KoçSistem

GELECEĞİ KEŞFET

Asterisk IP PBX

Telekomünikasyon Uygulaması Çözümleri

Doğan BULUT | Yazılım Geliştirme Lideri

EĞİTİM İÇERİĞİ

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Asterisk Nedir?

- Asterisk Linux üzerinde çalışan açık kaynak kodlu bir yazılımsal santral'dir.
- Hiçbir özel donanım veya yazılım gereksinimi olmadan telefon görüşmeleri yapılmasına olanak verir.
- 1999 yılından beri geliştirilmekte olup, bir iletişim platformu haline gelmiştir.
- https://www.asterisk.org/get-started/

Tarihçe

❖1999 yılında, Mark Spencer adında genç bir adam tarafından geliştirildi.

❖Kendi ihtiyacı olan santral sistemine 50.000 $ üzerinde fiyat teklifleri verilmesi üzerine kendi santralini geliştirmeye başladı.

❖Başlangıçta kesinlikle bu şekilde devam edeceğini düşünmedi.

❖2001 yılında firmasının adını Digium olarak değiştirdi.

❖Artık açık kaynak kodlu olan yazılım dünyanın her yerinden katkı sağladığı bir proje haline dönüştü.

Kullanım Alanları

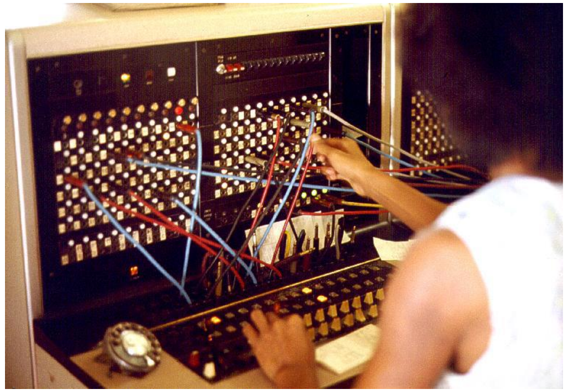
- ❖IP PBX
- ❖VoIP Gateway
- ❖Voicemail Server
- ❖Conference Bridge
- ❖Call Center
- ❖IVR Server

https://www.asterisk.org/get-started/features/

Temel Bilgiler

PSTN

Public Switched Telephone Network, PSTN, dünya genelinde kullanılan devre aktarmalı telefon ağıdır. Başlangıçta sabit analog telefon şebekesi olarak kurulan bu ağ günümüzde neredeyse tamamen sayısal(dijital)dır ve sabit telefonların yanı sıra mobil telefon hatlarını da içermektedir.



Santral Ölçekleri

private branch exchange (PBX).

Karel Ms26s Santral
2 Dış hatlı 6 İç hatlı analog telefon santral s

IPG500 IP Telefon Santralı

Orta ölçekli işletmelerin, kurumsal iletişim ihtiyaçlarını 4 - 20 dış hat ve 12 - 124 dahili kullanıcı kapasite seçenekleri, modüler yapısı ve 100’ün üzerinde programlanabilir kullanım özelliği ile karşılar.

DS200 IP

VOIP

Voice Over Internet Protocol, IP üzerinden ses,video veya mesaj gönderilmesidir. İnternet veya bilgisayar ağları üzerinden çalıştığı için genellikle daha ucuz bazen bedavadır. Bu nedenle günümüzden en çok tercih edilen telekomünikasyon iletişim yönetimidir.

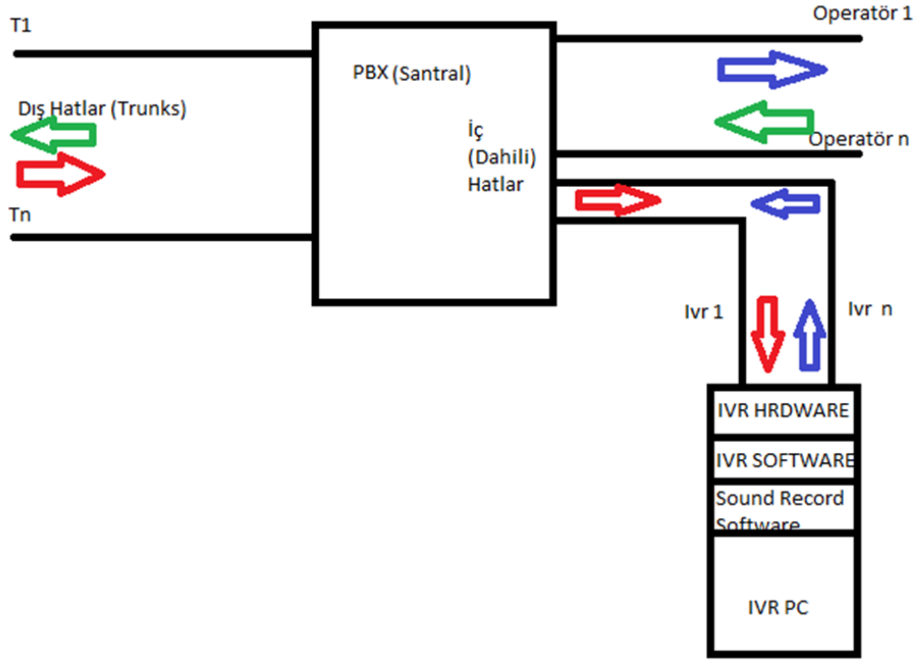
PBX Private Branch Exchange Özel Santral demektir, örneğin şirket içerisinde kullanılan özel telefon şebekesidir. PBX telefon sistemi kullanıcıları, dışarıya telefon etmek için birkaç tane dış telefon hattını paylaşır.

VOIP

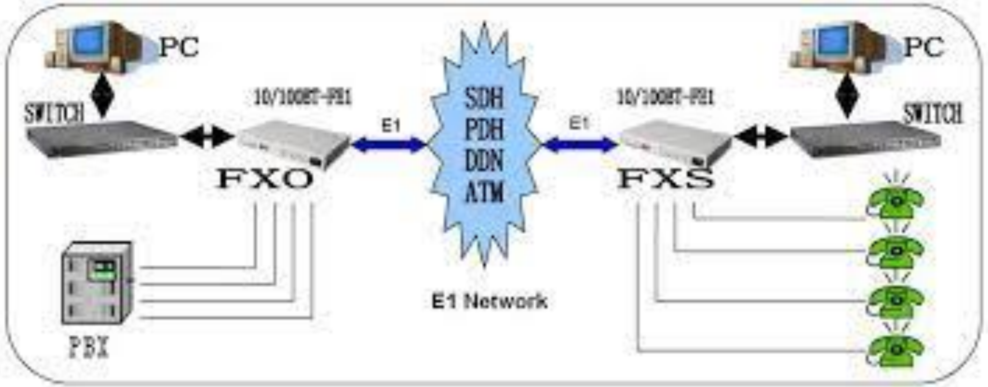
- Bilgisayar ağını kullandığı için ilave telefon altyapısına ihtiyaç duymaz
- Teknolojisi gereği daha maliyetsiz olduğundan görüşme ücretleri daha düşüktür
- Numara tahsis ücretsizdir
- IP Telefon, bilgisayar, akıllı cep telefonu ve yazılımlar ile çalışabilir
- Entegrasyonu analog sistemlere göre daha kolaydır.
- Eski tip analog santral ve telefonlar ile uyumludur. İlave donanımlar ile analog sistemler ile iletişim kurabilir.(FXO,FXS)
- Analog sistemde bir hat üzerinden bir telefon görüşmesi yapılırken VoIP'de böyle bir limit yoktur
- IPv6 ile çalışabilir.

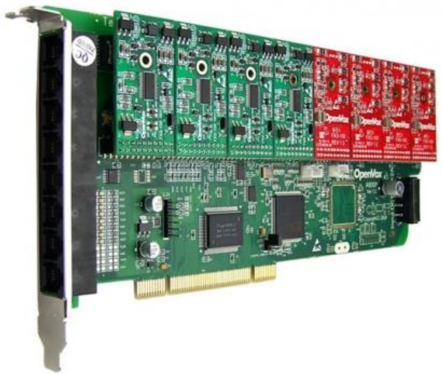
Terimler

- Trunk – Santralimizin içerideki dahililerinden hatlarından dışarı yönünde yapılan aramaların gerçekleştirildiği bağlantısıdır.
- Extension – Santralimizin dahili hatlarıdır.
- IVR – Interactive Voice Response – Sesli Yanıtlama Sistemleri
- CTI – Computer Telephony Integration – Telefon ve bilgisayarların koordine kullanıldığı sistemlerdir. Screen popping, Dialing, Phone Control, Call center ....



FXO FXS Cihazları





Asterisk Kurulum

- ❖Repo: https://github.com/asterisk
- ❖Download: https://www.asterisk.org/downloads/
- ❖RassberryPi: http://www.raspberry-asterisk.org/downloads/
- ❖Docker: https://hub.docker.com/r/tiredofit/freepbx

Git Repo

- https://github.com/doganbulut/asterisk

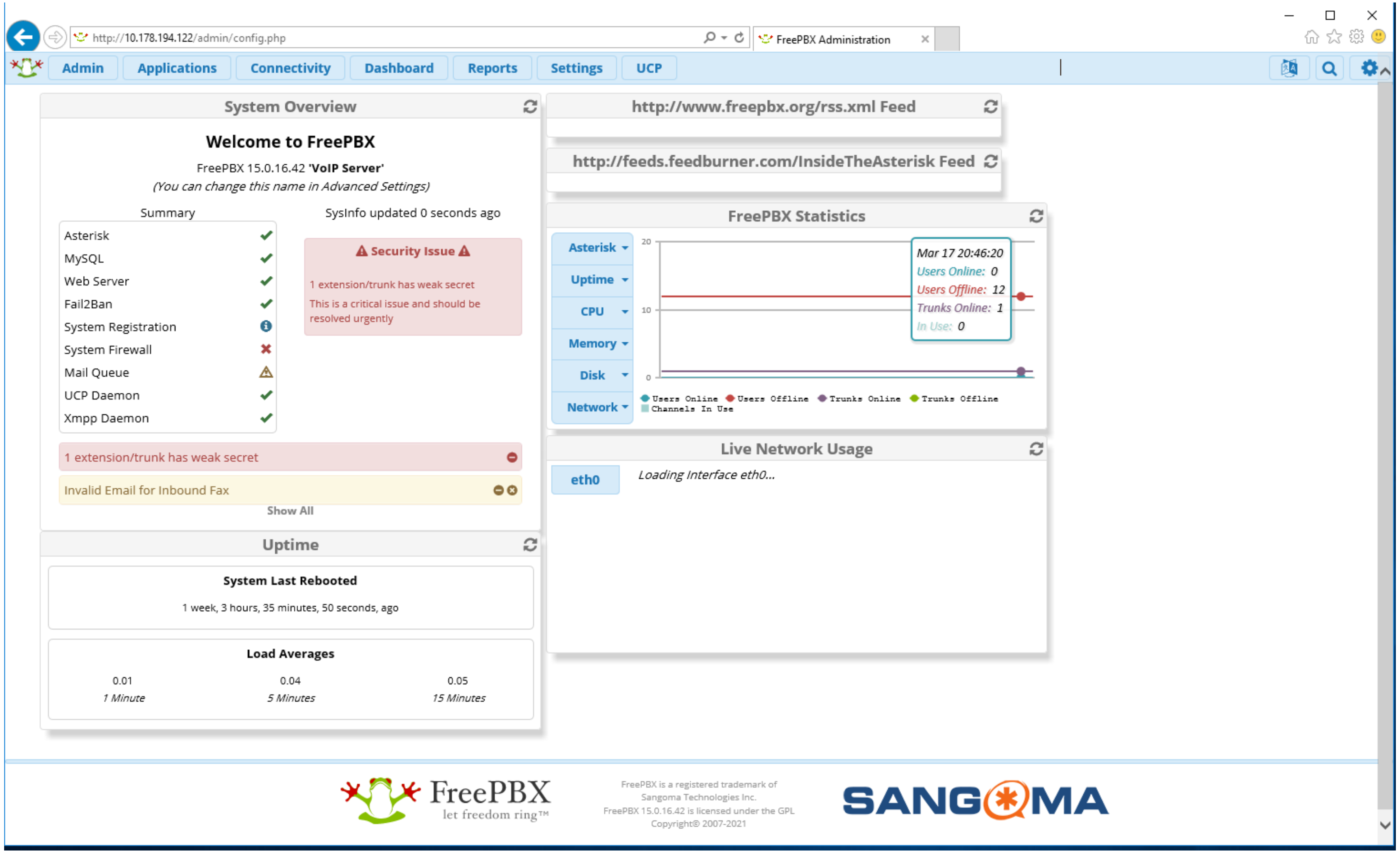
CLI - Log - Debug

- asterisk –rvvvvv
- asterisk -rddddd (core set debug off)
- core show help

Path

/var/log/asterisk

FreePBX


http://10.178.194.122/admin/config.php
FreePBX Administration
Admin
Applications
Connectivity
Dashboard
Reports
Settings
UCP
System Overview
Welcome to FreePBX
FreePBX 15.0.16.42 'VoIP Server'
(You can change this name in Advanced Settings)
Summary
SysInfo updated 0 seconds ago
Asterisk
MySQL
Web Server
Fail2Ban
System Registration
System Firewall
Mail Queue
UCP Daemon
Xmpp Daemon
Security Issue
1 extension/trunk has weak secret
This is a critical issue and should be resolved urgently
1 extension/trunk has weak secret
Invalid Email for Inbound Fax
Show All
Uptime
System Last Rebooted
1 week, 3 hours, 35 minutes, 50 seconds, ago
Load Averages
0.01
1 Minute
0.04
5 Minutes
0.05
15 Minutes
http://www.freepbx.org/rss.xml Feed
http://feeds.feedburner.com/InsideTheAsterisk Feed
FreePBX Statistics
Asterisk
Uptime
CPU
Memory
Disk
Network
Mar 17 20:46:20
Users Online: 0
Users Offline: 12
Trunks Online: 1
In Use: 0
Users Online
Users Offline
Trunks Online
Trunks Offline
Channels In Use
Live Network Usage
eth0
Loading Interface eth0...
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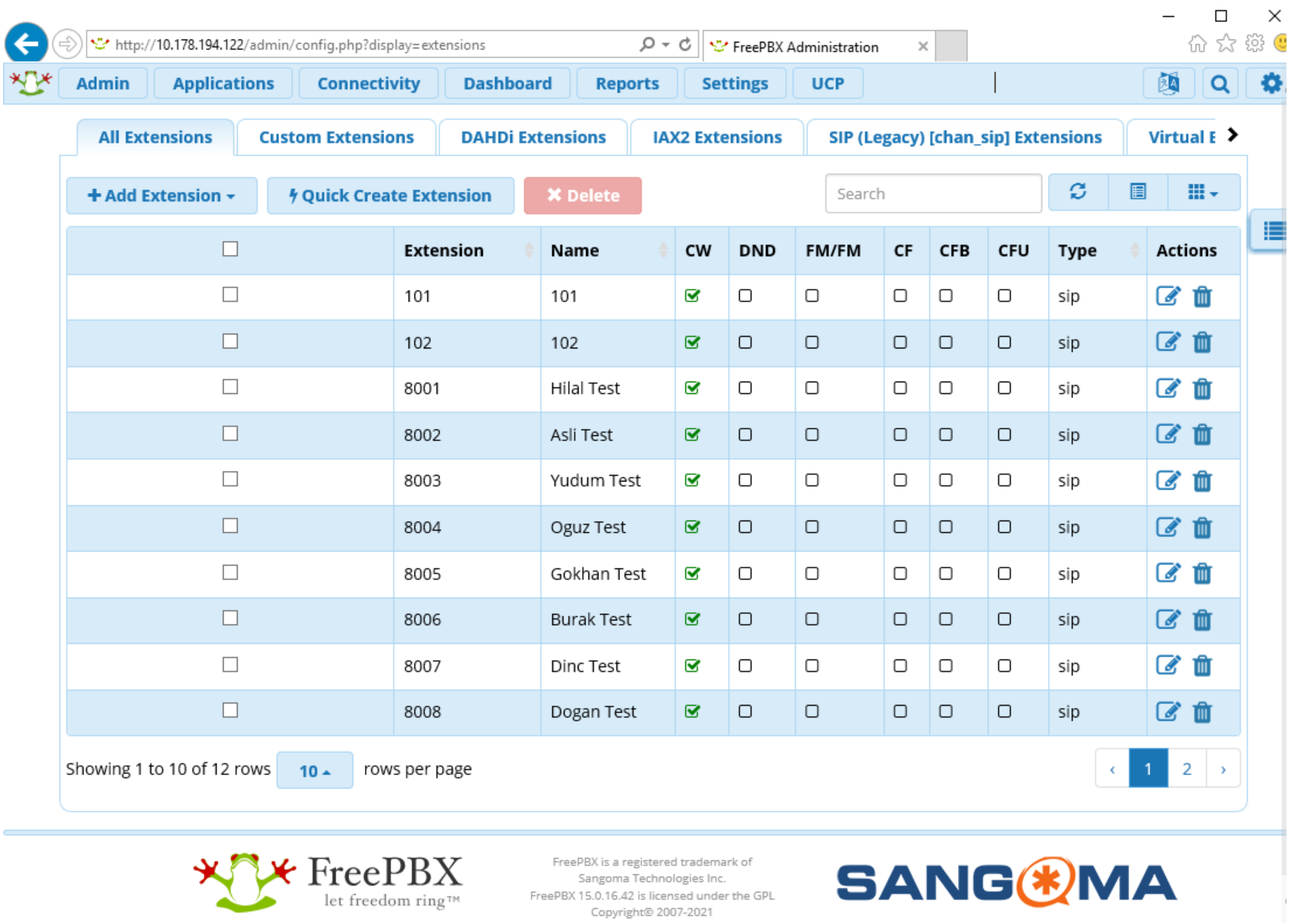
Extensions


http://10.178.194.122/admin/config.php?display=extensions
FreePBX Administration
Admin
Applications
Connectivity
Dashboard
Reports
Settings
UCP
All Extensions
Custom Extensions
DAHDi Extensions
IAX2 Extensions
SIP (Legacy) [chan_sip] Extensions
Virtual E
+ Add Extension
Quick Create Extension
Delete
Search
Extension
Name
CW
DND
FM/FM
CF
CFB
CFU
Type
Actions
101 101 sip
102 102 sip
8001 Hilal Test sip
8002 Asli Test sip
8003 Yudum Test sip
8004 Oguz Test sip
8005 Gokhan Test sip
8006 Burak Test sip
8007 Dinc Test sip
8008 Dogan Test sip
Showing 1 to 10 of 12 rows
10
rows per page
1
2
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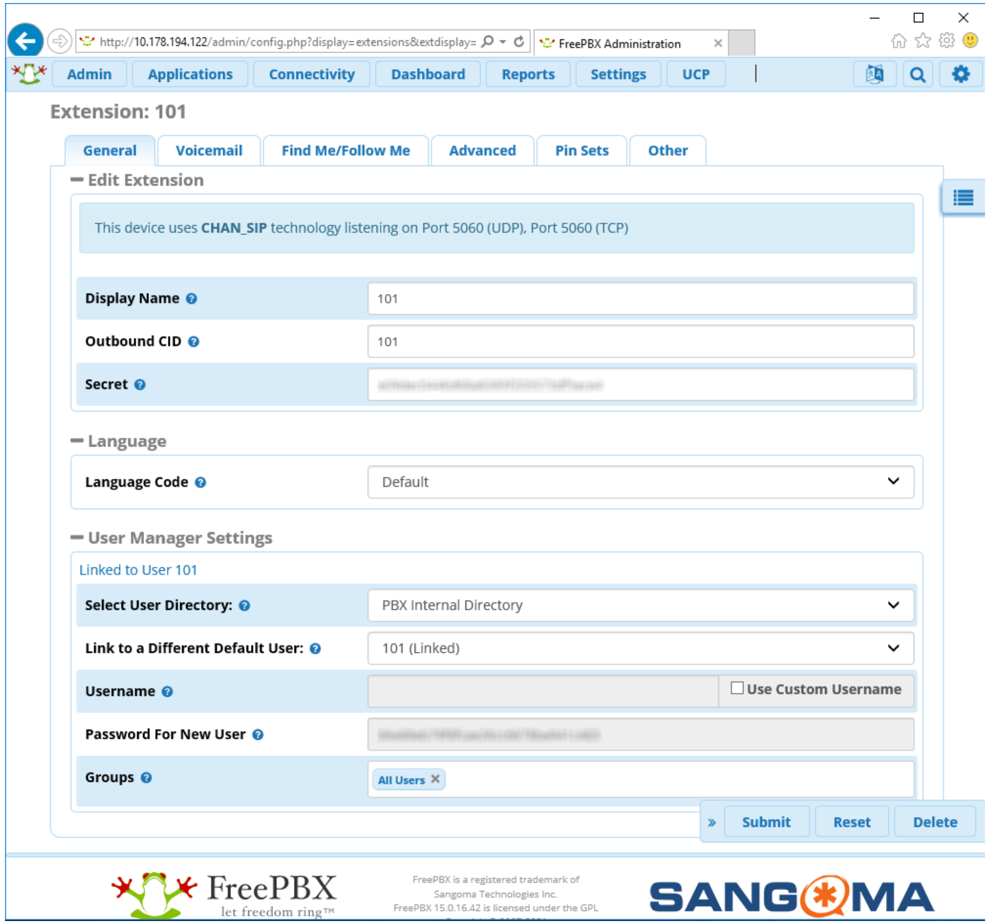
http://10.178.194.122/admin/config.php?display=extensions&extdisplay=
FreePBX Administration
Admin
Applications
Connectivity
Dashboard
Reports
Settings
UCP
Extension: 101
General
Voicemail
Find Me/Follow Me
Advanced
Pin Sets
Other
Edit Extension
This device uses CHAN_SIP technology listening on Port 5060 (UDP), Port 5060 (TCP)
Display Name
101
Outbound CID
101
Secret
Language
Language Code
Default
User Manager Settings
Linked to User 101
Select User Directory:
PBX Internal Directory
Link to a Different Default User:
101 (Linked)
Username
Use Custom Username
Password For New User
Groups
All Users
Submit
Reset
Delete
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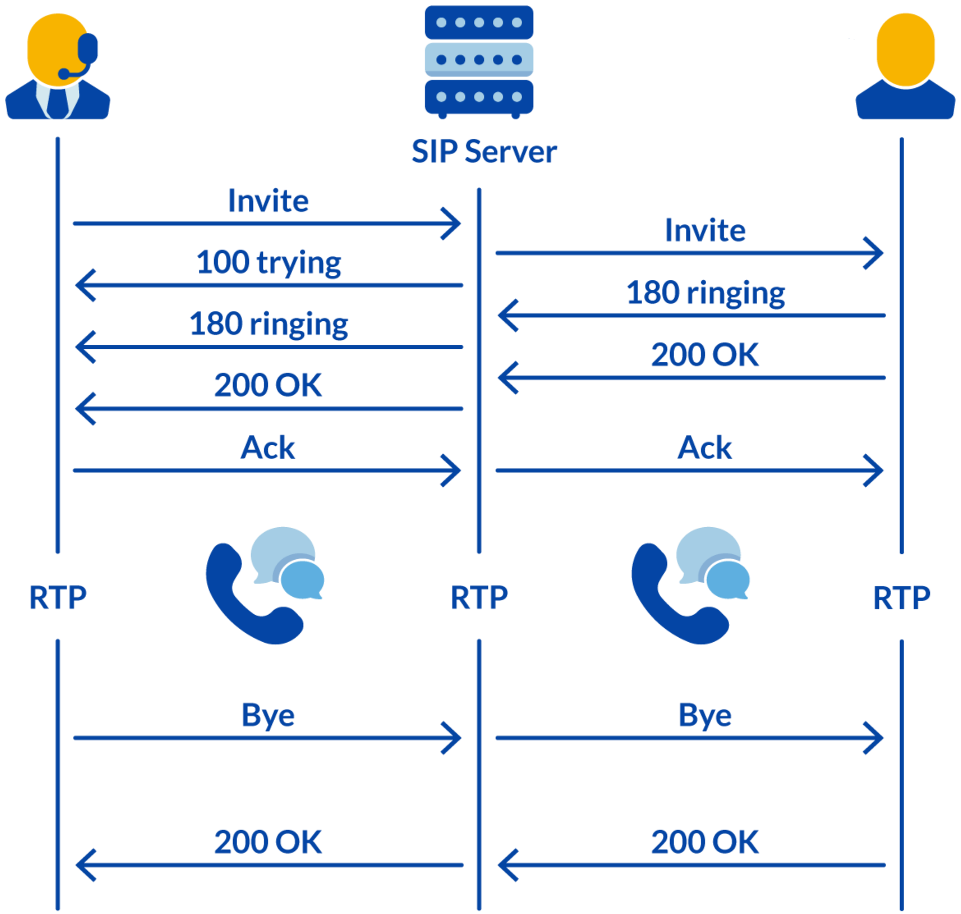
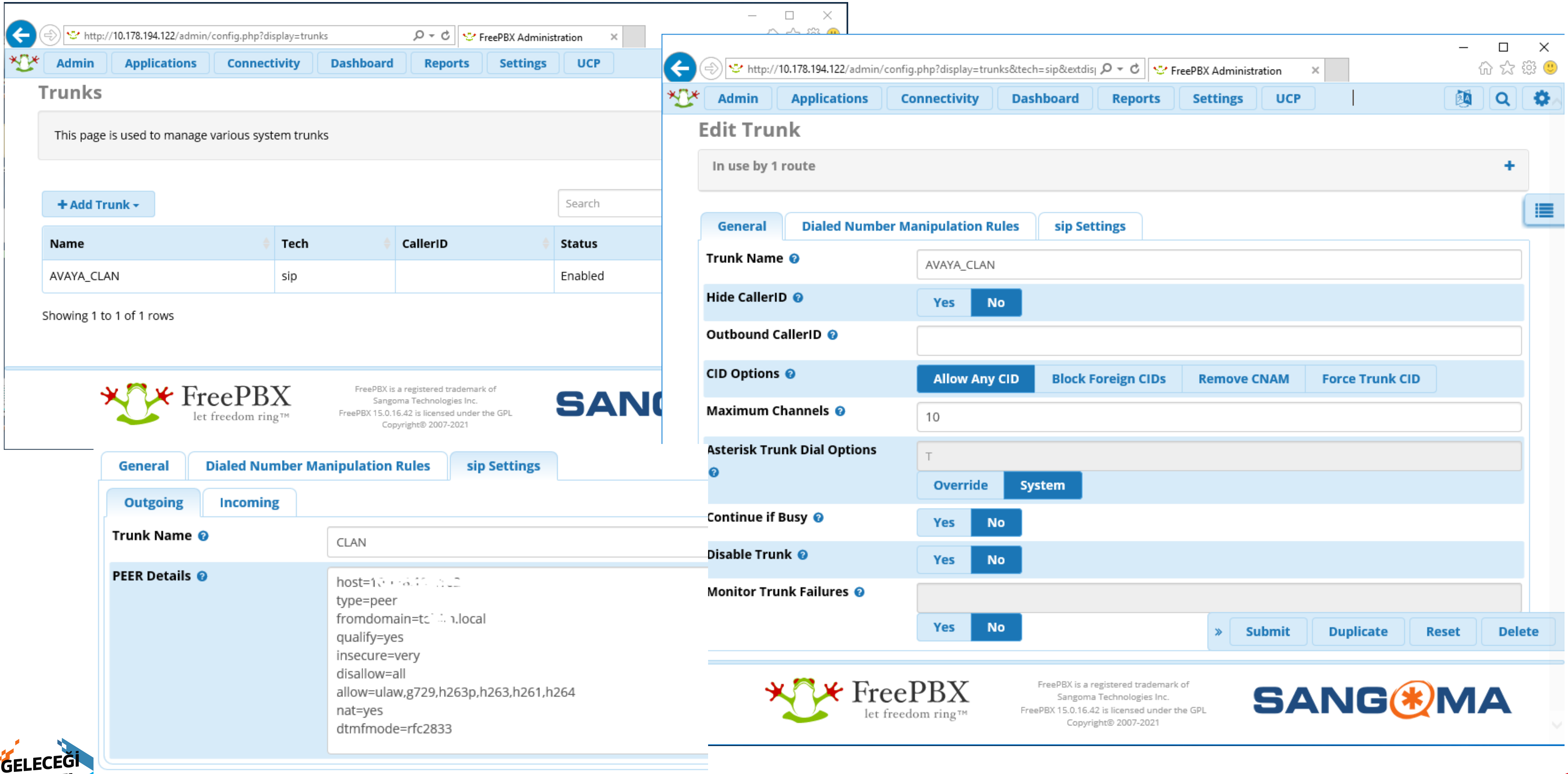
SIP (Session Initiation Protocol - Oturum Başlatma Protokolü)



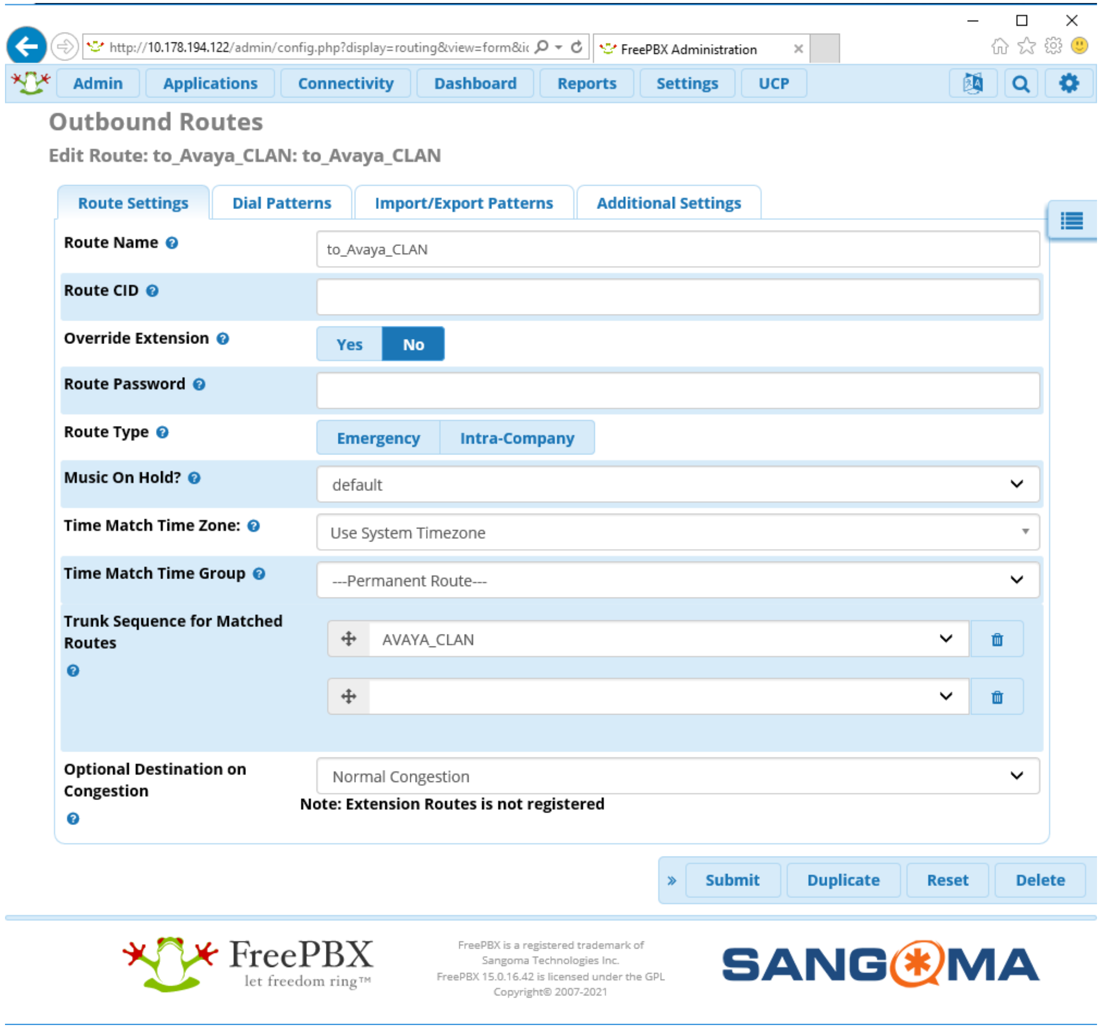
WIRESHARK

WinPcap

Trunk Tanımı



Routes


http://10.178.194.122/admin/config.php?display=routing&view=form&ic
FreePBX Administration
Admin
Applications
Connectivity
Dashboard
Reports
Settings
UCP
Outbound Routes
Edit Route: to_Avaya_CLAN: to_Avaya_CLAN
Route Settings
Dial Patterns
Import/Export Patterns
Additional Settings
Route Name
to_Avaya_CLAN
Route CID
Override Extension
Yes
No
Route Password
Route Type
Emergency
Intra-Company
Music On Hold?
default
Time Match Time Zone:
Use System Timezone
Time Match Time Group
---Permanent Route---
Trunk Sequence for Matched Routes
AVAYA_CLAN
Optional Destination on Congestion
Normal Congestion
Note: Extension Routes is not registered
Submit
Duplicate
Reset
Delete
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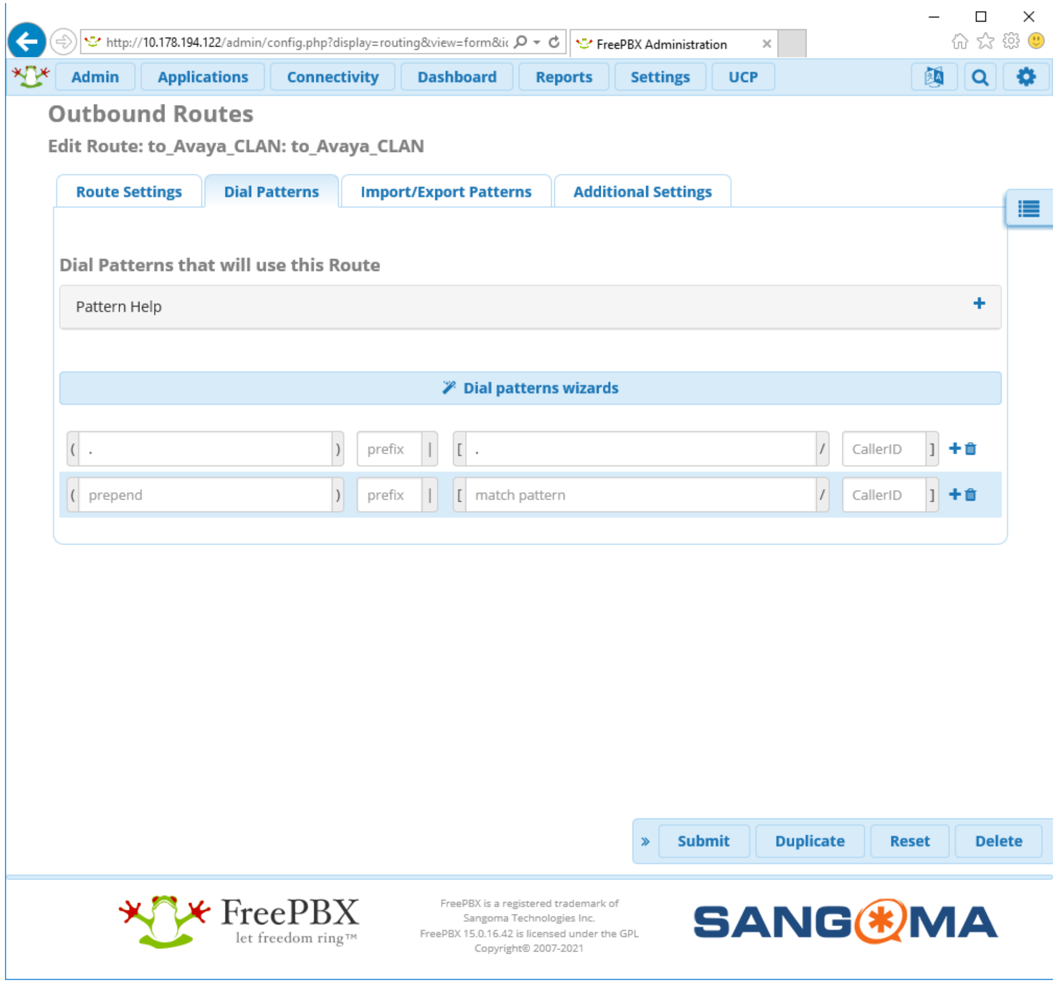
http://10.178.194.122/admin/config.php?display=routing&view=form&ic
FreePBX Administration
Admin
Applications
Connectivity
Dashboard
Reports
Settings
UCP
Outbound Routes
Edit Route: to_Avaya_CLAN: to_Avaya_CLAN
Route Settings
Dial Patterns
Import/Export Patterns
Additional Settings
Dial Patterns that will use this Route
Pattern Help
Dial patterns wizards
prefix
CallerID
prepend
match pattern
Submit
Duplicate
Reset
Delete
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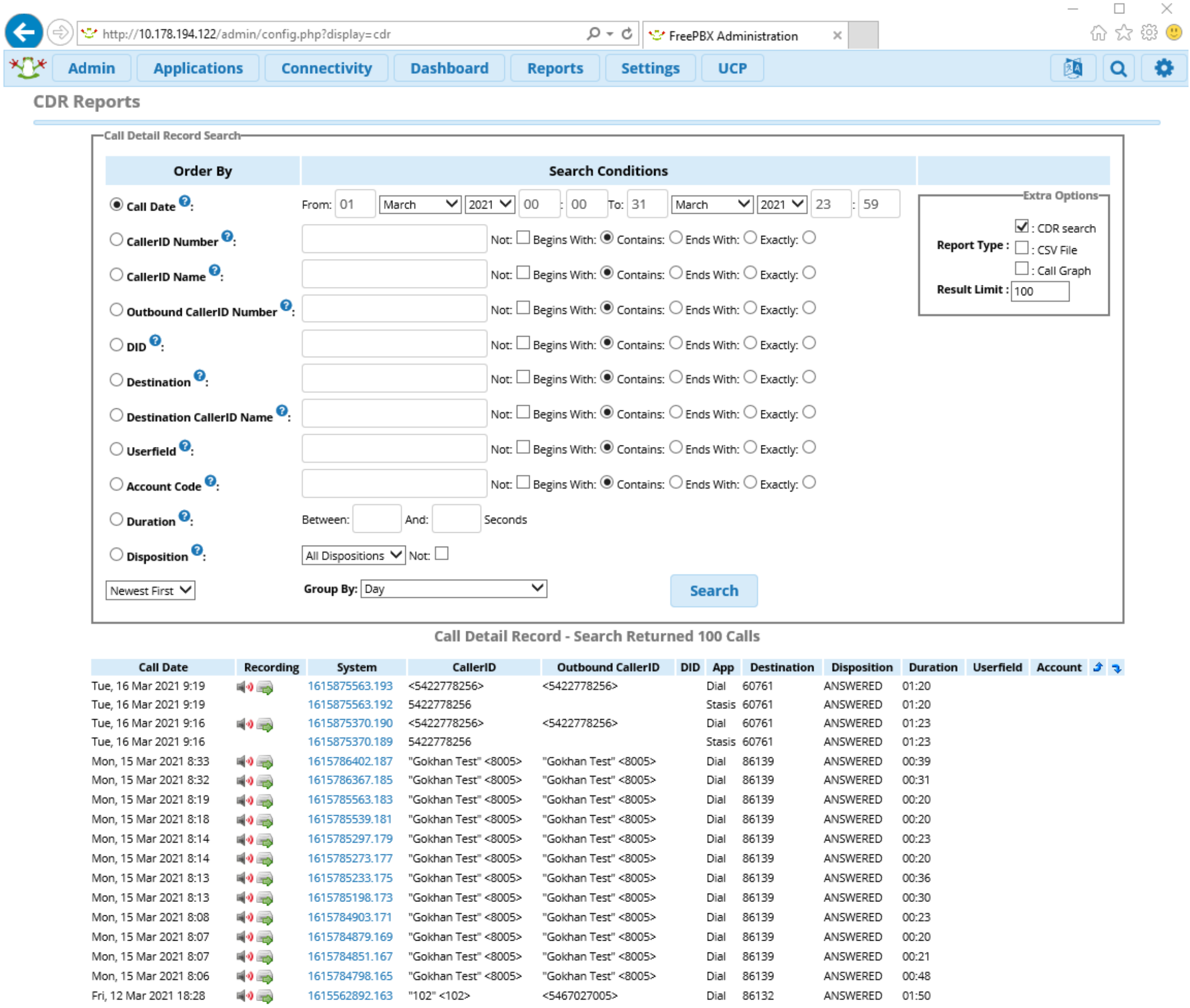
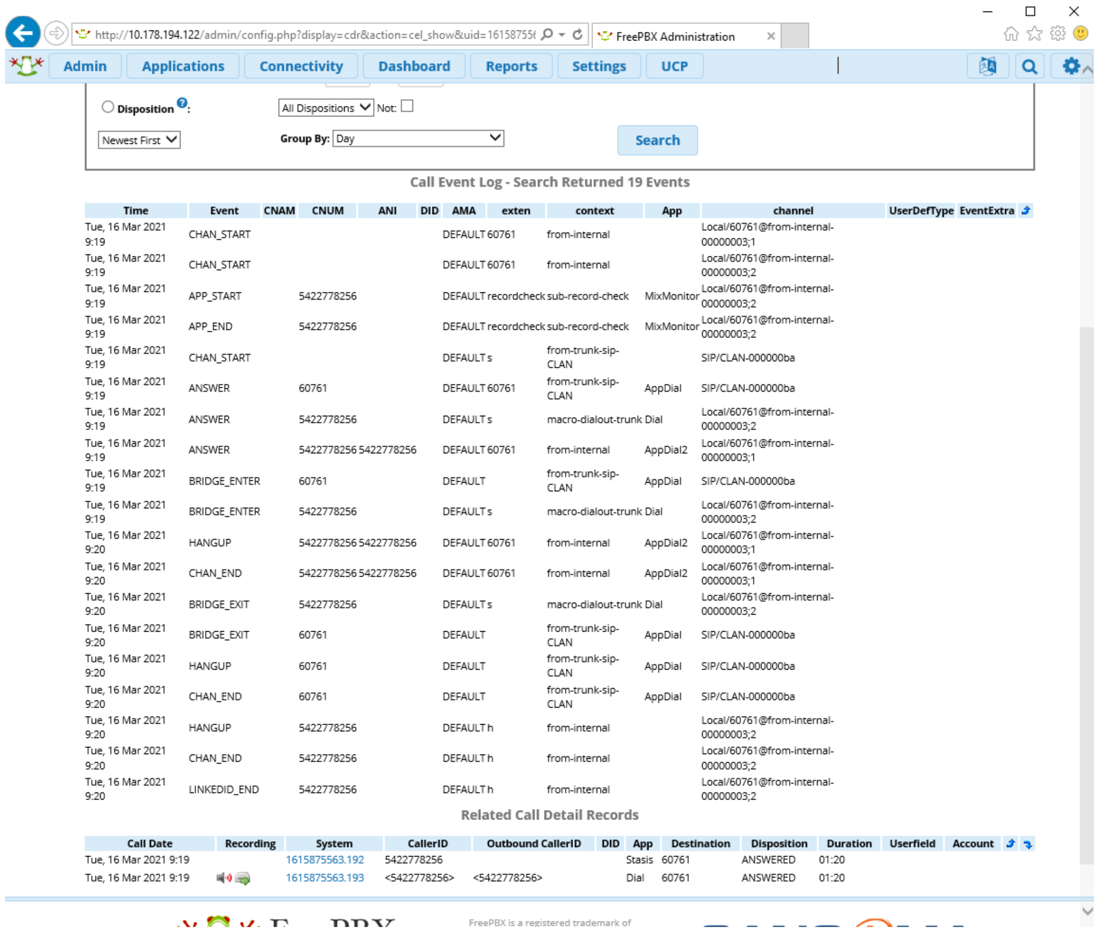
CDR (Call Detail Record)



Call Date	Recording	System	CallerID	Outbound CallerID	DID	App	Destination	Disposition	Duration	Userfield	Account
Tue, 16 Mar 2021 9:19		1615875563.193	<5422778256>	<5422778256>		Dial	60761	ANSWERED	01:20		
Tue, 16 Mar 2021 9:19		1615875563.192	5422778256			Stasis	60761	ANSWERED	01:20		
Tue, 16 Mar 2021 9:16		1615875370.190	<5422778256>	<5422778256>		Dial	60761	ANSWERED	01:23		
Tue, 16 Mar 2021 9:16		1615875370.189	5422778256			Stasis	60761	ANSWERED	01:23		
Mon, 15 Mar 2021 8:33		1615786402.187	"Gokhan Test" <8005>	"Gokhan Test" <8005>		Dial	86139	ANSWERED	00:39		
Mon, 15 Mar 2021 8:32		1615786367.185	"Gokhan Test" <8005>	"Gokhan Test" <8005>		Dial	86139	ANSWERED	00:31		
Mon, 15 Mar 2021 8:19		1615785563.183	"Gokhan Test" <8005>	"Gokhan Test" <8005>		Dial	86139	ANSWERED	00:20		
Mon, 15 Mar 2021 8:18		1615785539.181	"Gokhan Test" <8005>	"Gokhan Test" <8005>		Dial	86139	ANSWERED	00:20		
Mon, 15 Mar 2021 8:14		1615785297.179	"Gokhan Test" <8005>	"Gokhan Test" <8005>		Dial	86139	ANSWERED	00:23		
Mon, 15 Mar 2021 8:14		1615785273.177	"Gokhan Test" <8005>	"Gokhan Test" <8005>		Dial	86139	ANSWERED	00:20		
Mon, 15 Mar 2021 8:13		1615785233.175	"Gokhan Test" <8005>	"Gokhan Test" <8005>		Dial	86139	ANSWERED	00:36		
Mon, 15 Mar 2021 8:13		1615785198.173	"Gokhan Test" <8005>	"Gokhan Test" <8005>		Dial	86139	ANSWERED	00:30		
Mon, 15 Mar 2021 8:08		1615784903.171	"Gokhan Test" <8005>	"Gokhan Test" <8005>		Dial	86139	ANSWERED	00:23		
Mon, 15 Mar 2021 8:07		1615784879.169	"Gokhan Test" <8005>	"Gokhan Test" <8005>		Dial	86139	ANSWERED	00:20		
Mon, 15 Mar 2021 8:07		1615784851.167	"Gokhan Test" <8005>	"Gokhan Test" <8005>		Dial	86139	ANSWERED	00:21		
Mon, 15 Mar 2021 8:06		1615784798.165	"Gokhan Test" <8005>	"Gokhan Test" <8005>		Dial	86139	ANSWERED	00:48		
Fri, 12 Mar 2021 18:28		1615562892.163	"102" <102>	<5467027005>		Dial	86132	ANSWERED	01:50		



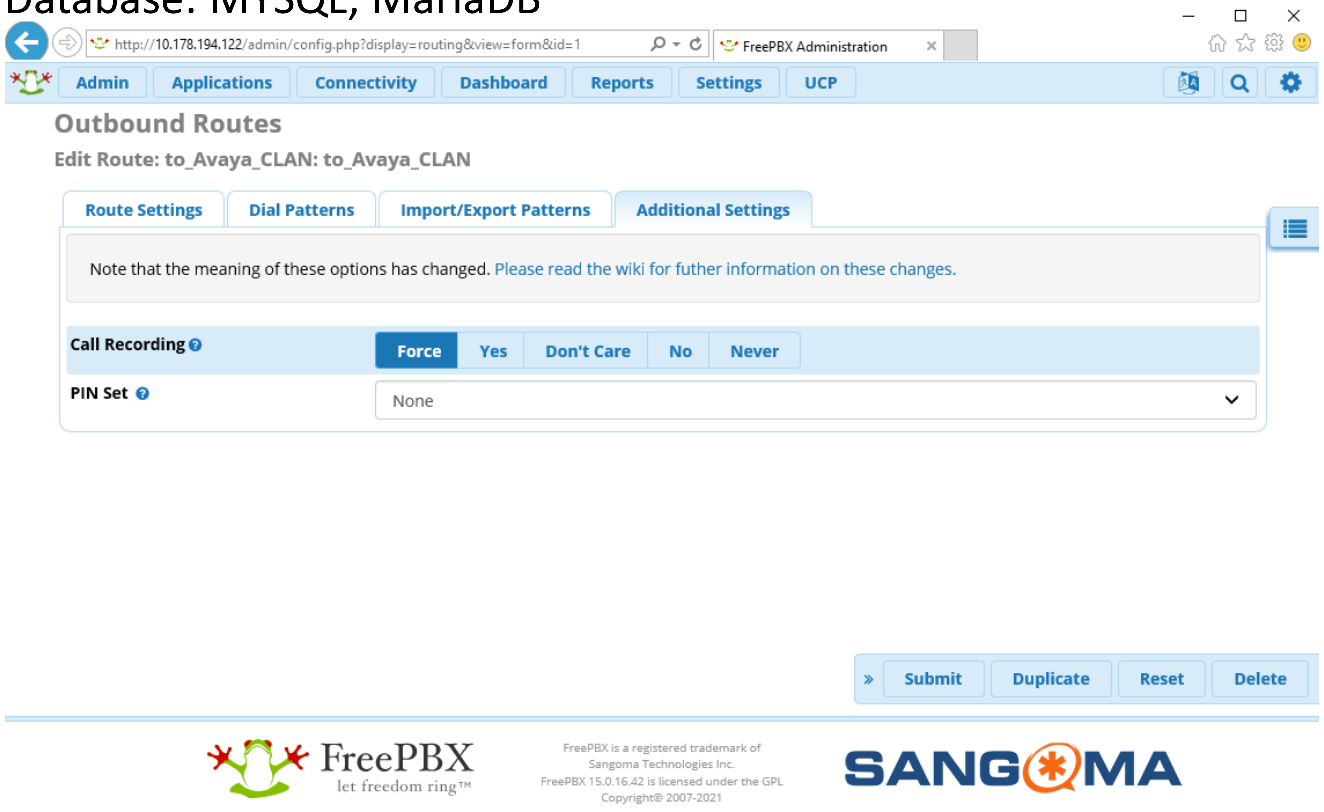
Time	Event	CNAM	CNUM	ANI	DID	AMA	exten	context	App	channel	UserDefType	EventExtra
Tue, 16 Mar 2021 9:19	CHAN_START					DEFAULT	60761	from-internal		Local/60761@from-internal-00000003;1		
Tue, 16 Mar 2021 9:19	CHAN_START					DEFAULT	60761	from-internal		Local/60761@from-internal-00000003;2		
Tue, 16 Mar 2021 9:19	APP_START		5422778256			DEFAULT	recordcheck	sub-record-check	MixMonitor	Local/60761@from-internal-00000003;2		
Tue, 16 Mar 2021 9:19	APP_END		5422778256			DEFAULT	recordcheck	sub-record-check	MixMonitor	Local/60761@from-internal-00000003;2		
Tue, 16 Mar 2021 9:19	CHAN_START					DEFAULT	s	from-trunk-sip-CLAN		SIP/CLAN-000000ba		
Tue, 16 Mar 2021 9:19	ANSWER		60761			DEFAULT	60761	from-trunk-sip-CLAN	AppDial	SIP/CLAN-000000ba		
Tue, 16 Mar 2021 9:19	ANSWER		5422778256			DEFAULT	s	macro-dialout-trunk	Dial	Local/60761@from-internal-00000003;2		
Tue, 16 Mar 2021 9:19	ANSWER		5422778256	5422778256		DEFAULT	60761	from-internal	AppDial2	Local/60761@from-internal-00000003;1		
Tue, 16 Mar 2021 9:19	BRIDGE_ENTER		60761			DEFAULT		from-trunk-sip-CLAN	AppDial	SIP/CLAN-000000ba		
Tue, 16 Mar 2021 9:19	BRIDGE_ENTER		5422778256			DEFAULT	s	macro-dialout-trunk	Dial	Local/60761@from-internal-00000003;2		
Tue, 16 Mar 2021 9:20	HANGUP		5422778256	5422778256		DEFAULT	60761	from-internal	AppDial2	Local/60761@from-internal-00000003;1		
Tue, 16 Mar 2021 9:20	CHAN_END		5422778256	5422778256		DEFAULT	60761	from-internal	AppDial2	Local/60761@from-internal-00000003;1		
Tue, 16 Mar 2021 9:20	BRIDGE_EXIT		5422778256			DEFAULT	s	macro-dialout-trunk	Dial	Local/60761@from-internal-00000003;2		
Tue, 16 Mar 2021 9:20	BRIDGE_EXIT		60761			DEFAULT		from-trunk-sip-CLAN	AppDial	SIP/CLAN-000000ba		
Tue, 16 Mar 2021 9:20	HANGUP		60761			DEFAULT		from-trunk-sip-CLAN	AppDial	SIP/CLAN-000000ba		
Tue, 16 Mar 2021 9:20	CHAN_END		60761			DEFAULT		from-trunk-sip-CLAN	AppDial	SIP/CLAN-000000ba		
Tue, 16 Mar 2021 9:20	HANGUP		5422778256			DEFAULT	h	from-internal		Local/60761@from-internal-00000003;2		
Tue, 16 Mar 2021 9:20	CHAN_END		5422778256			DEFAULT	h	from-internal		Local/60761@from-internal-00000003;2		
Tue, 16 Mar 2021 9:20	LINKEDID_END		5422778256			DEFAULT	h	from-internal		Local/60761@from-internal-00000003;2		

Related Call Detail Records

Call Date	Recording	System	CallerID	Outbound CallerID	DID	App	Destination	Disposition	Duration	Userfield	Account
Tue, 16 Mar 2021 9:19		1615875563.192	5422778256			Stasis	60761	ANSWERED	01:20		
Tue, 16 Mar 2021 9:19		1615875563.193	<5422778256>	<5422778256>		Dial	60761	ANSWERED	01:20		

Call Recording

Ses kayıt klasörü: /var/spool/asterisk/monitör
Database: MYSQL, MariaDB



https://wiki.asterisk.org/wiki/display/AST/Getting+Asterisk+Connected+to+MySQL+via+ODBC

DialPlan

```
[incoming]
exten => s,1,Answer()
exten => s,n,Playback(hello-world)
exten => s,n,Hangup()
```

DialPlans.txt

Path : /etc/asterisk

https://wiki.asterisk.org/wiki/display/AST/Dialplan

https://www.oreilly.com/library/view/asterisk-the-future/9780596510480/ch05.html

CallingFile

Call File Syntax

Channel: <channel> - The channel to use for the new call, in the form technology/resource as in the Dial application. This value is required.

Callerid: <callerid> - The caller id to use.

WaitTime: <number> - How many seconds to wait for an answer before the call fails (ring cycle). Defaults to 45 seconds.

MaxRetries: <number> - Number of retries before failing, not including the initial attempt. Default = 0 e.g. don't retry if fails.

RetryTime: <number> - How many seconds to wait before retry. The default is 300 (5 minutes).

Account: <account> - The account code for the call. This value will be assigned to CDR(accountcode)

Note : mv /tmp/a-test.call /var/spool/asterisk/outgoing/

We create a call file called a-test.call in /tmp/ with the following content:

Channel: SIP/2000

MaxRetries: 2

RetryTime: 60

WaitTime: 30

Context: call-file-test

Extension: 10

https://wiki.asterisk.org/wiki/display/AST/Asterisk+Call+Files

AGI

https://wiki.asterisk.org/wiki/display/AST/Asterisk+18+AGI+Commands

- AGI is analogous to CGI in Apache. AGI provides an interface between the Asterisk dialplan and an external program that wants to manipulate a channel in the dialplan. In general, the interface is synchronous - actions taken on a channel from an AGI block and do not return until the action is completed.

Name	Language	Website	Protocols
Adhearsion	Ruby	http://www.adhearsion.com/	AMI/FastAGI
Asterisk-Java	Java	https://asterisk-java.org/	AMI/FastAGI
PAGI	PHP	https://github.com/marcelog/PAGI	AGI
PHPAGI	PHP	http://phpagi.sourceforge.net/	AGI
Panoramisk	Python+AsyncIO	https://github.com/gawel/panoramisk	AMI/FastAGI
Pyst2	Python	https://github.com/rdegges/pyst2	AMI/AGI
StarPy	Python+Twisted	https://github.com/asterisk/starpy	AMI/FastAGI
Nanoagi	C++	http://sourceforge.net/projects/nanoagi/	AGI
AsterNET	.NET (C#/VB.net)	https://github.com/skrusty/AsterNET	AMI/FastAGI
Ding-dong	node.js	https://www.npmjs.com/package/ding-dong	AGI

AGI Settings

AGI scripts often reside in the AGI directory (usually located in */var/lib/asterisk/agi-bin*)

- exten => 123,1,Answer()
- exten => 123,2,AGI(agi-test.agi)

AGI(), EAGI(), DEADAGI(), AND FASTAGI()

In addition to the AGI() application, there are several other AGI applications suited to different circumstances. While they won't be covered in this chapter, they should be quite simple to figure out once you understand the basics of AGI scripting.

The EAGI() (enhanced AGI) application acts just like AGI() but allows your AGI script to read the inbound audio stream on file descriptor number three.

The DeadAGI() application is also just like AGI(), but it works correctly on a channel that is dead (i.e., a channel that has been hung up). As this implies, the regular AGI() application doesn't work on dead channels.

The FastAGI() application allows the AGI script to be called across the network, so that multiple Asterisk servers can call AGI scripts from a central location.

AMI

- The manager is a client/server model over TCP. With the manager interface, you'll be able to control the PBX, originate calls, check mailbox status, monitor channels and queues as well as execute Asterisk commands.
- AMI is the standard management interface into your Asterisk server. You configure AMI in manager.conf. By default, AMI is available on TCP port 5038 if you enable it in manager.conf.

Asterisk-Java	Java	https://blogs.reucon.com/asterisk-java/	AMI/FastAGI
StarPy	Python+Twisted	https://github.com/asterisk/starpy	AMI/FastAGI
Panoramisk	Python+AsyncIO	https://github.com/gawel/panoramisk	AMI/FastAGI
PAMI	PHP	https://github.com/marcelog/PAMI	AMI
Pyst2	Python	https://github.com/rdegges/pyst2	AMI/AGI
Adhearsion	Ruby	http://www.adhearsion.com/	AMI/FastAGI
node-asterisk	Node.js	https://github.com/danjenkins/node-asterisk-ami	AMI
AMI-IO	Node.js	https://github.com/NumminorihSF/ami-io	AMI
NodeJS-AsteriskManager	Node.js	https://github.com/pipobscure/NodeJS-AsteriskManager	AMI
AsterNET	.NET	https://github.com/AsterNET/AsterNET	AMI/FastAGI
AmiClient	.NET	https://github.com/alexforster/AmiClient	AMI

AMI

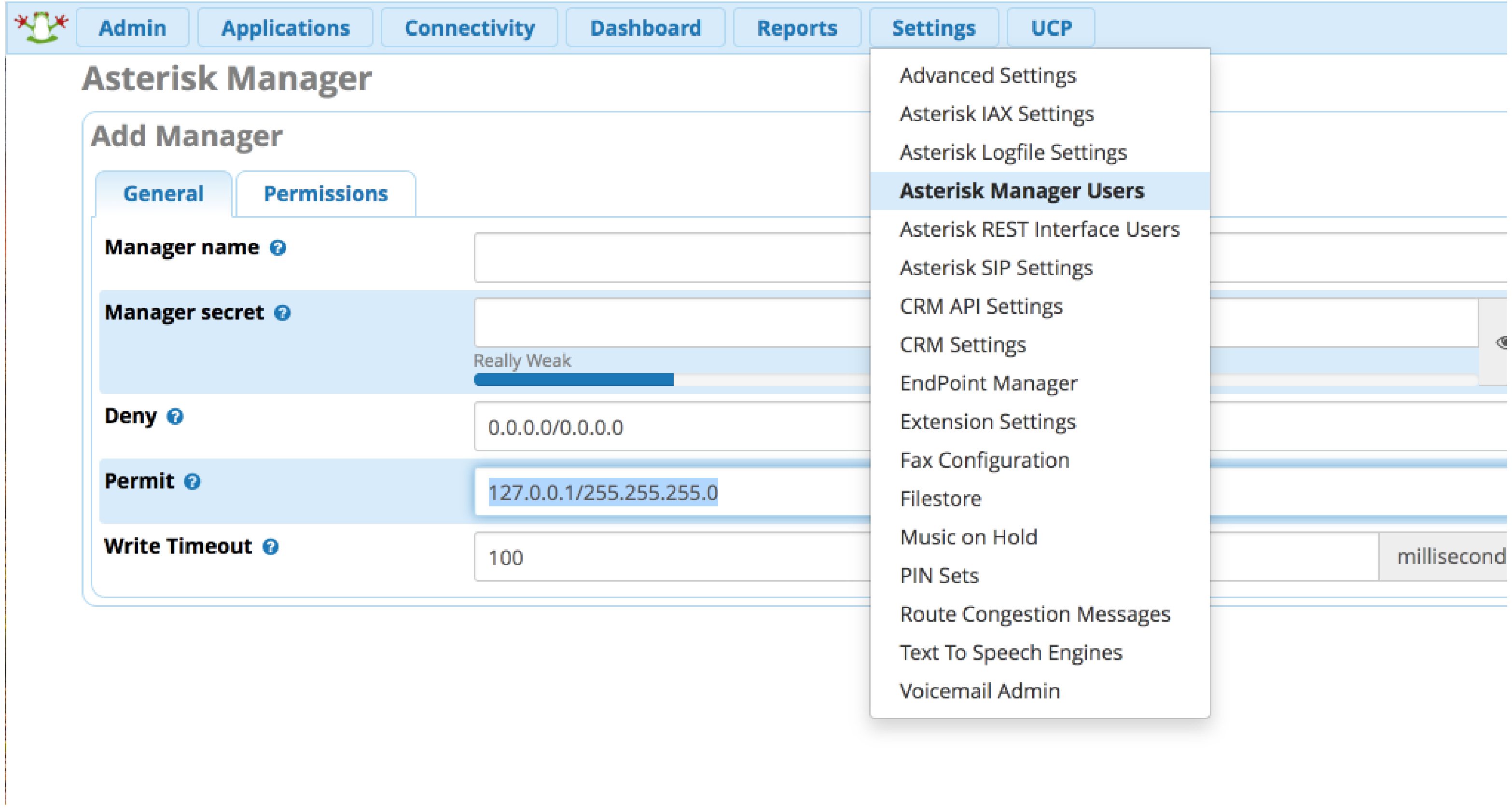
https://wiki.asterisk.org/wiki/display/AST/The+Asterisk+Manager+TCP+IP+API

telnet localhost 5038

Action: Login
ActionID: 1
Username: admin
Secret: secret

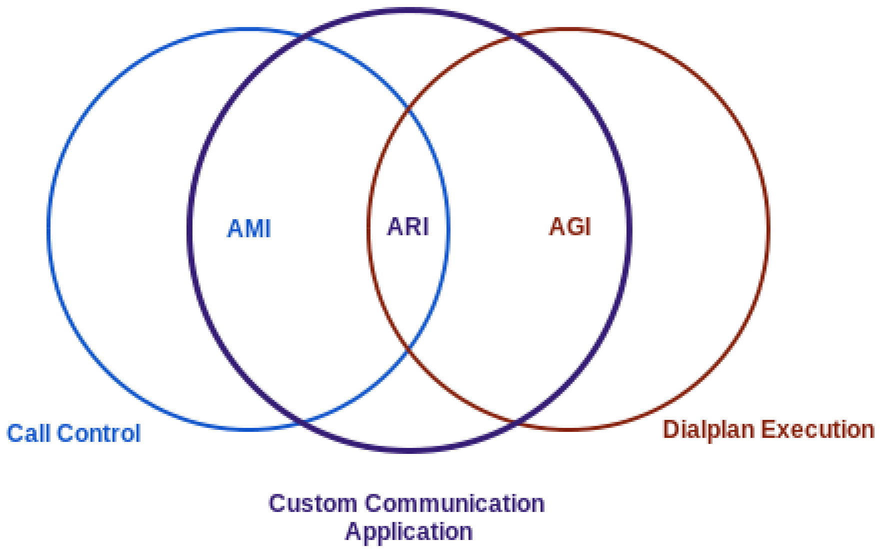
Action: Originate
Channel: PJSIP/103
Exten: 200
Context: from-internal
Priority: 1

Action: Logoff

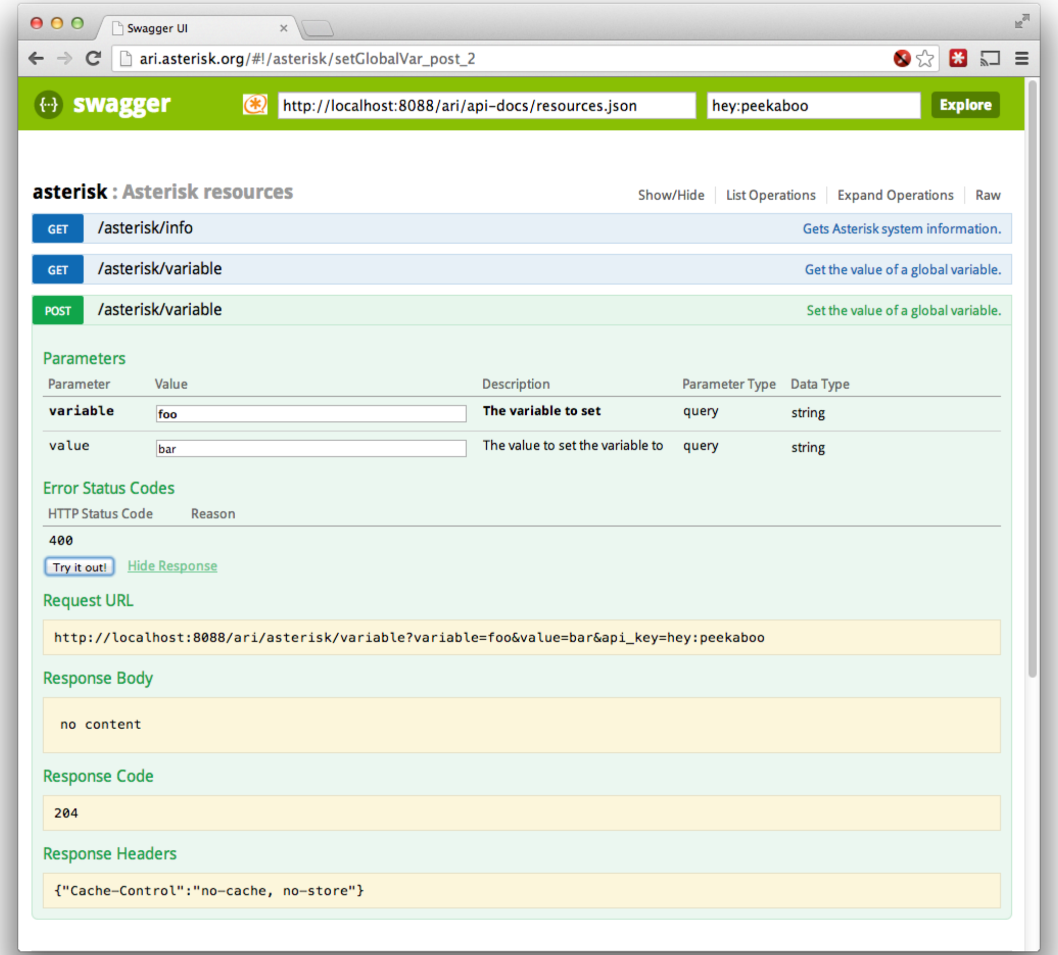


ARI Asterisk Rest Interface

- DialPlan > AMI and AGI > ARI



ARI Ayarları



Sınıf Uygulaması Kurulum

Sınıf Uygulaması Geliştirici

Teşekkürler.

KoçSistem
GELECEĞİ KEŞFET