voip - Nove funkcije #14710

gizmo/asterisk sigma-com officesa, podešenje asteriska, kvalitet call out-a

01.07.2008 10:02 - Ernad Husremović

Status:	Zatvoreno	**Početak:**	01.07.2008
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Odgovorna osoba:	Ernad Husremović	**% završeno:**	100%
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Ciljna verzija:			

Opis

podesiti gizmo account za asterisk

Povezani tiketi:

korelira sa voip - Nove funkcije #14728: asterisk.bring.out.ba <-> gizmo regi...	**Zatvoreno**	**02.07.2008**
korelira sa voip - Podrška #14730: gizmo tarifiranje callout-a	**Zatvoreno**	**02.07.2008**

Historija

#1 - 01.07.2008 10:04 - Ernad Husremović

How do I setup my Asterisk PBX to allow the Gizmo Project softphone to act as an extension number?

Solution This is an advanced topic. If you are new to Gizmo Project and VoIP, then you should probably just ignore this for now.

As you may be aware, one of the ways that Gizmo Project supports Asterisk, is by making it possible for the Gizmo Project softphone to act as an extension number on your Asterisk PBX, just like any other phone (or extension number) in the office.

Please note, that when properly setup, this will work, even when the user is not in your office. This means that even when someone is working from home (or on a business trip), they will still be able to send and receive calls on their office extension number, just like they would if they were sitting at their desk!

To do this, you will need:

1. An installed, and running Asterisk server. (Basic installation of Asterisk is beyond the scope of this article)
2. A Fully Qualified Domain Name that can be used to access your Asterisk server over the Internet. (The actual connection will come from the Gizmo Project server, NOT from the Gizmo Project softphone)

You will need to create an extension number, and password for each user. Then create an entry for each user in the Asterisk "sip.conf" file, and modify the dialplan in the Asterisk "extensions.conf" file, to tell Asterisk to send calls to that extension to the associated SIP account.

New sip.conf settings:

```
[general]
realm=YourDomain
domain=YourDomain

[UserExtension]

type=friend ; allows incoming and outgoing calls
username=UserExtension
secret=UserPassword
mailbox=UserExtension
host=dynamic
dtmfmode=rfc2833
canreinvite=yes
allowguest=yes
insecure=very
promiscredir=yes
```

New extensions.conf settings:

```
[default]
exten => UserExtension,1,Dial(SIP/UserExtension,130,t)
; tells Asterisk where to send the call when someone
```

```
; dials that extension number.
```

#2 - 01.07.2008 11:10 - Ernad Husremović

http://support.gizmoproject.com/index.php?_m=knowledgebase&_a=viewarticle&kbarticleid=402&nav=0

Connect Asterisk & Callweaver to Gizmo5 Sample Config:

Note: Callweaver is about the same except some additional built in STUN parameters.

File:sip.conf

```
[proxy01.sipphone.com]
type=peer
disallow=all
allow=ulaw
allow=ilbc
dtmfmode=rfc2833
host=proxy01.sipphone.com
fromdomain=proxy01.sipphone.com
insecure=very
qualify=yes
fromuser=YOURSIP
authuser=YOURSIP
username=YOURSIP
secret=YOURPASS
canreinvite=no
```

#3 - 02.07.2008 18:19 - Ernad Husremović

evo kako stvar radi na našem asterisku:

postavke za call-out

```
[general]
context=demo
;allowoverlap=no                        ; Disable overlap dialing support. (Default is yes)
bindaddr=0.0.0.0                ; IP address to bind to (0.0.0.0 binds to all)
canreinvite=no
srvlookup=yes
language=bs
realm=asterisk.bring.out.ba
domain=asterisk.bring.out.ba
localnet=192.168.45.0/255.255.255.0
externhost=internet.sigma-com.net
;Specify how often (in seconds) a hostname DNS lookup should be performed for the value entered in 'externhost
'. Default 10 seconds
externrefresh=10
nat=yes
register =>  17473375695:ppppppwddddddddddddd@proxy01.sipphone.com/17473375695

disallow=all
allow=ulaw
allow=alaw
;allow=ilbc
allow=g729
qualify=yes

[proxy01.sipphone.com]
type=peer
disallow=all
allow=ulaw
allow=alaw
;allow=ilbc
dtfmode=rfc2833
host=proxy01.sipphone.com
fromdomain=proxy01.sipphone.com
insecure=very ; To allow registered hosts to call without re-authenticating
qualify=yes
fromuser=17473375695
authuser=17473375695
username=17473375695
secret=pppppppwddddddddd
```

```
canreinvite=no
nat=yes
```

postavka za podešenje gizmo klijenta u naš asterisk lan

```
[80]
type=friend
username=80
password=xxxxxxx
host=dynamic
callerid="gizmo_sigmacom 80"
callgroup=3
pickupgroup=3
nat=yes
qualify=yes
disallow=all
allow=ulaw
allow=alaw
;allow=ilbc
dtfmode=rfc2833
canreinvite=yes
```

gizmo se podešava tako da se logira na asterisk.bring.out.ba, user=80, password=pwd

#4 - 02.07.2008 18:20 - Ernad Husremović

kod prvog testa callout-a (zvao sašu na njegov 032 privatni broj) jasko je prijavio metalni zvuk

onda sam isključio ilbc codec (vidi se da je gore komentarisan) pa je taj metalni zvuk prestao

#5 - 02.07.2008 18:23 - Ernad Husremović

kod posljednjih testova imao sam prekide, a kada sam nazvao moju mamu, žalila se da me ne čuje - da joj prekidam

#6 - 02.07.2008 18:25 - Ernad Husremović

- Naslov promijenjeno iz gizmo/asterisk sigma-com officesa u gizmo/asterisk sigma-com officesa, podešenje asteriska, kvalitet call out-a

#7 - 10.11.2008 10:11 - Ernad Husremović

- Status promijenjeno iz Dodijeljeno u Zatvoreno

- % završeno promijenjeno iz 0 u 100

kod svih ovih provajdera jedna stvar je bitna: minimalno tarifiranje je 1 minuta.

testirao sada poziv 70038732440170 - (officeze via gizmo, pozvao sa siemens gigaset-a) - sale kaže da je veza dobra. i kod mene je veza skroz čista, jedino što osjetim onaj silence supression kada ne razgovaramo

ja sam imao inače kredit nekih 7.9 $

evo gledam sada na mraka-2/vista ima instaliran gizmo klijent. cijena poziva je 0.171 $ a dužina poziva 0:47 - naplatio je 0.171$ gizmo

dobar je ovaj gizmo, kada sam ga tek postavio imao sam probleme sa registracijom na servere ali u zadnje vrijeme to provjerim sa onim echo call-om i to radi