VoIP @ UniSI

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/* OpenSIPS Summit 2017, Amsterdam */

/* THE CONTEXT */

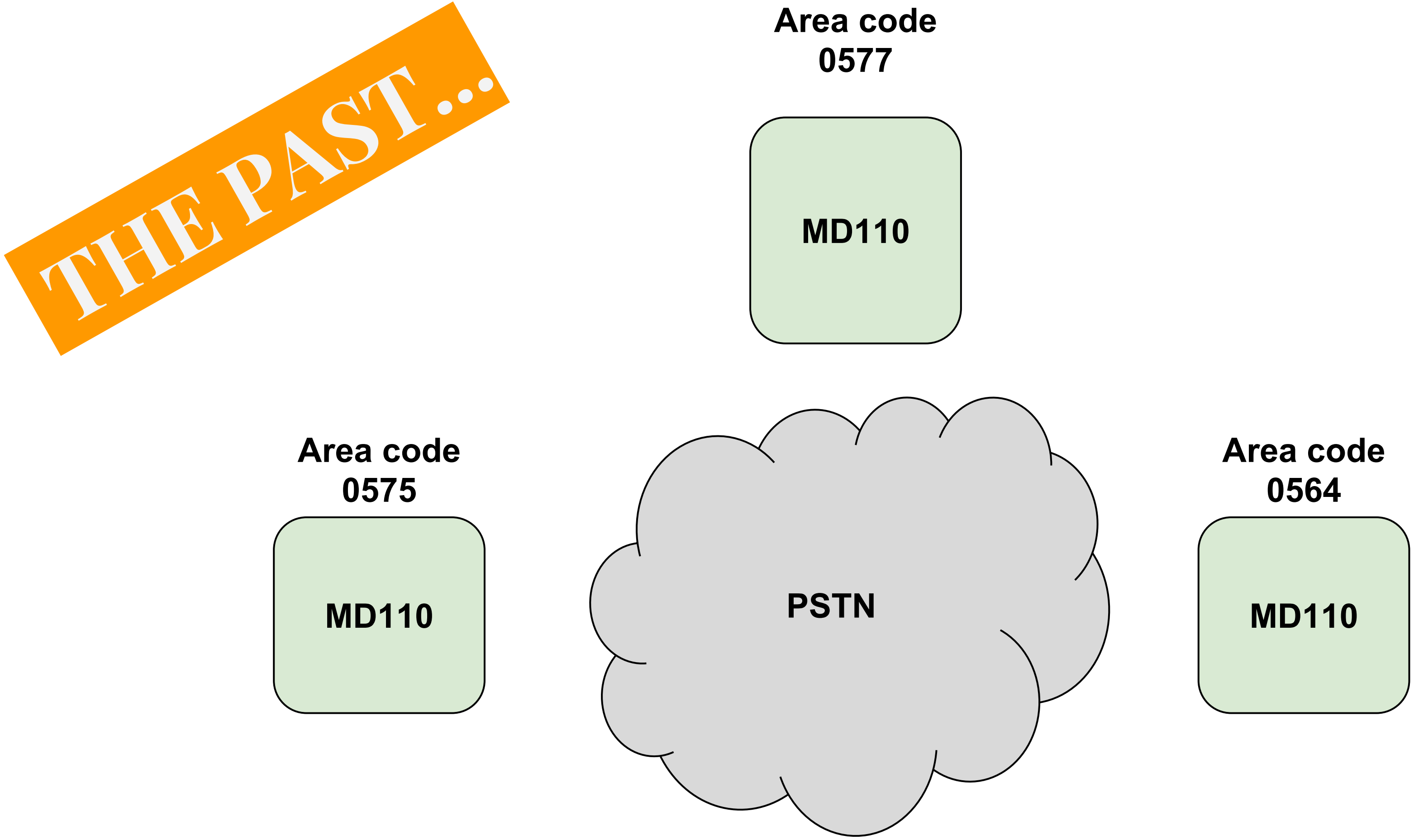
3 Ericsson MD110 phone systems (Siena, Arezzo, Grosseto)

13 Ericsson LIMs in Siena, 2 in Arezzo, 1 in Grosseto

6 PRI (4 in Siena, 1 in Arezzo and 1 in Grosseto)

3 different area codes (0577, 0575, 0564)

<u>1 WAN for the whole University</u>


THE PAST ...
Area code
0577
MD110
Area code
0575
MD110
PSTN
Area code
0564
MD110

/* CHOOSE VOIP SYSTEM */

Free software, based on SIP standards

Stable, carrier grade system, able to support more than 2000 UAC

Modular and expandible design: do less but do it well

Straight configuration, simple to understand and modify

Highly documented and, more important, with an active community behind

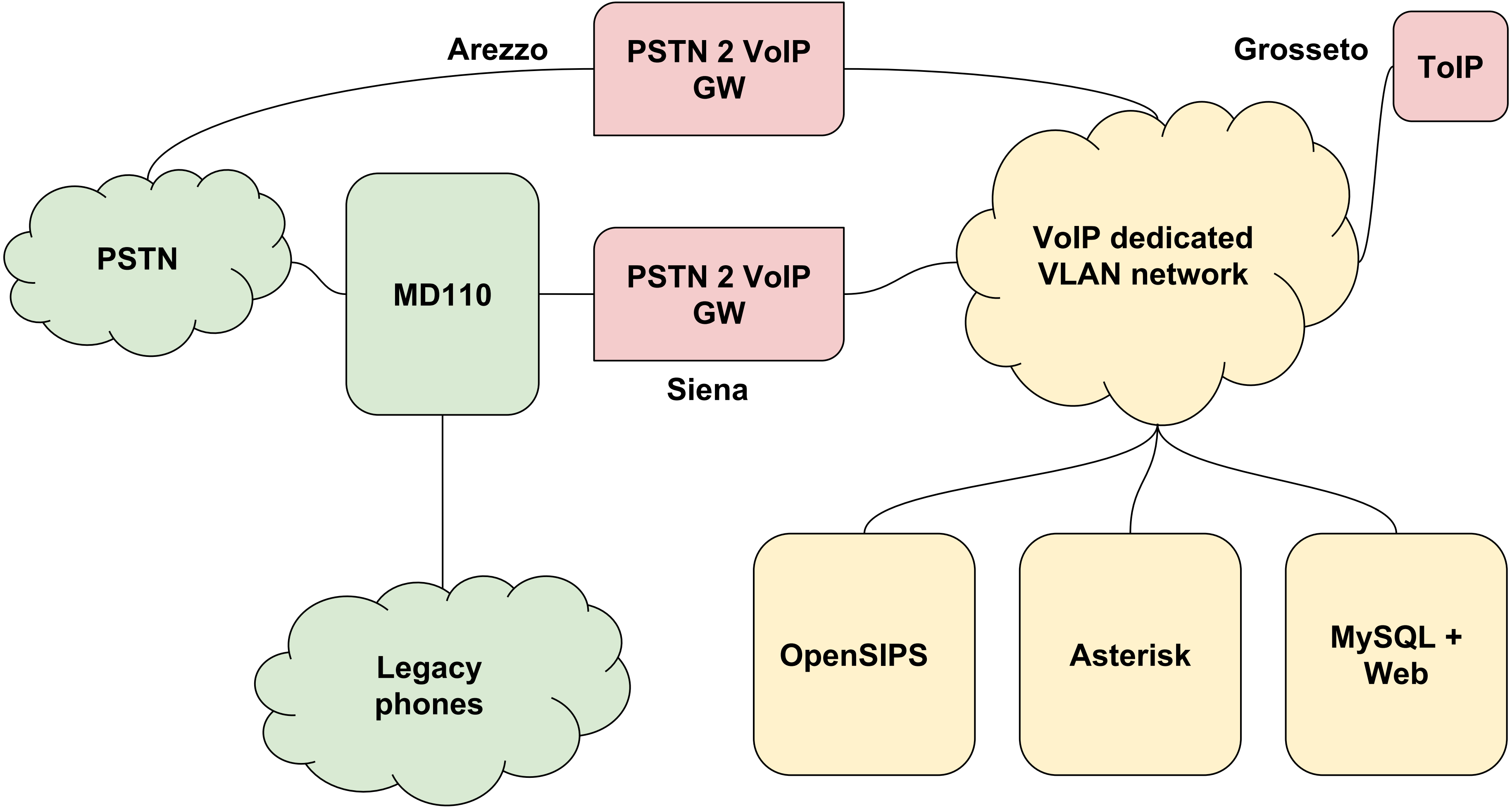
/* PREPARE FOR MIGRATION */

Configure the network and define VLAN and IP ranges

Prepare servers: OpenSIPS for routing, Asterisk for media services, MySQL as DBMS

Prepare PSTN to VoIP gateways and choose a dedicated numeric range

Keep basic services in order to avoid inconvenience from users


Arezzo
PSTN 2 VoIP GW
Grosseto
ToIP
PSTN
MD110
PSTN 2 VoIP GW
VoIP dedicated VLAN network
Siena
Legacy phones
OpenSIPS
Asterisk
MySQL + Web

/* PLANNING ROUTING */

Keep users routines in order to avoid inconveniences in the new hybrid context (legacy and VoIP): **Dialplan** and **Dynamic Routing** module:

- **Dialplan** detect the context of a certain call using regular expressions
- **Dynamic Routing** allows to route a call to the appropriate carrier

id	dpid	pr	match_op	match_exp	match_flags	subst_exp	repl_exp	disabled	attrs
25	0	1	1	^6.{3}$	0			0	voip
2	0	1	1	^5.{3}$	0			0	voip
3	0	2	1	^0(([0-2]|[4-9])[1-9]|80)[0-9].*	0			0	national
6	0	1	1	^2([1-2]|[4-9]).{2}$	0			0	legacy
10	0	1	1	^80.*	0			0	legacy
13	0	1	1	^88.*	0			0	media
14	0	1	1	^05771530(.{3})$	0			0	toip
15	0	1	1	^03[0-9]{7,10}$	0			0	mobile
16	0	2	1	^000([1-9])[0-9].*	0			0	international

```
if(!dp_translate("0", "$rU/$rU", "$avp(dest)")) {
        xlog("L_ERR","420 - Invalid destination $rU\n");
        sl_send_reply("420","Invalid Destination");
        exit;
} else {
        xlog("L_INFO", "$ci - Destination for $rU is $avp(dest)\n");
}
```

/* MULTIPLE INPUT/OUTPUT */

Multiple input/output points:

- PSTN2VoIP gateways
- ToIP
- directly from Internet via E.164
- VoIP phones, software (i.e. softphone) and physical

check_address() and **get_source_group()** from **Permission module,**
is_from_user_enum() from **Enum module.**

/* ASTERISK */

setup a trunk between **OpenSIPS** and **Asterisk**

alias_db_find() from **Alias DB module** to convert accounts into requests for Asterisk (i.e if *5500* is a FAX, convert *$rU* to *FAX_5500*) and simply forward it.

On Asterisk side: IVR, Queues, virtual FAXes, Voicemail, dynamic queues (Agent log-in/log-off) but also fallback handler (i.e. *busy or not available ext*) and other services.

Opensips (opensips.cfg)

```
if(alias_db_find("dbaliases" , "$ru", "$avp(to_alias)")) {
    xlog("L_INFO", "$ci - ALIAS FOUND for $rU (call from $fU): replaced with $avp(to_alias)\n");
    $ru = $avp(to_alias);
}

# IVR number ?
if($rU=~"^\IVR") {
    xlog("L_INFO", "$ci - Forwarding call to $rU\n");
    route(mediabox);
}

# FAX number ?
if($rU=~"^\FAX") {
    xlog("L_INFO", "$ci - Forwarding call to $rU\n");
    route(mediabox);
}

….
```

Asterisk (extensions.conf)

```
[from-voip]
...
; FAXes
exten => _FA[X]_.,1,Noop("from-voip: FAX ${CALLERID(num)} ${EXTEN}")
exten => _FA[X]_.,n,Set(DID=${EXTEN:4})
exten => _FA[X]_.,n,Goto(fax-services,s,1)
; IVR
exten => _IVR_.,1,Set(DID=${EXTEN:4})
exten => _IVR_.,n,Goto(ivr-${DID},s,1)
…
[ivr-5000]
exten => s,1,Queue(queue-5000)
...
```

Don’t forget to setup sip2sip trunk on sip.conf !

/* VOIP.UNISI.IT */

Self-made front-end management system, built on a LAMP system (Linux, Apache, MySQL, PHP), with lightweight frontend made with Bootstrap 3.

System is deeply linked to University employments management system (LDAP for credentials, e-mail, office and other metadata).

VoIP accounts were directly assigned to users and automatically provisioned to VoIP phones via DHCP, TFTP and HTTP, using unique **MAC address**.

Users can log-in on voip.unisi.it anytime and check their **voip account**, their **calls log** and more...

/* REINVENT THE WHEEL ? */

Our reasons to build an self-made VoIP management system:

- built around our systems and compatible with them
- made to fit our needs
- engaging users with innovative services like Telegram Bot for alerts or notifies
- the “Agile” way: we can change everything we want, every time we wish to
- we can build new services anytime
- we can contribute to the Open Source community releasing our codes
- the sky is the only limit: *it’s a nice boost of freedom, don’t you think so?!?*

VoIP calls in real-time

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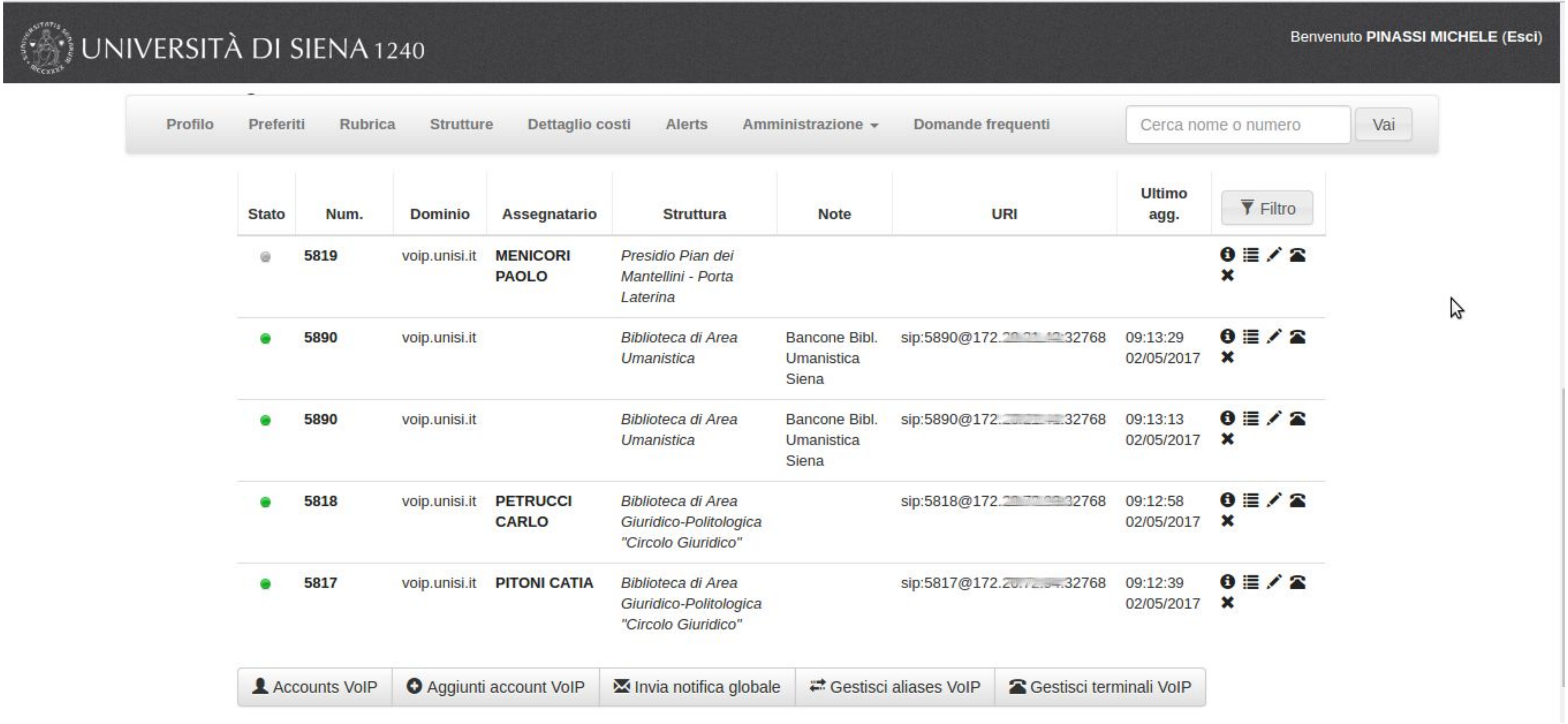
✱ Sistema VoIP

Calls

CallID	From	To	Time	State
31343933373039313039393737-dejke0	[illegible]	[illegible]	09:11:56	On call
313439333730393132323338313734-m	[illegible]	[illegible]	09:12:07	On call
a8754eb330a9dc0c	[illegible]	[illegible]	09:12:25	On call
313439333730393133313539313337-d	[illegible]	[illegible]	09:12:22	On call
313439333730393032353133373836-x	[illegible]	[illegible]	09:10:39	On call
b8f2b7abae528474	[illegible]	[illegible]	09:10:09	On call
952dc85c7380ae9e	[illegible]	[illegible]	09:05:11	On call
cf95336898af3ea4	[illegible]	[illegible]	09:09:55	On call
347e2c1155e7a4e6	[illegible]	[illegible]	09:07:46	On call
d3d1e45a3d262363	[illegible]	[illegible]	09:12:00	On call
6dde7cad618269d2		[illegible]	09:09:33	On call
313439333730383930323335313733-3	[illegible]	[illegible]	09:08:33	On call
313439333730383334323334353938-o	[illegible]	[illegible]	08:59:08	On call
c2832ed1abb8b1d4	[illegible]	[illegible]	09:11:59	On call

VoIP accounts overview and management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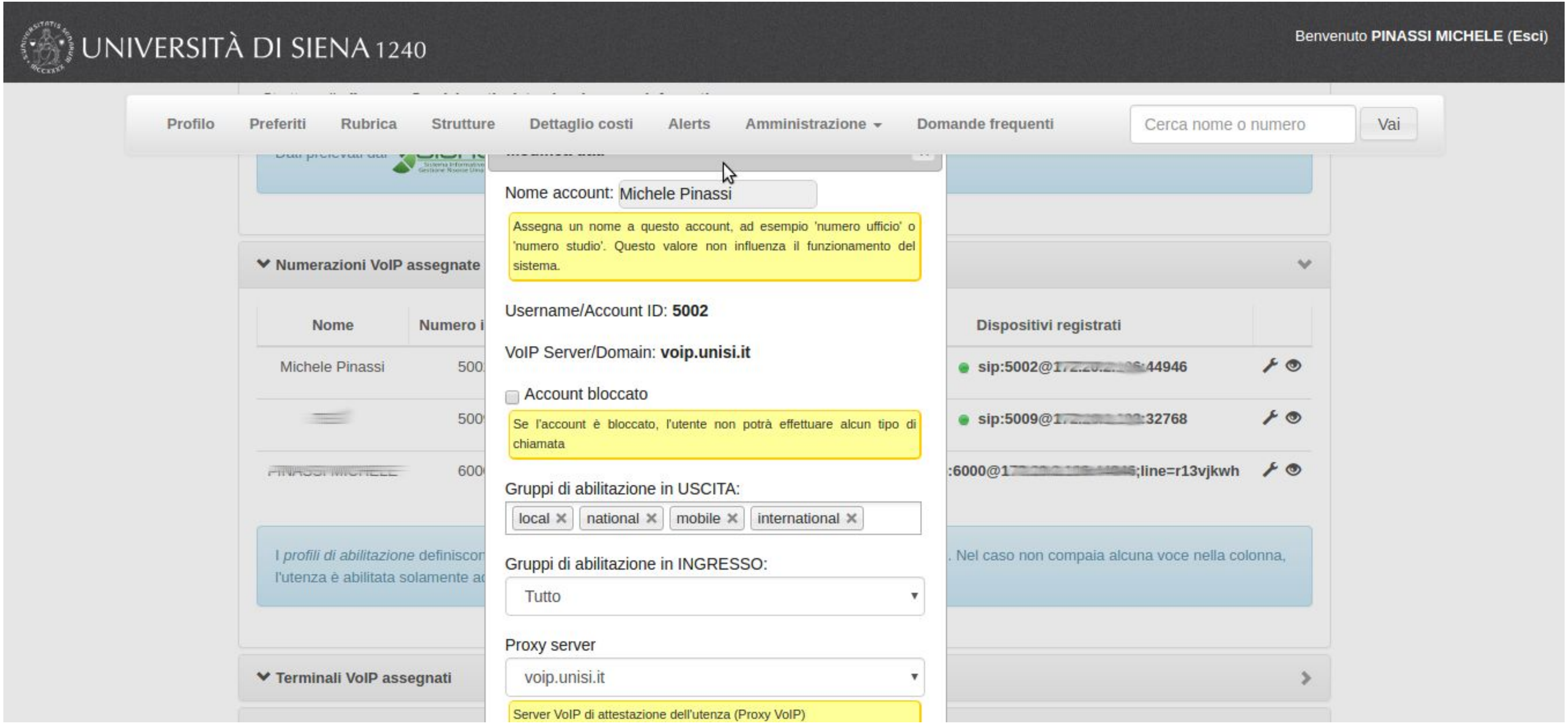
Stato	Num.	Dominio	Assegnatario	Struttura	Note	URI	Ultimo agg.	Filtro
	5819	voip.unisi.it	**MENICORI PAOLO**	*Presidio Pian dei Mantellini - Porta Laterina*				
	5890	voip.unisi.it		*Biblioteca di Area Umanistica*	Bancone Bibl. Umanistica Siena	sip:5890@172.[illegible]32768	09:13:29 02/05/2017	
	5890	voip.unisi.it		*Biblioteca di Area Umanistica*	Bancone Bibl. Umanistica Siena	sip:5890@172.[illegible]32768	09:13:13 02/05/2017	
	5818	voip.unisi.it	**PETRUCCI CARLO**	*Biblioteca di Area Giuridico-Politologica "Circolo Giuridico"*		sip:5818@172.[illegible]32768	09:12:58 02/05/2017	
	5817	voip.unisi.it	**PITONI CATIA**	*Biblioteca di Area Giuridico-Politologica "Circolo Giuridico"*		sip:5817@172.[illegible]32768	09:12:39 02/05/2017	

Accounts VoIP | Aggiunti account VoIP | Invia notifica globale | Gestisci aliases VoIP | Gestisci terminali VoIP

User account details



Users phones and numbers overview

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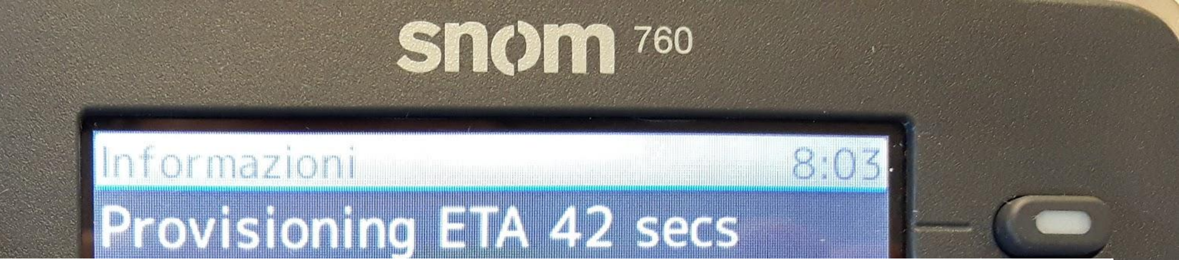
Terminali VoIP assegnati

MAC	Marca/Modello	Nome dispositivo	Linee attestate	Ultimo agg.	
00041[illegible]	SNOM 710	Telefono di PROVA	**5009**	08:46:50 02/05/2017	
00041[illegible]	SNOM 760	Michele Pinassi	**5002**	08:34:14 02/05/2017	
00e1[illegible]	Cisco SPA112	ATA di test	**5009**	15:58:40 24/09/2015	

Numerazioni assegnate

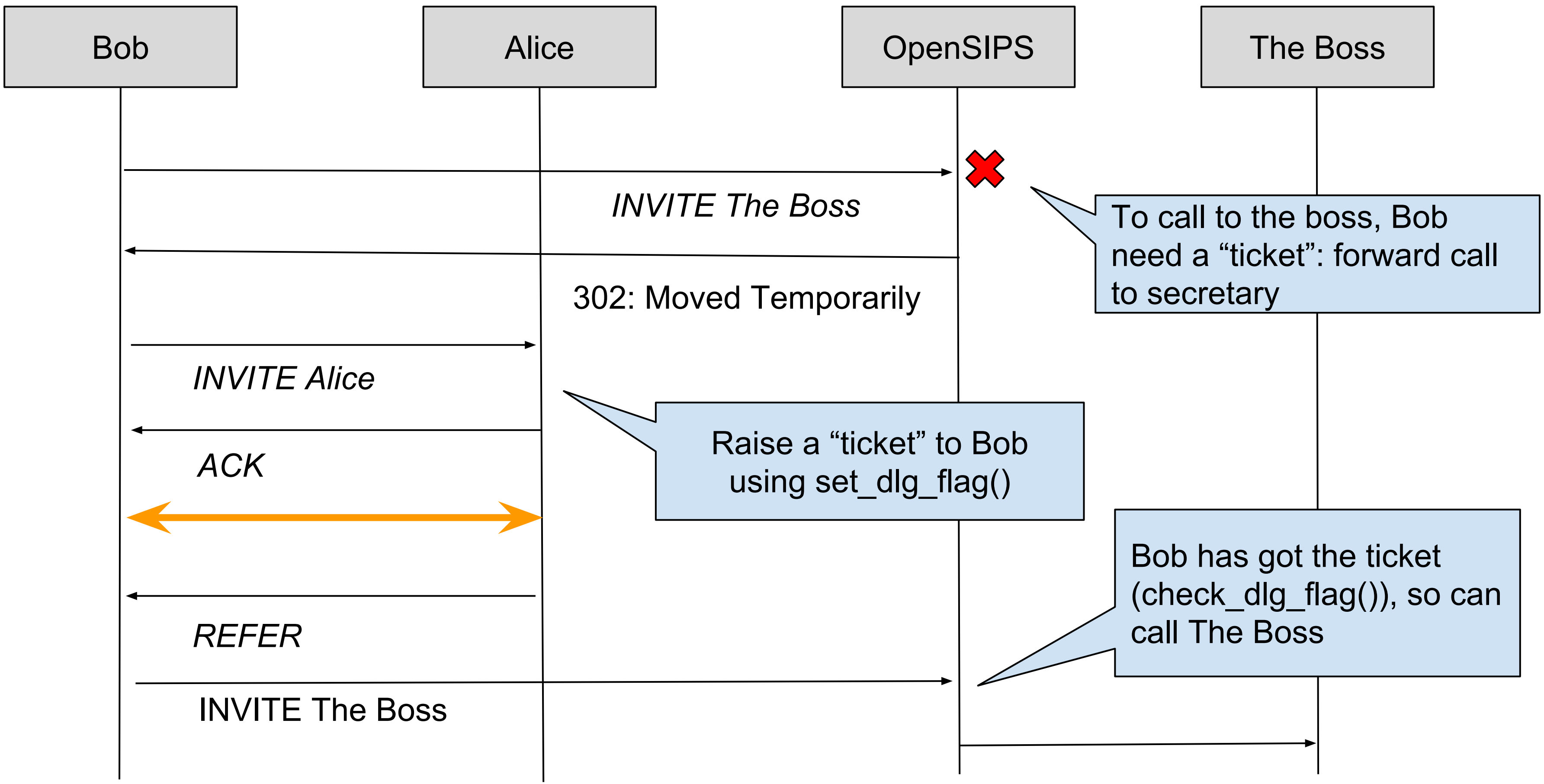
Tipologia	Numerazione	Struttura	Stato	
Interno	**2169**	**Servizio reti, sistemi e sicurezza informatica**	Attiva	
Mobile	[illegible]	**Servizio reti, sistemi e sicurezza informatica**	Attiva	
Interno	**5002**	**Servizio reti, sistemi e sicurezza informatica**	Attiva	
Interno	[illegible]	**Servizio reti, sistemi e sicurezza informatica**	Attiva	
Interno	[illegible]	**Servizio reti, sistemi e sicurezza informatica**	Attiva	

/* OTHER SERVICES */



- **Boss-Secretary** (calls to boss will be forwarded to his/her secretary on first place)
- **Anonymous calls blockage**
- **Conditional/blind call forward**
- **User and global blacklist**
- **Presence** and **Subscribe** *(BLF on phones)*
- **TLS** *(still on testing)*

more to come...


Bob
Alice
OpenSIPS
The Boss
INVITE The Boss
To call to the boss, Bob need a “ticket”: forward call to secretary
302: Moved Temporarily
INVITE Alice
Raise a “ticket” to Bob using set_dlg_flag()
ACK
Bob has got the ticket (check_dlg_flag()), so can call The Boss
REFER
INVITE The Boss

/* AT THE END, CHECKS */

- daily reports about **number of incoming and outgoing calls** and its length;
- **check on calls** to or from disabled numbers (legacy side);
- **trigger if contemporary calls** is near SIP trunks upper limit;
- **changes on employment** database (new, moved or retired employers);
- **monit** to keep *daemons* up;
- **fail2ban** hooked on *auth failed* in opensips log;
- **pike** module;
- **per-user heuristic checks** *(still on planning)* to detect unexpected behaviors;
- **HOMER** (*WIP*);

Questions ?