operator must open a trouble-ticket in AG Projects customer support system
3. The trouble-ticket must contain the description of the problem and how it can be reproduced
4. Engineers employed by the Operator must have understanding about the platform setup
5. Operator must be able to provide remote access[19] to equipment(s) where the problem occurs
6. Operator must provide remote hands and eyes at the facility when necessary

The standard response time is next business day. Optional, 4 hours response time during business days and weekend coverage on a 24 hours basis can be provided.

Maintenance for third party software

AG Projects is not responsible for supporting software developed by third parties, it does however provide best-effort support and try to help customer proactively to find solutions for such problems. MySQL, Asterisk, Freeradius are examples of software not directly supported by AG Projects.

Support or extra development related for third party software components may be provided if and when possible on a case-by-case basis and will be charged per hour at a rate specified in the pricing list.

Training

Training on location is provided upon request. Per day charges apply.

Extra development

Based on individual project evaluations, AG Projects may further develop features on installed platform. New development can be done only after delivery and acceptance of the standard platform design. Per day or per project charges apply.

[18] http://support.ag-projects.com
[19] SSH root access

Disaster recovery plan

It is the responsibility of the Operator to execute regular back-ups and implement a disaster recovery plan. AG Projects provides during the initial platform rollout information about the configuration files and database that need to be backed-up on a regular basis.

Particular requirements

<>

Deliverables

Most of the work performed by AG Projects is done from remote by installing and configuring software on hardware already purchased and installed by the Operator.

Operator responsibilities

The Operator will perform the following tasks:

1. Purchase the server hardware
2. Procure IP connectivity, telephone numbers and PSTN connectivity
3. Bring the equipment in service including OS installation[20]
4. Provide AG Projects with administrator level access to all machines
5. Have remote hands and eyes available at the facilities
6. Allocate the resources for the development of its web portal
7. Allows AG Projects to access the servers 24/7
8. Allow AG Projects to register and use its own SIP phones on the platform

AG Projects responsibilities

AG Projects will install and configure the software, provide as-built plans and will hand-over this documentation to the Operator once the installation is complete. The as-build documentation is provided as a walk-trough the installation phases logged in the working tickets. Based on the log the Operator may recreate the platform from scratch in case of a disaster.

Operator responsibilities

The Operator may NOT publish, rent, copy, lease, sell, lend or sublicense any CONFIGURATION of the software installed on the operator infrastructure to other parties without prior permission from AG Projects.

The usage license of the platform is revoked if the payments have not been done in full.

Multimedia Service Platform may not be used for any purpose without a support contract with AG Projects. The usage license of the platform is revoked if the operator cancels its support contract.

[20] AG Projects supports only LINUX Debian

Timescale

Providing all prerequisites are met at Day 1:

1. Commercial agreement with AG Projects is in place[21]
2. Operator facilities are ready (power/network/servers, remote hands and eyes) see Operator responsibilities above

AG Projects can deliver based on the following schedule:

Phase 1 Specification of the provisioning API (start phase)

Day 1

AG Projects provides access for the Operator operational team to its support system. The planned work activities for platform installation will be logged via the ticketing system.

AG Projects delivers to the Operator the description file for the SOAP/XML provisioning API. The Operator starts developing the web provisioning front-end by connecting to AG Projects test facilities.

Phase 2 Prototype platform (delivery phase)

Day 1 + 30/45 days

AG Projects plans, installs and tests the individual components and the platform as a whole. AG Projects is notifying in a timely manner the progress via the ticketing system.

AG Projects does the initial provisioning of SIP accounts, numbering and rating plans and configures the SIP Proxy to inter-work with the designated PSTN gateway(s).

AG Projects announces the Operator via the ticketing system the date when the planned works have been completed and the platform is considered delivered.

Phase 3 Production platform (acceptance phase)

Day 1 + 45/60 days

AG Projects announces the Operator via the ticketing system the date when the platform is considered delivered (Delivery date).

AG Projects provides as-built documentation and hands over the platform to the Operator operations department.

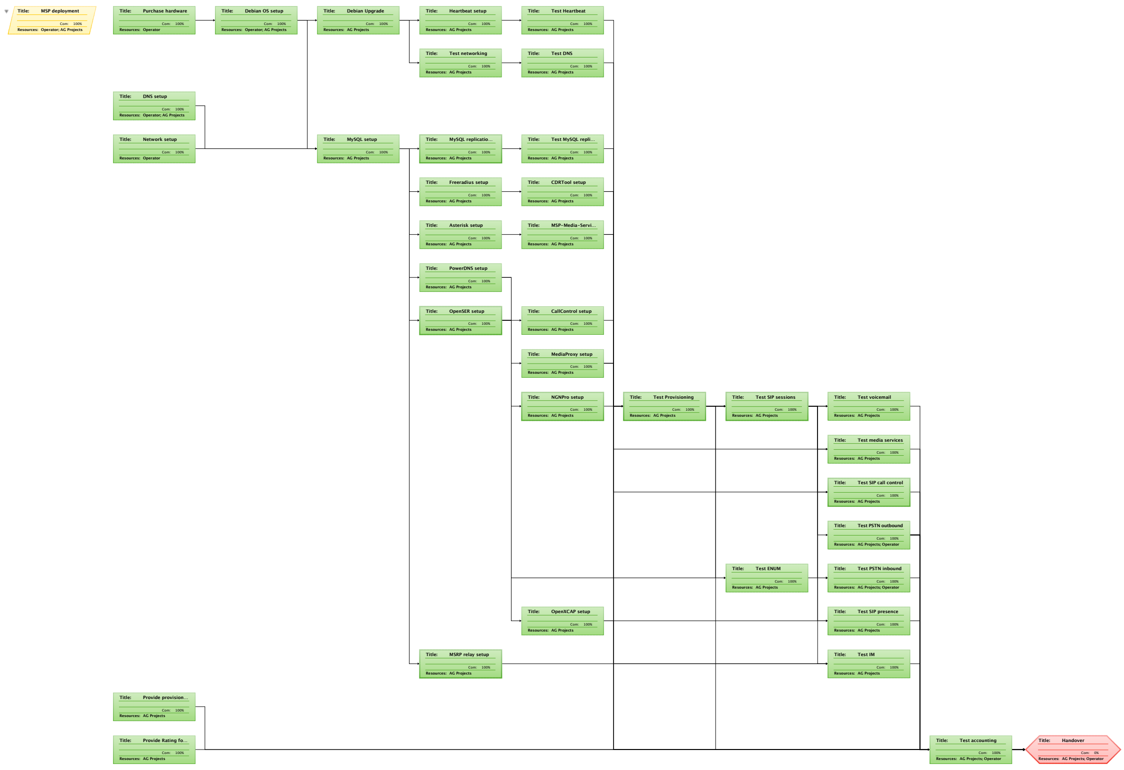
The Operator executes its own tests and provides a snag-list in maximum 15 days after the delivery date. If no snag-list is handed over to AG Projects with 15 days after the delivery date, the platform is considered accepted by the Operator (Acceptance date).

The acceptance date is considered the start date for the post-installation support activities.

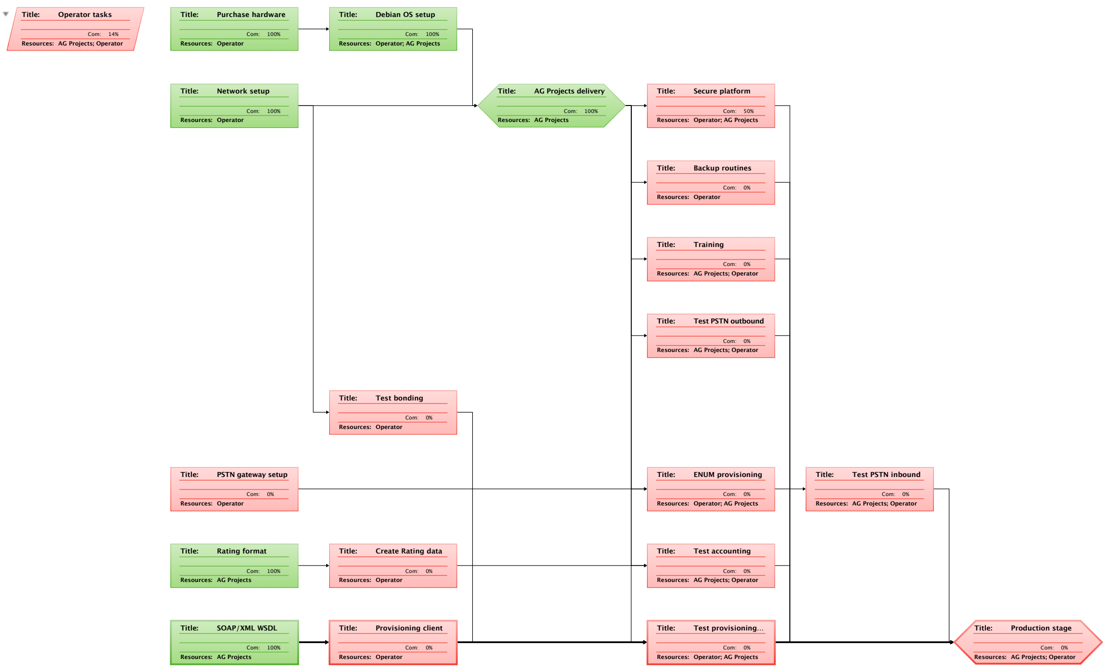
[21] Customer Order Form (COF) is signed

Rollout plan

Below is an outline of AG Projects activities required for putting the platform in service.



Bellow is an outline of the activities that must be performed by the Operator for getting to production stage.



Company information

AG Projects provides since 2002 SIP turnkey solutions for cable companies, Internet service providers and telecom operators. Having a broad experience in Internet technologies, AG Projects is a skilled vendor able to provide and integrate the key infrastructure elements required for providing real-time communications services like VoIP, IM and Presence.

AG Projects applications are following open standards and makes use of components for which there is a large knowledge base available. This guarantees the right track for new developments and a rapid disaster recovery should such event arises.

For more information:
www.ag-projects.com