# Future of VoIP

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# I am Adrian Georgescu

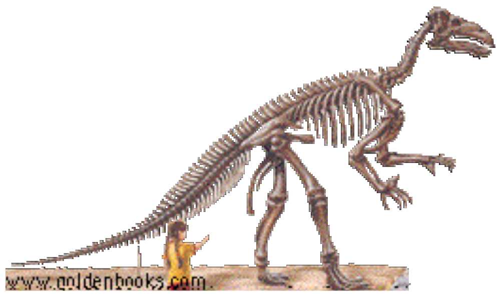
The opinions expressed in this presentation belong
to myself,
my company
and most of my friends

# The Future of VoIP

# Current telecommunications landscape

or the present of VoIP

From the old PSTN only the E.164 numbering plan remains



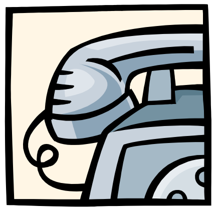
because SS7 and ISDN, the circuit switched networks reached their end-of-life replaced by one “all IP” global network, the Internet

Replacement for circuit-switched signaling protocols has been proposed by IETF and ITU, first one was H323 (ITU) and SIP came second (IETF)

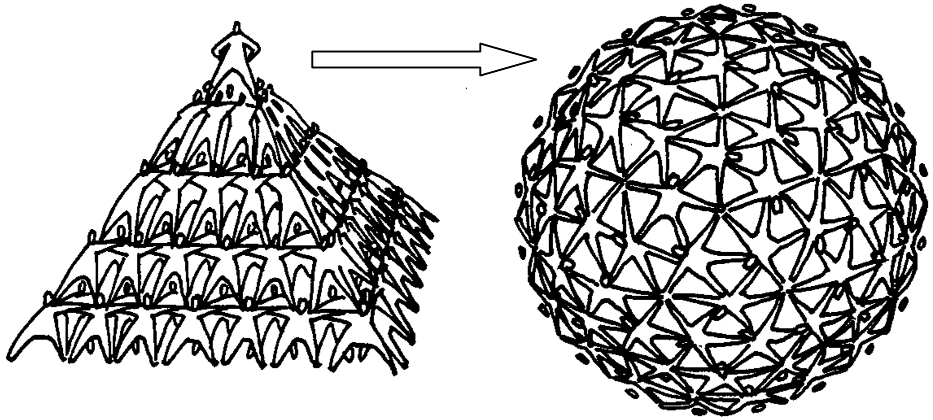
and ENUM complements SIP, it allow IP telephones
to be reached from PSTN

The are alternatives to SIP like H323, MGCP/Megaco, Skinny or IAX.

but most of them focus on voice alone



So, we get rid of PSTN and we move to Internet based real-time communications

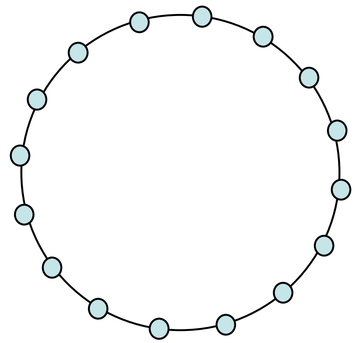


http://www.tmdenton.com

How do we achieve this?

By using SIP - the Session Initiation Protocol

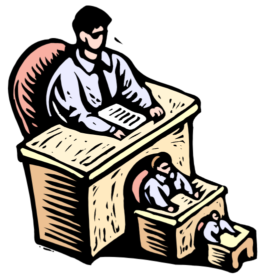
Because SIP allows innovation by placing the intelligence in the end-user nodes



SIP is a horizontal, it allows end-points to find each other (location server) and initiate sessions over the Internet any type of sessions, audio, video, text or whatever comes next

SIP intelligence is distributed among all participating nodes, new applications can be rolled out without upgrades in the network

Which was not possible in the telecom realm



In Internet we learn from experiences of others

SIP makes use of existing proven protocols for addressing and security, it did not re-invent the wheel (DNS, SMTP, HTTP, TLS)

NAT traversal has been addressed in SIP (not H323)

Security and identity has been addressed in SIP

SIP coexists happy with all other protocols

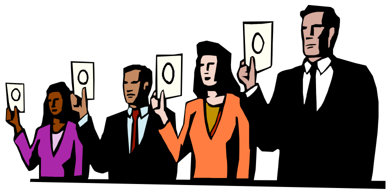
because SIP has the unique “all vendor support” feature

see http://www.sipcenter.com

SIPCenter

SIP enables convergence between fixed and mobile networks, you can change access, device or provider and it still works

But most important, everybody said Yes to SIP



So, what can you do with SIP today?

You can emulate existing telephony services (VoIP)

You can enable IP mobile communications (UMTS) not this release, the next one

You can enable convergence between fixed and mobile networks so ADSL, cable, WiFi, UMTS, WiMax eventually converge for those who did not get it, convergence means from Telecom + Internet we end up with only one network, that is the Internet so Next Generation Networks is the Internet

Internet is a dumb network, the service is at the edge and not in the center

In the center you have only DNS, everybody rely on it for name resolution the end-user is the center and not the service provider

You, the telco, have to adapt to this change to survive

You stay in business as long as your service is interesting enough

because you cannot build walled gardens anymore because by the time you finish building up your walled garden, the customers will be safely outside

Back to SIP services

Voice over IP is the current driving factor

but introduction of VoIP was delayed for years

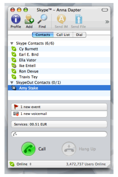
because standardization involves many people, it takes time and is still not complete

just look at the progress of ENUM

and mostly because there was not enough bandwidth

Now we have enough bandwidth so everybody jumped on board and opportunity is sized by the fastest

this is why Skype had so much success



or maybe because the CEO ordered a product
he wanted to use himself

Now we have Voice over IP providers

Which all compete to reach the bottom line, till price per minute equals zero absolute zero is an impossibility

When everybody has left the PSTN and VoIP is free where is the cake?

New services will emerge

contextual communications services, like:

Mobility between networks or access devices, porting identity

# Presence beyond online/offline

# Digital identity and certificate management

# Privacy services

# Notifications based on business triggers

Synchronize data among devices

# Address personalization, ENUM and domain names

# Merging content with communications

All these are multiple color faceplates
of our Internet communications

The end-users may combine all above themselves in their device

So, the cake is there

The future is beyond Voice over IP alone you just need an enabler

and the enabler is Internet

Is just only one “carrier”, the Internet and SIP protocol, the logic on top of it

So, hopefully, you want to move forward

and enable SIP based communications

you do your homework, study IETF standards, visit popular websites like voip-info.org, www.sipcenter.com, go to VON conferences, Google a bit and finally you know what questions to ask and you find out is more than SIP alone, you need some extra ingredients to make it work after all this preparing work, you go and ask a vendor

Do your have a SIP Proxy?

Do your have a SIP Registrar?

Do your have a SIP Redirect server?

Do your have an ENUM management system?

Do you have a system to do NAT traversal?

Do you provide PSTN gateways?

Do you have a class 5 telephony features?

Do your have presence feature, Interactive messaging capability?

Do you have voicemail server, can you do voice to email?

Do you have a system to do CDR mediation and accounting?

Do you have a Provisioning system for all these?

Is it scalable and resilient?

Has it been tested before?

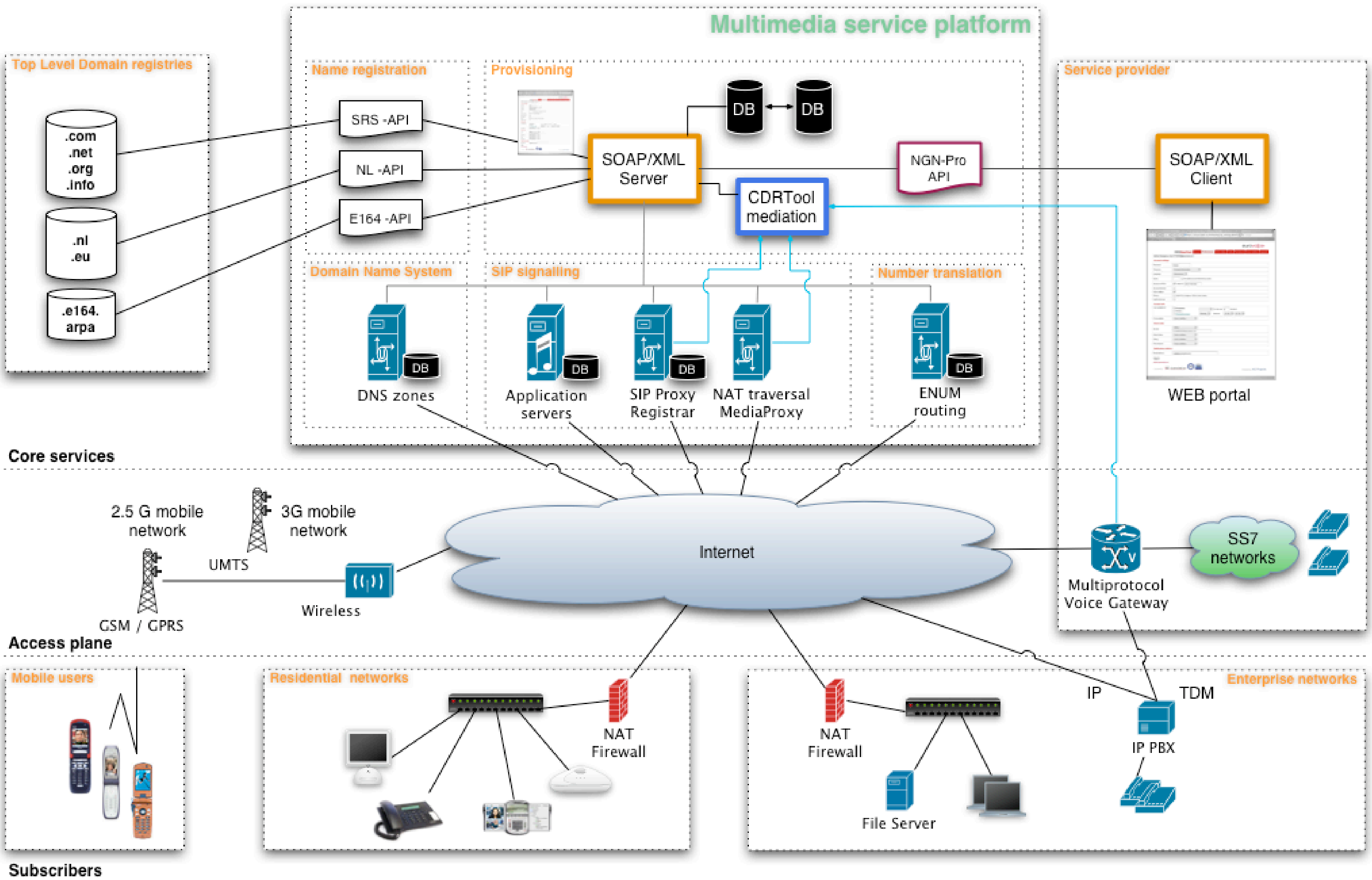
So you end up with maybe five or more vendors and different systems
and different helpdesk to cope with

or you choose an Integrator and you look at 12 months of work ahead
with some unpredictable results

or you come to AG Projects

because we did all this work for you from 2002 we tested it, fine tuned it and trialed it in 2003 we deployed it in commercial services since January 2004 the solution is Multimedia Service Platform

# Multimedia service platform






Powered by AG Projects

is complex but don’t worry we take care of it it simply works, maybe because we are also the end-users of our own solutions

Yes, I use myself for 2 years now

We provide it as hosted service for ourselves and our customers, you can start with it in a matter of weeks so you have from day 1 Audio (Voice over IP), Video, Interactive Messaging and Presence capability

Or we can install it at your location, we deliver it with source code, we provide training about how is build and how it works

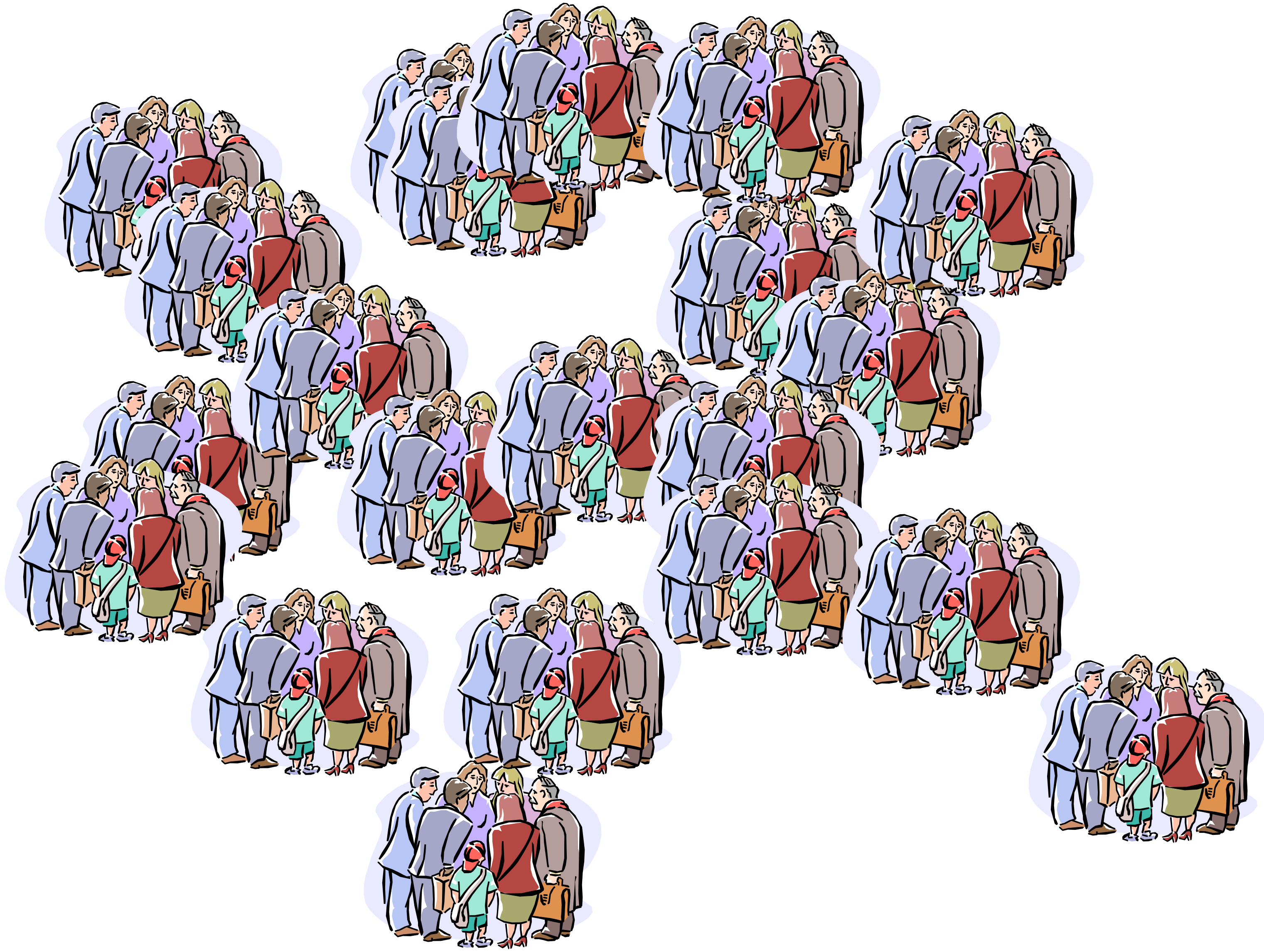
You may set PSTN interconnections with your own gateways, IP PBXs
or global players like MCI and Level3 or Global Crossing

You may build your own website around the platform by using a SOAP/XML provisioning API


# 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

You may scale the platform to handle millions of subscribers



and support all SIP devices available on the market

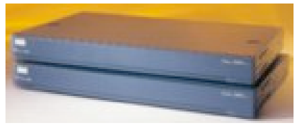
# Supported SIP devices

All SIP compliant software and hardware phones are supported

## ATA adaptors

Cisco ATA


Asterisk

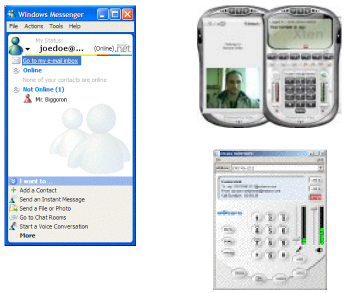


## PSTN gateways

## Video Phones

## Desktop phones

## Software phones




## WiFi Phones

some companies that took the step already

Talkin 2 YA - BPC
Eurovoice - Euroweb
SIP2GO - Sentiro
@Call - ARCOR

# Access to Call Detail Records, traces and rating

CDRTool
Powered by AG Projects

**SIP Express Router (Global Switch)**

Call search | Rating tables| Log| Accounts | 2005-08-05 10:08:50 (Europe/Amsterdam) | CDRTool 3.2.9

Logged in as **adriang (Adrian Georgescu)** Logout

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

105 records found.

Found 6 CDRs for normalization.

From 2005-08-05 09:08 to 2005-08-05 23:55

| | Session start time | SIP caller | In | SIP destination | Out | Dur | Price | KBIn | KBOut | Status | Codecs |
|---|---|---|---|---|---|---|---|---|---|---|---|
| 1N | 2005-08-05 10:08:02 | 0613159157@voipgw01.budgetphone.nl | In | 31747110327@budgetphone.nl | Out | 00:00 | | 0.00 | 0.00 | InternalServerError (500) | |
| 2N | 2005-08-05 10:07:22 | 2460306@sip2go.com | In | +919892546052 (India mobiel 9198) | | 00:00 | | 0.00 | 0.00 | Canceled (487) | |
| 3N | 2005-08-05 10:06:19 | 31162517145@budgetphone.nl | In | +31115567366 (Nederland 31) | Out | 00:00 | | 0.00 | 0.00 | Canceled (487) | |
| 4N | 2005-08-05 10:06:08 | 31717110340@budgetphone.nl | In | +31653779120 (Nederland mobiel 31653) | Out | 00:52 | 0.2010 | 457.80 | 448.55 | Ok (200) | GSM |

**Signalling information**
SIP Session: 1AC66B29-1A10-44B7-945F-EE8E556E57DB@83.85.113.250
SIP Method: Invite from *83.85.113.250:5060*
SIP Status: Ok (200) DelayTime: 11(s)
SIP From: sip:31717110340@budgetphone.nl
SIP To: sip:0031653779120@budgetphone.nl
PSTN Caller ID: 717110340 Privacy disabled
Start time (caller): 2005-08-05 10:06:08 Europe/Amsterdam
Start time (proxy): 2005-08-05 10:06:08 Europe/Amsterdam
Stop time (proxy): 2005-08-05 10:07:00 Europe/Amsterdam
Session duration: 00:52
SIP Proxy: 81.23.228.139
SIP Canonical URI: sip:0031653779120@voipgw01.budgetphone.nl
Next SIP hop: sip:0031653779120@voipgw01.budgetphone.nl
Destination name: Nederland mobiel
Billing Party: 31717110340@budgetphone.nl

**Media information**
Application: Audio
Codecs: GSM
Caller UA: Talkin 2 Ya release 1103m
Called UA: Asterisk PBX

**Rating information**
ConnectFee: 0.0450
--
Span: 1
Duration: 52 s
Appl: PSTN voice (906 KB)
Destination: 31653 (Nederland mobiel)
Customer: domain=budgetphone.nl
StartTime: 2005-08-05 10:06:08 (Europe/Amsterdam)
ProfileId: 441 for weekday
RateId: 441 for 8-19h
Rate: 0.1800 / 60 s
Price: 0.1560

| | Session start time | SIP caller | In | SIP destination | Out | Dur | Price | KBIn | KBOut | Status | Codecs |
|---|---|---|---|---|---|---|---|---|---|---|---|
| 5N | 2005-08-05 10:04:44 | 31237110322@budgetphone.nl | In | +31534302876 (Nederland 31) | Out | 00:43 | 0.0308 | 1,019.92 | 1,090.59 | Ok (200) | G711u |
| 6N | 2005-08-05 10:04:13 | 0434571887@voipgw01.budgetphone.nl | In | 31437110334@budgetphone.nl | | 00:00 | | 0.00 | 0.00 | NotFound (404) | |
| 7N | 2005-08-05 10:03:53 | 0434571887@voipgw01.budgetphone.nl | In | 31437110334@budgetphone.nl | | 00:00 | | 0.00 | 0.00 | NotFound (404) | |
| 8N | 2005-08-05 10:03:13 | 31107110332@budgetphone.nl | In | +31627284919 (Nederland mobiel 31627) | Out | 00:43 | 0.1740 | 247.45 | 252.54 | Ok (200) | G729 |
| 9N | 2005-08-05 10:02:18 | 2460306@sip2go.com | In | +919892546052 (India mobiel 9198) | | 00:00 | | 0.00 | 0.00 | ServiceUnavailable (503) | |
| 10N | 2005-08-05 10:02:09 | 31307110365@budgetphone.nl | In | +31206485016 (Nederland 31) | Out | 00:29 | 0.0273 | 649.14 | 700.78 | Ok (200) | G711u |
| 11N | 2005-08-05 10:01:07 | 2460306@sip2go.com | In | +912228845554 (India (Bombay) 9122) | | 00:44 | 0.1833 | 268.36 | 352.38 | Ok (200) | G729 |

# Management of subscribers

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

| Accounts | Order | Records | Zones | Domains | Support | Help

Logged in as **Adrian Georgescu** Logout

Type: SIP accounts | All accounts | | All record types | Order by: Change date | Desc | Search

| | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|
| 1 | Update | NAPTR | Number | +878102233344209 | Forward to | SIP | sip:2233344209@sip2sip.info | Pref 0 | Owner 2983 | Delete |
| 2 | Update | SIP | Name | Mark Hewitt | Account | | 2233344209@sip2sip.info | Password ••••• | Owner 2983 | Delete |
| 3 | Update | SIPA | Address | mark.hewitt@sip2sip.info | Forward to | | 2233344209@sip2sip.info | | Owner 2983 | Delete |
| 4 | Update | NAPTR | Number | +878102233344208 | Forward to | SIP | sip:2233344208@sip2sip.info | Pref 0 | Owner 2982 | Delete |
| 5 | Update | SIP | Name | demuregoat goaty | Account | | 2233344208@sip2sip.info | Password ••••• | Owner 2982 | Delete |
| 6 | Update | SIPA | Address | demuregoat.goaty@sip2sip.info | Forward to | | 2233344208@sip2sip.info | | Owner 2982 | Delete |
| 7 | Update | NAPTR | Number | +878102233344207 | Forward to | SIP | sip:2233344207@sip2sip.info | Pref 0 | Owner 2970 | Delete |
| 8 | Update | SIP | Name | xxxx xxxxxx | Account | | 2233344207@sip2sip.info | Password ••••• | Owner 2970 | Delete |
| 9 | Update | SIPA | Address | alex.buss@sipline.nl | Forward to | | 2233344040@sipline.nl | | Owner 2969 | Delete |
| 10 | Update | NAPTR | Number | +878102233344040 | Forward to | SIP | sip:2233344040@sipline.nl | Pref 0 | Owner 2969 | Delete |
| 11 | Update | SIP | Name | alex buss | Account | | 2233344040@sipline.nl | Password ••••• | Owner 2969 | Delete |
| 12 | Update | SIP | Name | Michiel Leenaars | Account | | 2233344197@sipline.nl | Password ••••• | Owner 2939 | Delete |
| 13 | Update | NAPTR | Number | +31208005134 | Forward to | SIP | sip:2233344197@sipline.nl | Pref 0 | Owner 2939 | Delete |
| 14 | Update | EMAILF | Address | program@agnext.com | Forward to | | toast@example.com | | Owner 0 | Delete |

Visit our booth outside

or

visit http://ag-projects.com

So finally

the Future of VoIP

VoIP is not alone

VoIP place is on the same shelf with to other successful stories of the Internet like Email and World Wide Web

The future is beyond VoIP

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

Questions?