

voip - Greške #13693

sip -> sip ne radi

26.03.2008 15:34 - Ernad Husremović

Status: Zatvoreno	Početak: 26.03.2008
Prioritet: Normalan	Završetak:
Odgovorna osoba: Ernad Husremović	% završeno: 0%
Kategorija:	Procjena vremena: 0.00 sat
Ciljna verzija:	
Opis pokušaj da direktnog spajanja officesa_hernad -> stan1 neuspješan	

Historija

#1 - 26.03.2008 15:35 - Ernad Husremović

evo šta asterisk kaže:

```
-- Executing [1@demo:1] Dial("SIP/officesa_hernad-08203dc0", "'sip/stan1'| 200") in new stack
-- Called stan1
-- SIP/stan1-0820d988 is ringing
-- SIP/stan1-0820d988 answered SIP/officesa_hernad-08203dc0
-- Native bridging SIP/officesa_hernad-08203dc0 and SIP/stan1-0820d988
== Spawn extension (demo, 1, 1) exited non-zero on 'SIP/officesa_hernad-08203dc0'
```

#2 - 26.03.2008 15:39 - Ernad Husremović

ako kažem u sip.conf canreinvite=no

```
[stan1]
type=friend
username=stan1
password=xxx
host=dynamic
callerid="stan-1"
allow=all
qualify=yes
nat=no
canreinvite=no

[officesa_hernad]
type=friend
username=officesa_hernad
password=xx
host=dynamic
callerid="officesa-hernad"
allow=all
qualify=yes
nat=no
canreinvite=no
```

dobijem isti rezultat

```
-- Executing [1@demo:1] Dial("SIP/officesa_hernad-0820b978", "'sip/stan1'| 200") in new stack
-- Called stan1
-- SIP/stan1-0820f8e0 is ringing
-- SIP/stan1-0820f8e0 answered SIP/officesa_hernad-0820b978
-- Packet2Packet bridging SIP/officesa_hernad-0820b978 and SIP/stan1-0820f8e0
```

jedino što prijavljuje **Packet2Packet bridging** a ne **Native bridging**

kvaka je što dođe do spajanja bez da druga strana digne slušalicu ?!

#3 - 26.03.2008 15:39 - Ernad Husremović

IAX -> SIP pak radi bez problema

#4 - 26.03.2008 20:33 - Ernad Husremović

ovo bi moglo biti interesantno

<http://www.mail-archive.com/wengophone-devel@lists.openwengo.com/msg05477.html>

[Wengophone-devel] Wengophone 2.2 and Asterisk problem

Alejandro Facultad

Thu, 11 Oct 2007 17:33:06 -0700

I'm using wengophone 2.2 for Linux and I log into an Asterisk SIP server. I have configured the Asterisk to permit "canreinvite=yes" so SIP control packets go among clients and Asterisk and RTP data packets go among clients directly (bypassing Asterisk).

But when I establish a call between two Wengophones, I sniff the communication and I realize that SIP+RTP packets go among clients and Asterisk always (RTP does not bypass Asterisk).

Does Wengophone have an option that set up "canreinvite=no" always or something like this to avoid RTP data packets travel directly among Wengos and not via Asterisk ???

Thanking in advance

greetings

A.F.

#5 - 10.11.2008 10:24 - Ernad Husremović

- Status promijenjeno iz Dodijeljeno u Zatvoreno