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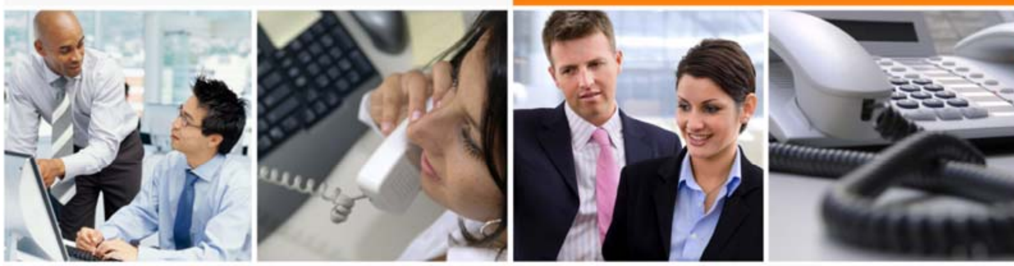
XO® SIP Service
Customer Configuration Guide
Digium Asterisk Appliance 50 (AA50)
versions 1.0.3.1 and 1.1.1



XO Communications

Table of Contents

1. Overview

About This Document

This document describes XO Communications SIP package 1 minimal configuration requirements, consisting of:

- Digium Asterisk Appliance 50 (AA50) software version 1.0.3.1 and
- Digium Asterisk Appliance 50 (AA50) software version1.1.1

both deployed with an XO-provided Cisco 2432 IAD as the router/demarcation device. This document assumes the audience has a general understanding of network provisioning, connectivity requirements of XO Communications SIP service offering.

Known Issues

While XO certifies interoperability between XO SIP service and the PBX as outlined herein, the following known issues were identified during Interoperability testing. The customer should be aware that certain features and functions may not be fully supportable. In some cases, special configurations and/or PBX software patches may be available from the vendor:

- Digium Asterisk Appliance 50 (AA50) software version 1.0.3.1:
  - **Inbound CNAM displays as "New User", rather than actual name**
- Digium Asterisk Appliance software version 1.1.1:
  - **Blind Call Transfer failed**
  - **Inbound CNAM displays as "New User", rather than actual name**

Registration Method

Digium Asterisk Appliance 50 (AA50) software versions 1.0.3.1 and 1.1.1 utilize static registration between IP phones and the IP PBX.

XO SIP Service Packages Supported

Digium Asterisk Appliance 50 (AA50) software versions 1.0.3.1 and 1.1.1 support XO SIP Service Package 1 only:

Pkg	Codec	DTMF	Fax
1	G.711	RFC2833 (in-band RTP DTMF fallback)	T.38; G.711 pass-through

XO SIP package 2 (G.729a compression) is <u>not</u> supported

2. Testing of Digium Asterisk Appliance IP PBX, Software Version 1.0.3.1

2.1 Software and Hardware Versions Tested

Cisco 2432 IAD CPE Used as a Router
Cisco 2432 Software Version: Cisco IOS Software, 2400 Software (C2430-IS-M), Version 12.4(7e), RELEASE SOFTWARE (fc5)
System image file is "flash:c2430-is-mz.124-7e.bin"
Cisco IAD2432 (R527x) processor (revision 4.1) with 119808K/11264K bytes of memory.
Processor board ID FHK1012F04P
R527x CPU at 225MHz, Implementation 40, Rev 3.1
DRAM configuration is 64 bits wide with parity disabled.
63K bytes of non-volatile configuration memory.
62720K bytes of ATA System CompactFlash (Read/Write)
0K bytes of ATA Slot0 CompactFlash (Read/Write)
Slot 0: C2400 Mother board 24FXS-2F-2FE (3 onboard DSPs) Port adapter, 33 ports
Version Identifier : V01
WIC Slot 0:
T1 (2 port) Multi-Flex Trunk WAN daughter card
Hardware revision 1.0 Board revision B0

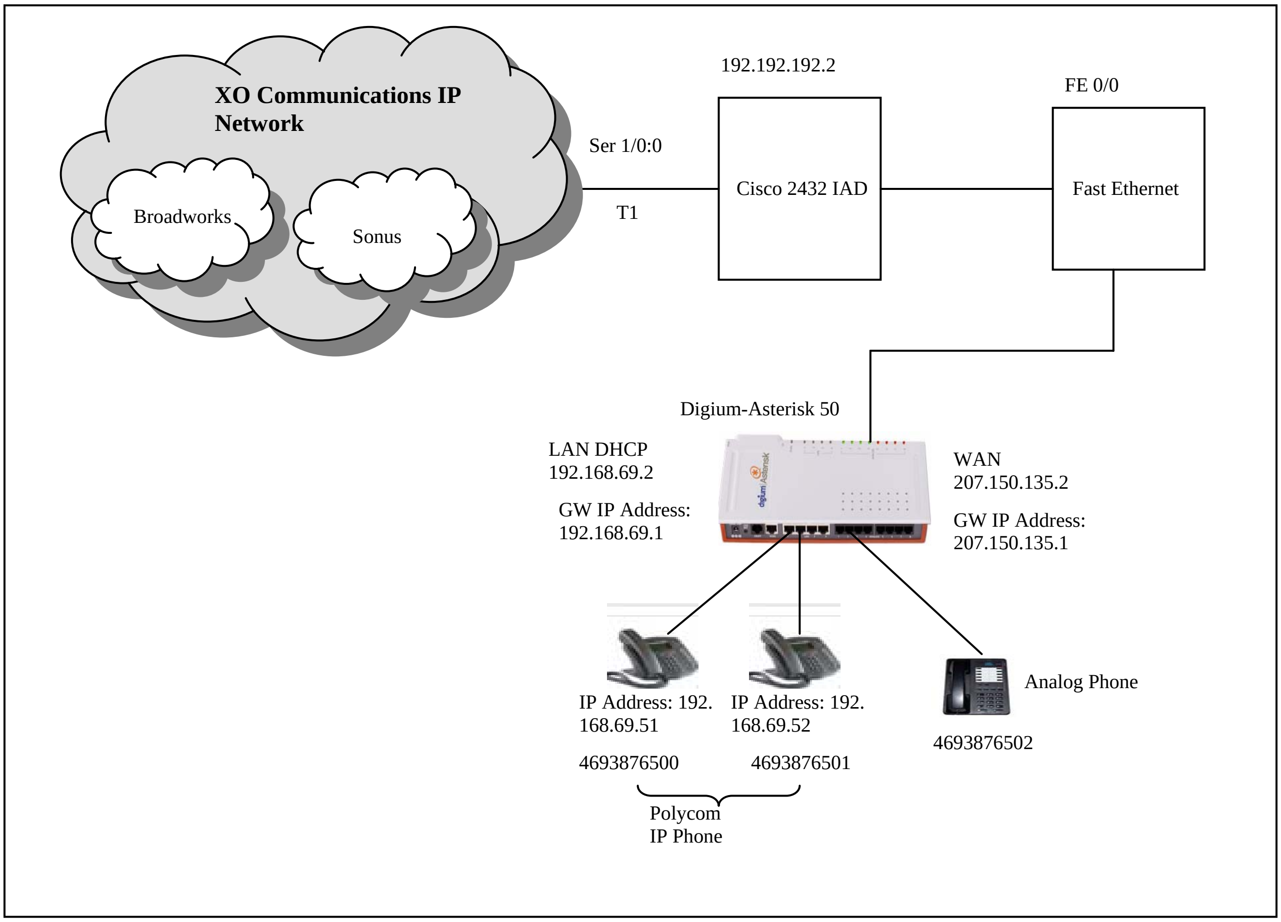
Digium Asterisk Appliance 50 (AA50) 50
Hardware No: S844i Rev: B1
The software: OS Version: Linux asteriskpbx 2.6.16.27sx00i-1.0.3.1
Asterisk Build: Asterisk Autotag for _sx00i-1.0.3

Polycom SIP Phone and Analog Phone
Polycom IP 430 release 2.2.1 revision C
LAN Port Mode: 100FD

2.2 Diagram of Lab Test Set-Up for Digium Asterisk Appliance IP PBX, Software Version 1.0.3.1

XO IP Network

The following diagram is used during lab testing to depict the configuration of Digium Asterisk Appliance 50 (AA50) deployed with Cisco2432 IAD CPE.



Digium Asterisk Appliance 50 (AA50) Deployed with Cisco 2432 IAD CPE

2.3 Digium-Asterisk Configuration for Software Version 1.0.3.1

In This Section

This section contains GUI and dump configuration on phone provisioning, service provider, users, voicemail, and SIP configuration.

Show Admin Settings

Show the Admin Settings. The **Admin Options** screen, illustrated in Figure 1, shows the Admin Settings.

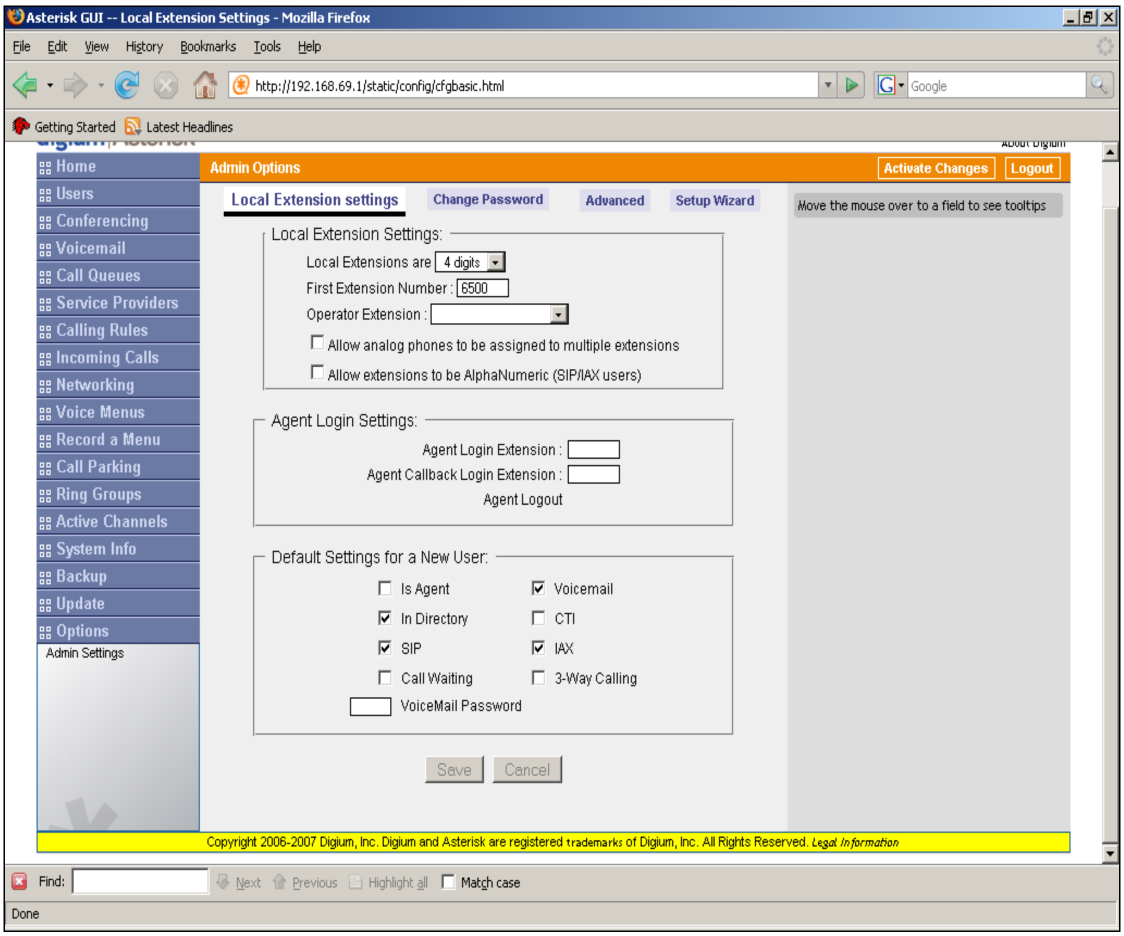


Figure 1. Admin Options Screen

▶

2.3 Digium-Asterisk Configuration *(continued)*

Verify Analog Port

Verify the analog port on the system. The Digium Asterisk Appliance 50 (AA50) has up to 8 **Analog Ports**, as illustrated in Figure 2.

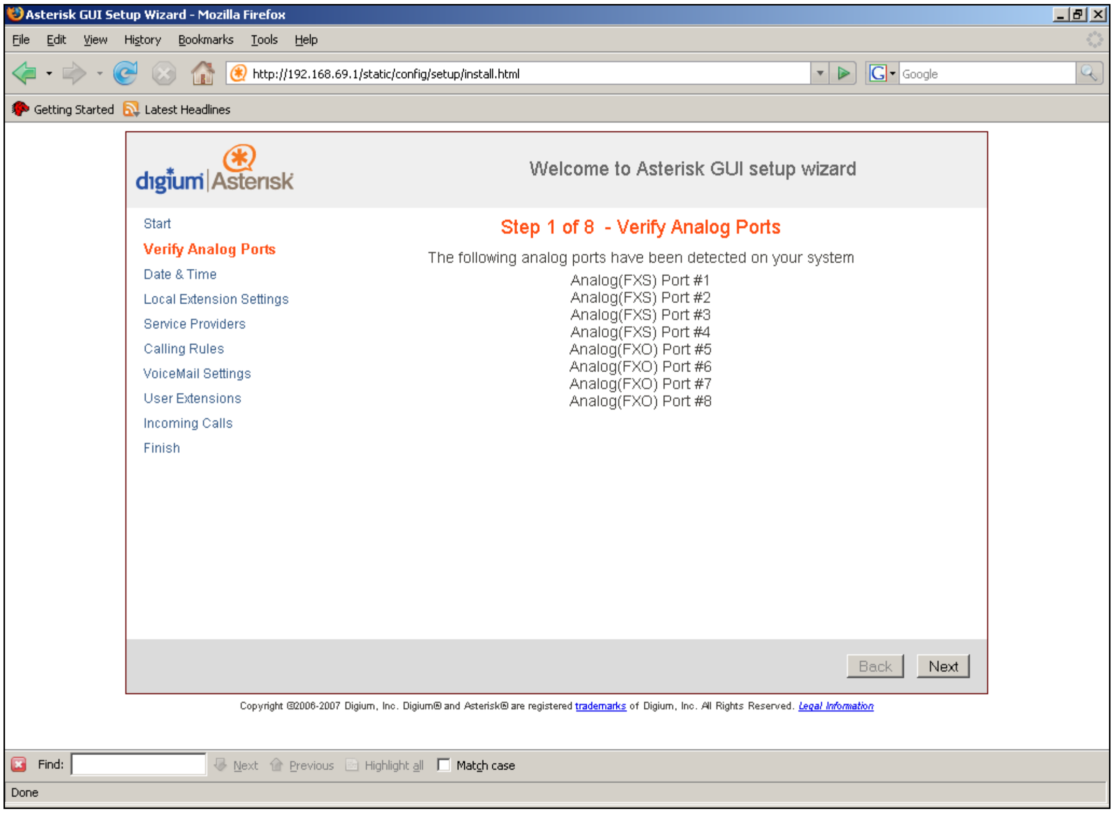


Figure 2. Verify Analog Ports Screen

2.3 Digium-Asterisk Configuration *(continued)*

Configure Date & Time Settings

Show the configuration of the current local system **date and time**, as illustrated in Figure 3.

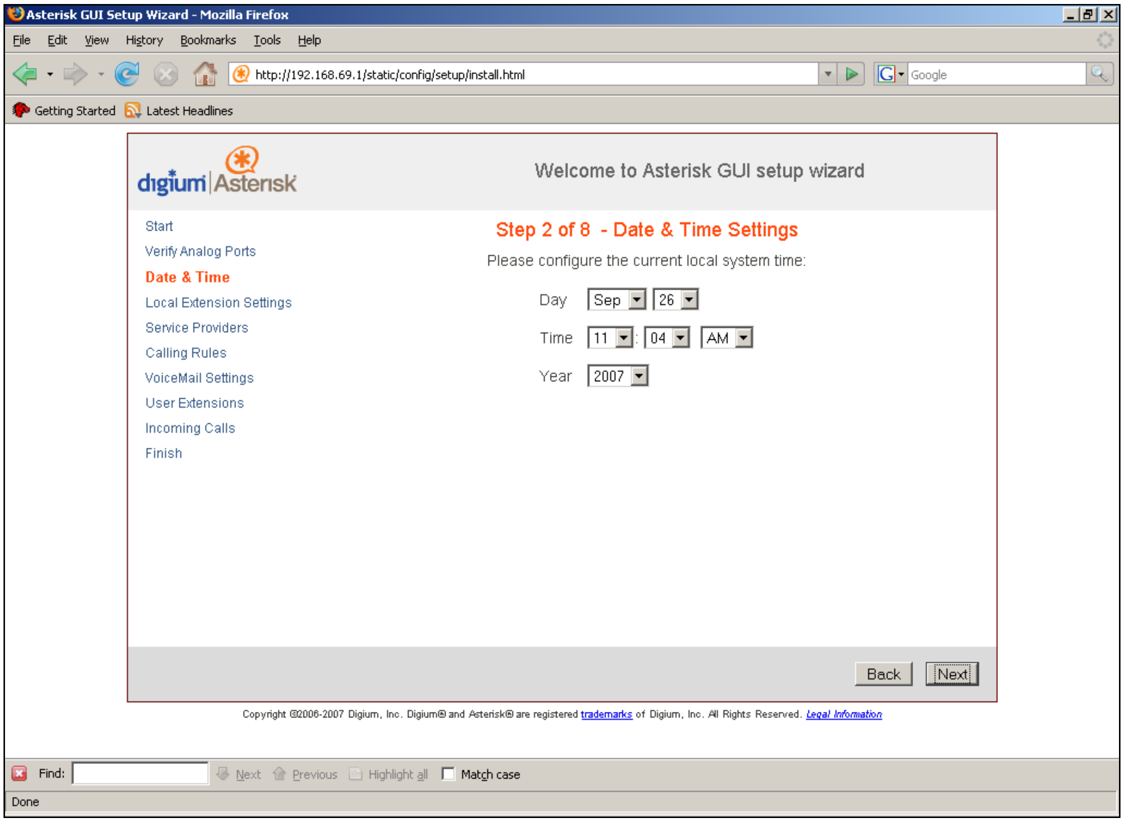


Figure 3. Date & Time Settings Screen

▶

2.3 Digium-Asterisk Configuration *(continued)*

Configure Local Extension Settings

Show the **Local Extension Settings**, as illustrated in Figure 4. The local extension is set to use a 4-digit extension.

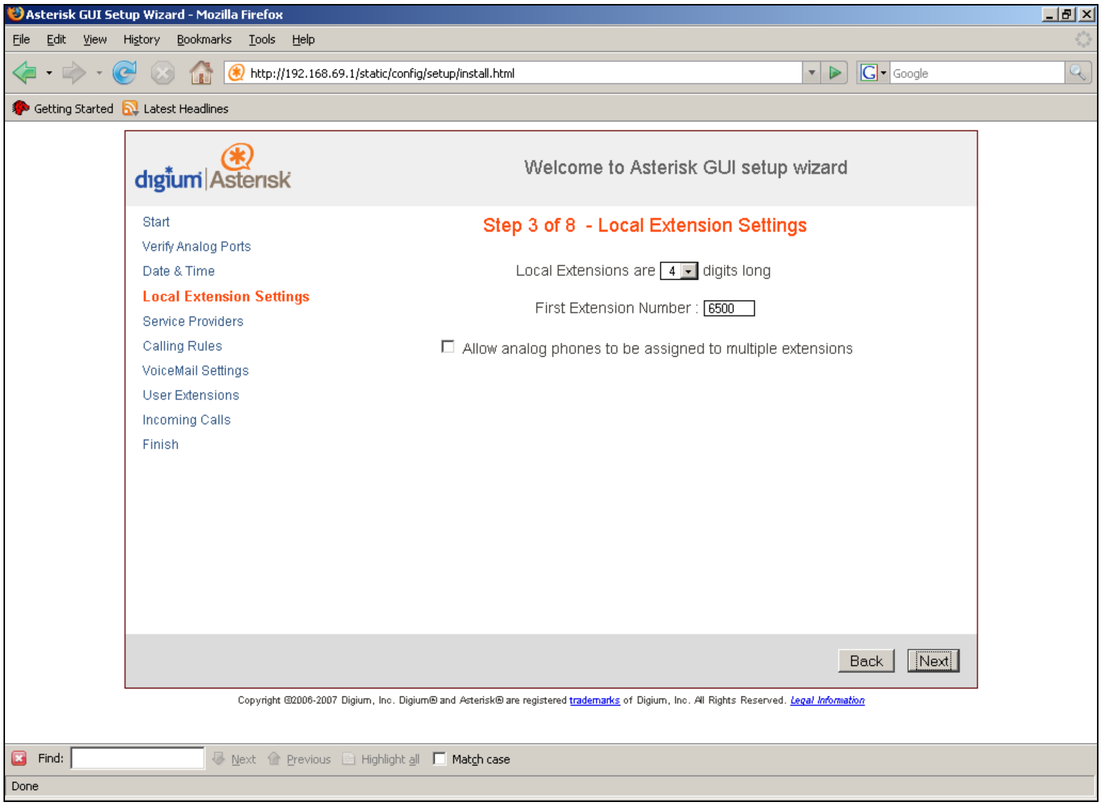


Figure 4. Local Extension Settings Screen

▶

2.3 Digium-Asterisk Configuration *(continued)*

Show Service Provider

Show the configuration of the **Service Provider**. The Digium-Asterisk uses G711 ulaw, as illustrated in Figure 5.

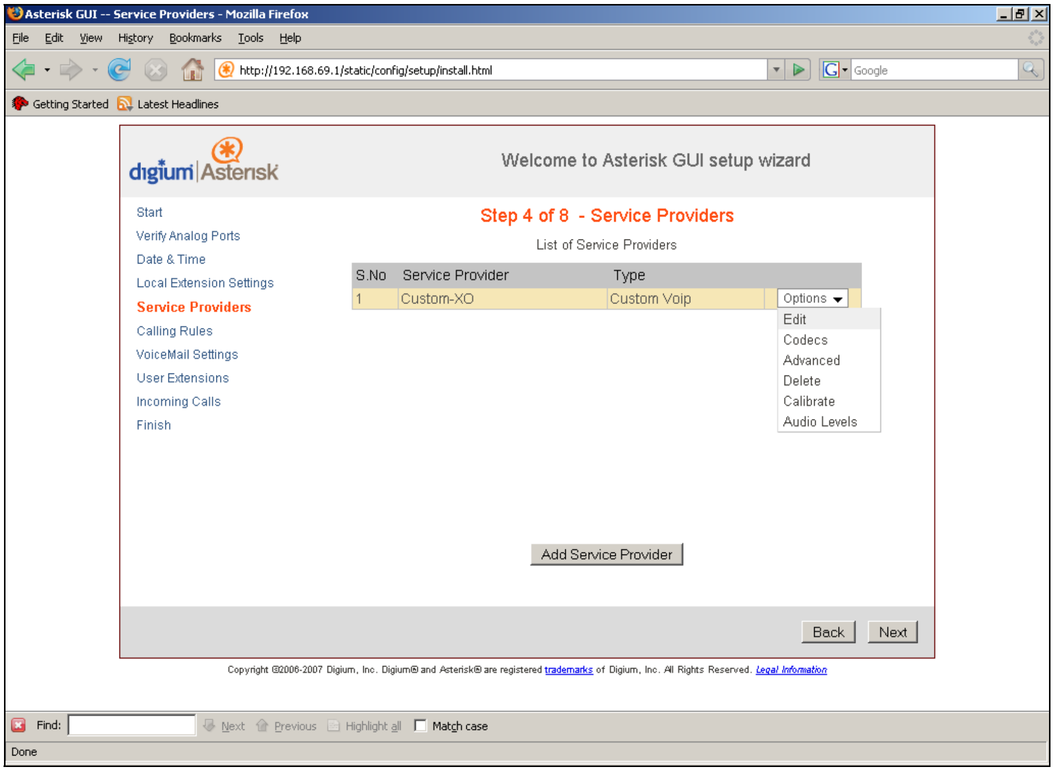


Figure 5. List of Service Providers

2.3 Digium-Asterisk Configuration *(continued)*

Edit Service Provider Information

The **Edit Service Provider** screen is illustrated in Figure 6. The SIP *Protocol* is used, and the *Register* box is not checked since it is static registration.

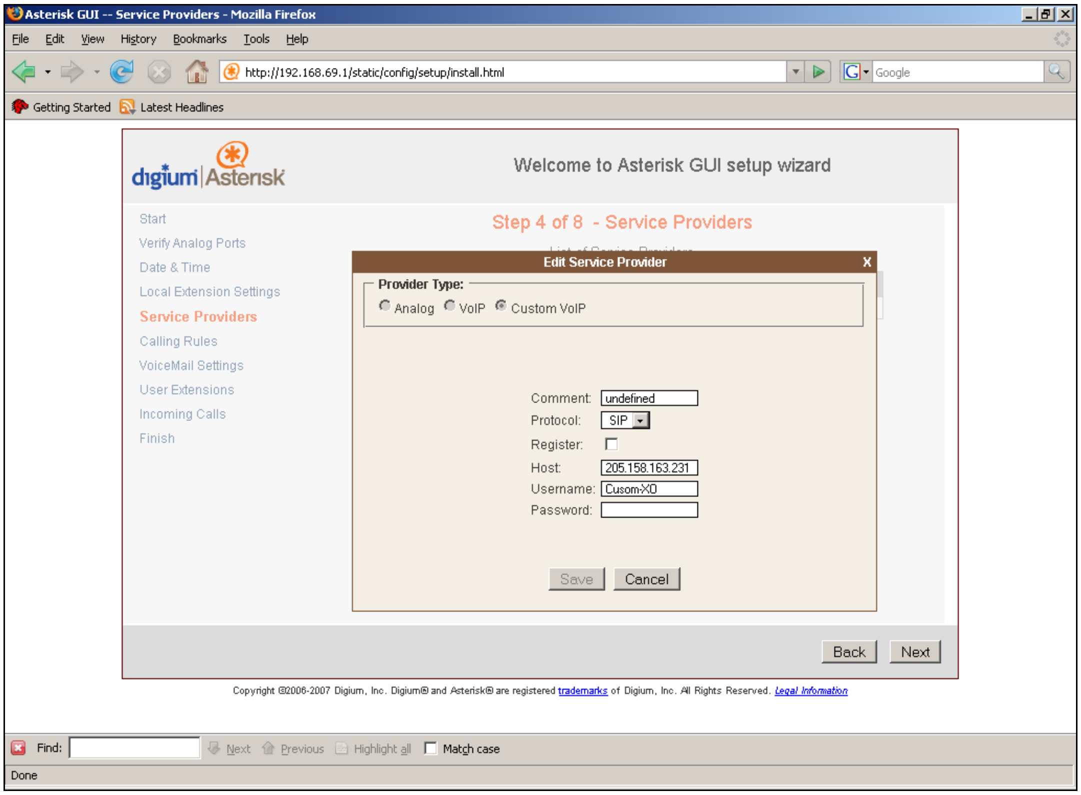


Figure 6. Edit Service Provider Screen

2.3 Digium-Asterisk Configuration *(continued)*

Show Outbound Calling Rules

Configuration for **Outbound Calling Rules** is shown in Figure 7. Local numbers are begun with 9 and long distance with a 91.

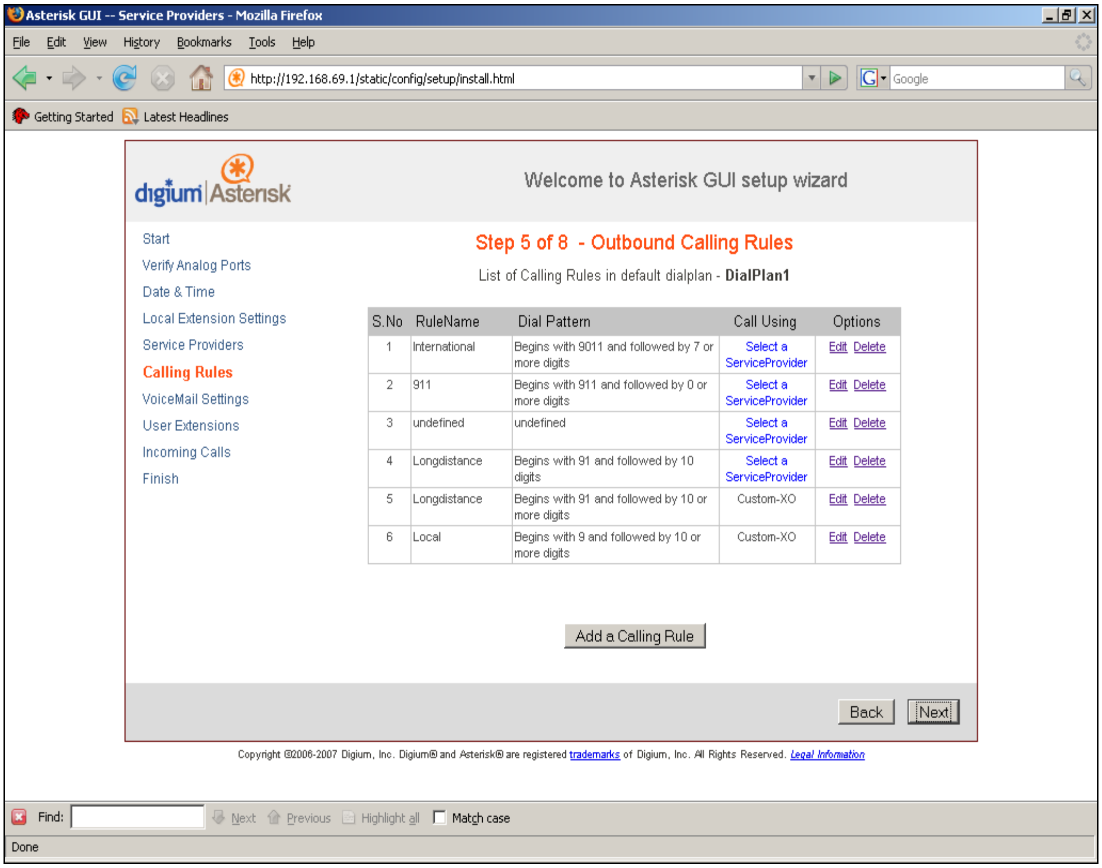


Figure 7. Outbound Calling Rules Screen

▶

2.3 Digium-Asterisk Configuration *(continued)*

Configure Voicemail Settings

Figure 8 shows the configuration of **VoiceMail Settings**. The *Attach recordings to email* field is selected. *Maximum messages per folder* and *maximum message time* can be selected in the drop-down box.

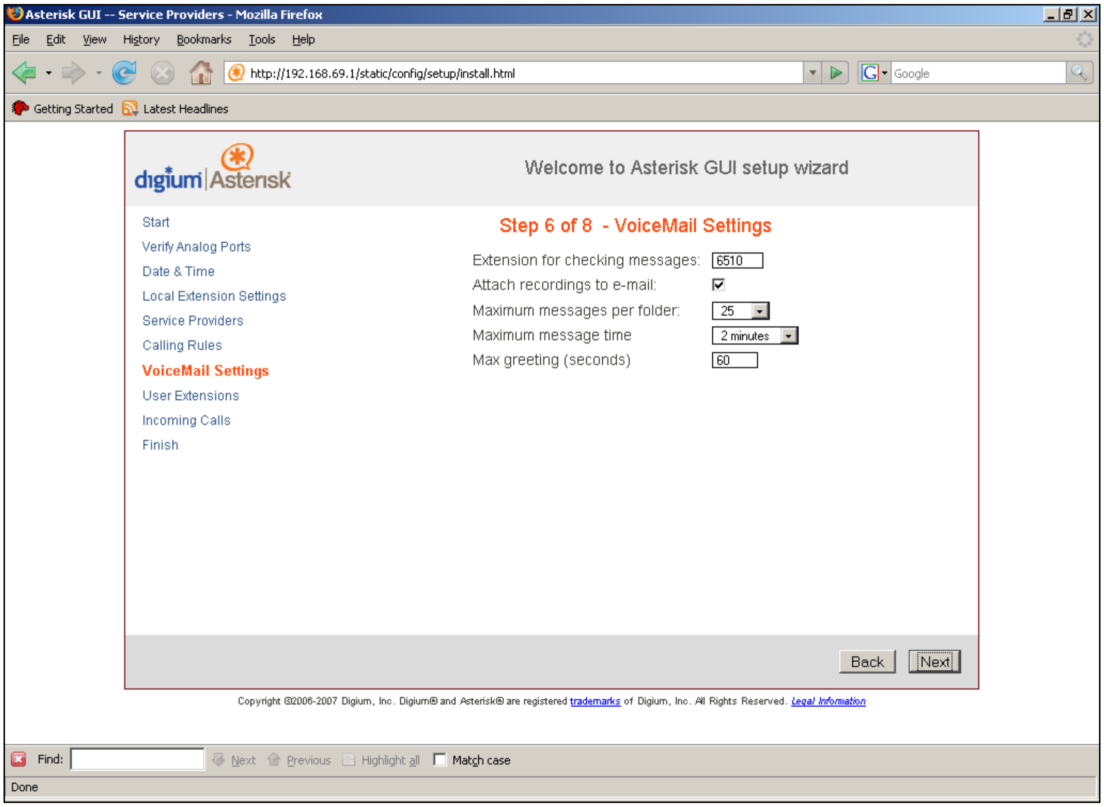


Figure 8. VoiceMail Settings Screen

▶

2.3 Digium-Asterisk Configuration *(continued)*

Configure User Extensions

Figure 9 show configuration of the **User Extensions**. Each user can be edited in the *Options* column.

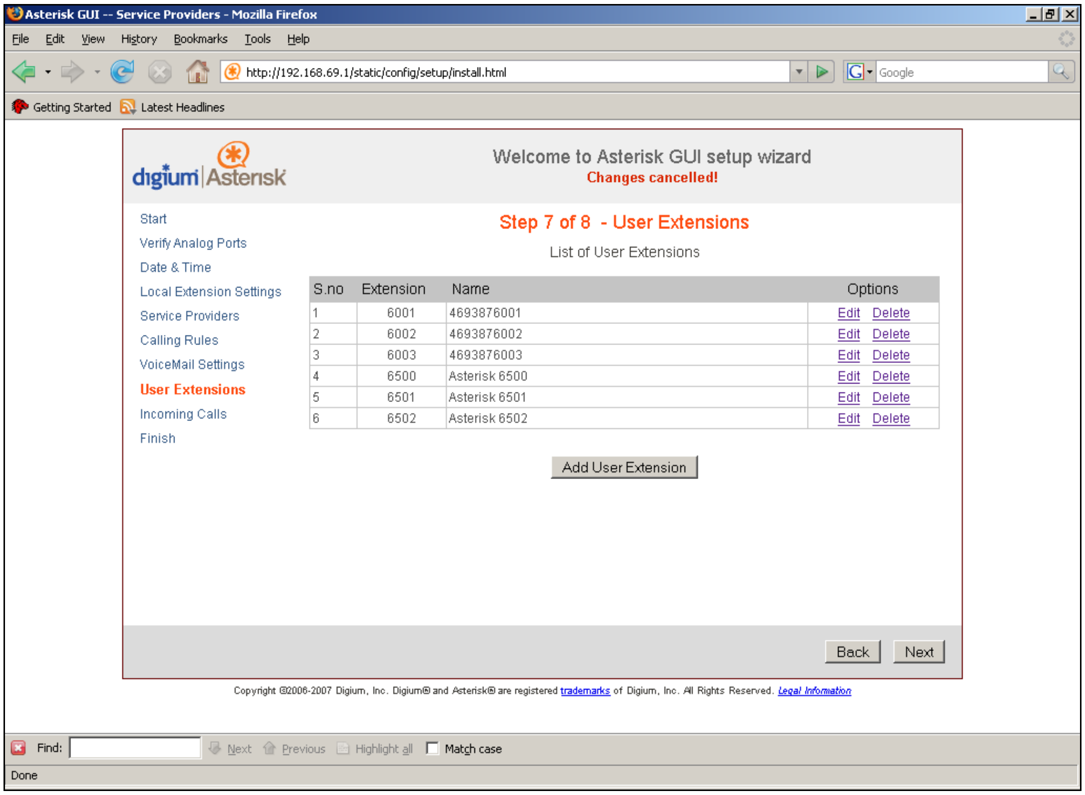


Figure 9. User Extensions Screen

▶

2.3 Digium-Asterisk Configuration *(continued)*

Edit User Information

Figure 10 shows how to edit **user information.** The *Dial Plan* is selected using the user extension. The *Phone Serial* is in the back of Polycom SIP Phone.

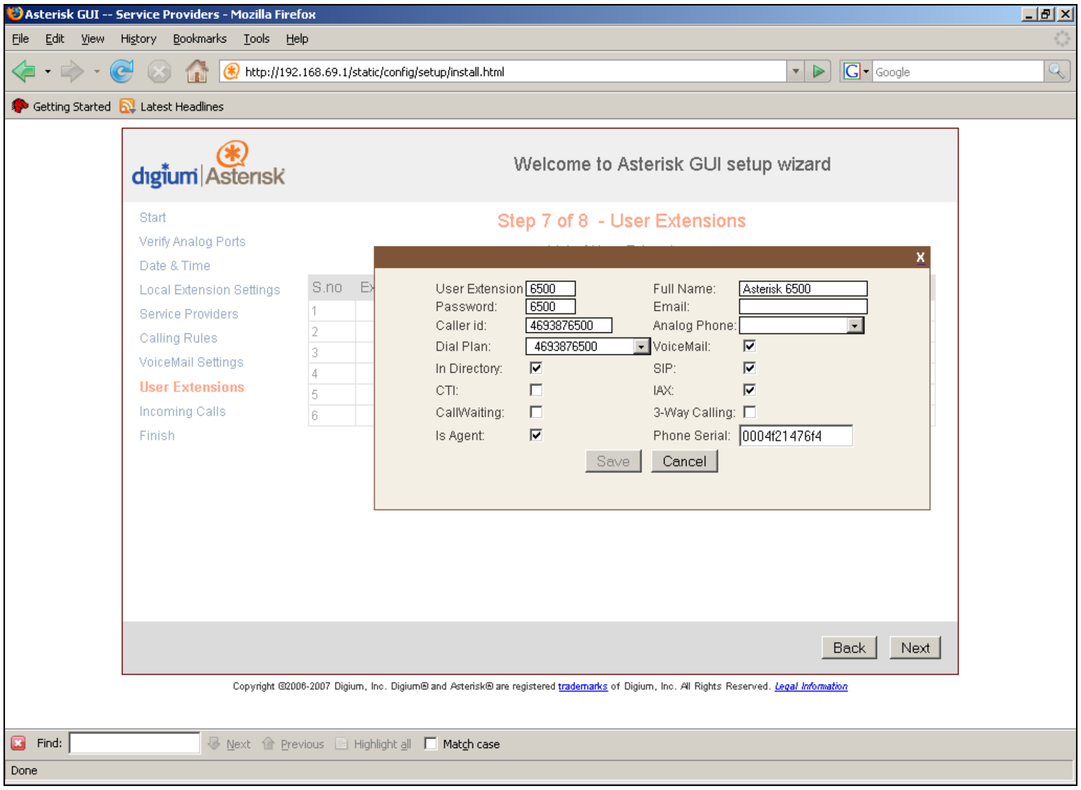


Figure 10. User Extensions Screen

▶

2.3 Digium-Asterisk Configuration *(continued)*

Show Incoming Call Rules

Figure 11 show the configuration of the **Incoming Calls** for each user extension. Each user extension is configured to have its own rule for incoming call.

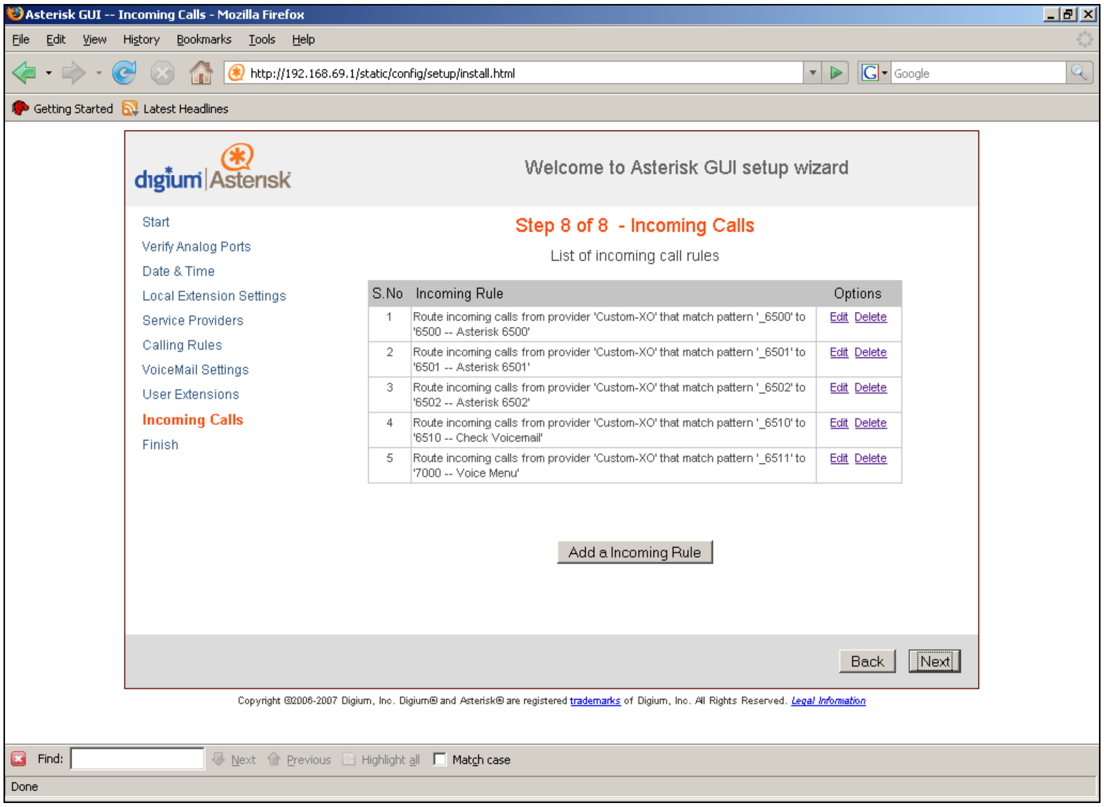


Figure 11. Incoming Calls Screen

▶

2.3 Digium-Asterisk Configuration *(continued)*

Configure Auto Attendant

Figure 12 show the configuration of the **Voice Menus / Auto Attendant**. The new step in the greeting menu can be added in the *Add a new Step* drop-down box.

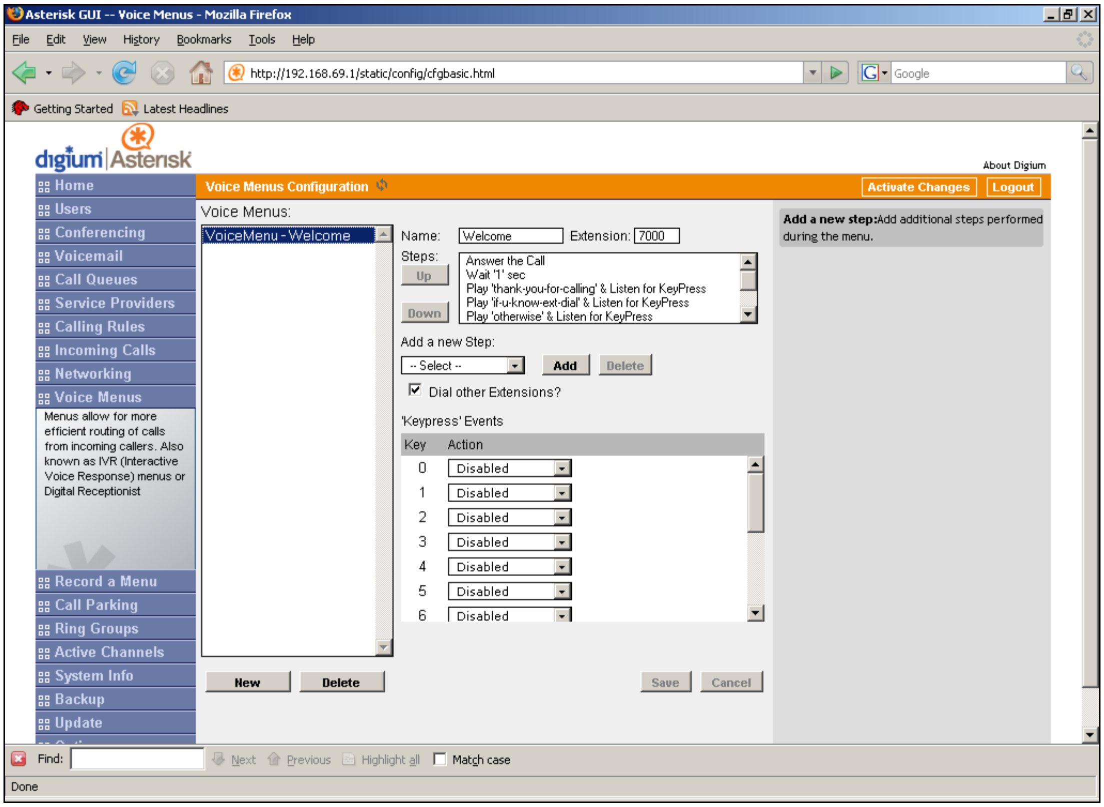


Figure 12. Auto Attendant Configuration Screen

▶

2.3 Digium-Asterisk Configuration *(continued)*

Configure Hunt Group

Figure 13 shows the configuration of the **Ring / Hunt Group**. The users are added into the *Add Ring Group*. It has two *Strategies* in the drop-down box:

- *Ring in Order* – Will ring the first agent in first list; if it's not answered in 4 rings, it will ring the second user.
- *Ring All* option – All the users in the Add Ring Group will ring simultaneously.

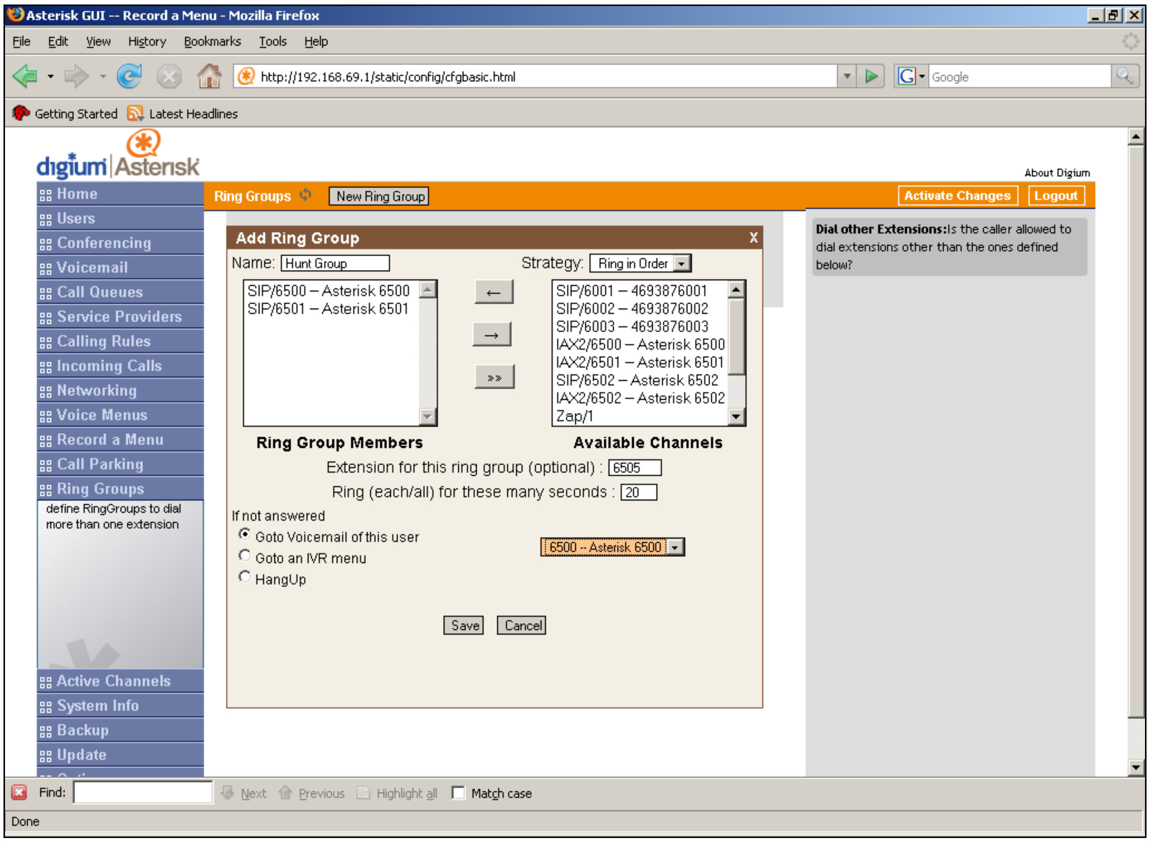


Figure 13. Hunt Group Screen

▶

2.3 Digium-Asterisk Configuration *(continued)*

Provisioning Dumps on SIP Configuration

This is the automatically generated configuration file in Digium appliance. This file will contain all the default SIP configuration information.

```
;! Filename: sip.conf (/etc/asterisk/sip.conf)
;! Generator: Manager
;! Creation Date: Tue Sep 11 19:48:47 2007
;!
[general]
context = default  ; Default context for incoming calls
;allowguest=no                  ; Allow or reject guest calls (default is yes, t
his can also be set to 'osp'
; if asterisk was compiled with OSP support.
;realm=mydomain.tld             ; Realm for digest authentication
; defaults to "asterisk"
; Realms MUST be globally unique according to RFC 3261
; Set this to your host name or domain name
bindport = 5060  ; UDP Port to bind to (SIP standard port is 5060)
bindaddr = 0.0.0.0  ; IP address to bind to (0.0.0.0 binds to all)
srvlookup = yes  ; Enable DNS SRV lookups on outbound calls
allowexternaldomains = no
allowexternalinvites = no
allowguest = yes
allowoverlap = yes
allowsubscribe = no
allowtransfer = yes
alwaysauthreject = no
autodomain = no
callevents = no
compactheaders = no
dumphistory = yes
g726nonstandard = no
ignoreregexpire = no
jbenable = no
jbforce = no
jblog = no
maxcallbitrate = 384
maxexpiry = 3600
minexpiry = 60
notifyringing = no
pedantic = no
promiscredir = no
```

▶

2.3 Digium-Asterisk Configuration *(continued)*

Provisioning Dumps on SIP Configuration (continued)

```
recordhistory = yes
relaxdtmf = no
rtcachefriends = no
rtsavesysname = no
rtupdate = no
sendrpid = no
sipdebug = yes
t1min = 100
t38pt_udptl = no
progressinband = no
trustrpid = no
usereqphone = no
videosupport = no
; Note: Asterisk only uses the first host
; in SRV records
; Disabling DNS SRV lookups disables the
; ability to place SIP calls based on domain
; names to some other SIP users on the Internet
;domain=mydomain.tld          ; Set default domain for this host
; If configured, Asterisk will only allow
; INVITE and REFER to non-local domains
; Use "sip show domains" to list local domains
;domain=mydomain.tld,mydomain-incoming
; Add domain and configure incoming context
; for external calls to this domain
;domain=1.2.3.4               ; Add IP address as local domain
;
```

This section contains the registration information.

```
;allowexternalinvites=no       ; Disable INVITE and REFER to non-local domains
; Default is yes
;autodomain=yes                ; Turn this on to have Asterisk add local host
; name and local IP to domain list.
;pedantic=yes                  ; Enable slow, pedantic checking for Pingtel
; and multiline formatted headers for strict
; SIP compatibility (defaults to "no")
;tos=184                       ; Set IP QoS to either a keyword or numeric val
;tos=lowdelay                  ; lowdelay,throughput,reliability,mincost,none
;maxexpiry=3600                ; Max length of incoming registration we allow
;defaultexpiry=120             ; Default length of incoming/outoing registratio
n
```

2.3 Digium-Asterisk Configuration *(continued)*

Provisioning Dumps on SIP Configuration (continued)

```
;notifymimetype=text/plain      ; Allow overriding of mime type in MWI NOTIFY
;checkmwi=10                ; Default time between mailbox checks for peers
;vmexten=voicemail      ; dialplan extension to reach mailbox sets the
; Message-Account in the MWI notify message
; defaults to "asterisk"
;videosupport=yes               ; Turn on support for SIP video
;recordhistory=yes              ; Record SIP history by default
; (see sip history / sip no history)
;disallow=all                   ; First disallow all codecs
;allow=ulaw                     ; Allow codecs in order of preference
;allow=ilbc                     ;
;musicclass=default             ; Sets the default music on hold class for all S
IP calls
; This may also be set for individual users/peers
;language=en                    ; Default language setting for all users/peers
; This may also be set for individual users/peers
;relaxdtmf=yes                  ; Relax dtmf handling
;rtptimeout=60                  ; Terminate call if 60 seconds of no RTP activit
y
; when we're not on hold
;rtpholdtimeout=300             ; Terminate call if 300 seconds of no RTP activi
ty
; when we're on hold (must be > rtptimeout)
;trustrpid = no                 ; If Remote-Party-ID should be trusted
;sendrpid = yes                 ; If Remote-Party-ID should be sent
;progressinband=never           ; If we should generate in-band ringing always
; use 'never' to never use in-band signalling, even in cases
; where some buggy devices might not render it
;useragent=Asterisk PBX         ; Allows you to change the user agent string
;promiscredir = no              ; If yes, allows 302 or REDIR to non-local SIP a
ddress
; Note that promiscredir when redirects are made to the
; local system will cause loops since SIP is incapable
; of performing a "hairpin" call.
;usereqphone = no               ; If yes, ";user=phone" is added to uri that con
tains
; a valid phone number
```

2.3 Digium-Asterisk Configuration *(continued)*

Provisioning Dumps on SIP Configuration (continued)

This section contains the DTMF information on Digium.

```
;dtmfmode = rfc2833            ; Set default dtmfmode for sending DTMF. Default
: rfc2833
; Other options:
; info : SIP INFO messages
; inband : Inband audio (requires 64 kbit codec -alaw, ulaw)
; auto : Use rfc2833 if offered, inband otherwise
;compactheaders = yes          ; send compact sip headers.
;sipdebug = yes                ; Turn on SIP debugging by default, from
; the moment the channel loads this configuration
;subscribecontext = default    ; Set a specific context for SUBSCRIBE requests
; Useful to limit subscriptions to local extensions
; Settable per peer/user also
;notifyringing = yes           ; Notify subscriptions on RINGING state
```

Provisioning Dumps on User Configuration

```
;! Automatically generated configuration file
;! Filename: users.conf (/etc/asterisk/users.conf)
;! Generator: Manager
;! Creation Date: Wed Sep 19 15:01:53 2007
;!
[general]
trunkstyle = voip
fullname = New User
phone = none
```

▶

2.3 Digium-Asterisk Configuration *(continued)*

Provisioning Dumps on User Configuration *(continued)*

This section contains the configuration file of user extension 6500.

```
userbase = 6500
hasvoicemail = yes
hasdirectory = yes
hassip = yes
hasiax = yes
hasmanager = no
callwaiting = no
threewaycalling = no
localextenlength = 4
switchtype = national
usecallerid = yes
hidecallerid = no
usecallingpres = yes
canpark = yes
cancallforward = yes
callreturn = yes
echocancel = yes
echocancelwhenbridged = yes
rxgain = 0.0
txgain = 0.0
immediate = no
callwaitingcallerid = yes
transfer = yes
allow_aliasextns = no
allow_an_extns = no
hasagent = no

[6500]
callwaiting = no
cid_number = 4693876500
context = numberplan-custom-1
fullname = Asterisk 6500
signalling = fxo_ks
group =
hasagent = yes
hasdirectory = yes
hasiax = yes
hasmanager = no
hassip = yes
hasvoicemail = yes
host = dynamic
```

▶

2.3 Digium-Asterisk Configuration *(continued)*

Provisioning Dumps on User Configuration (continued)

```
mailbox = 6500
secret = 6500
threewaycalling = no
vmsecret = 1234
registeriax = yes
registersip = yes
autoprov = yes
canreinvite = no
nat = no
dtmfmode = rfc2833
macaddress = 0004f21476f4
label = 6500
```

This section contains the configuration file of user extension 6501.

```
[6501]
callwaiting = no
cid_number = 4693876501
context = numberplan-custom-2
fullname = Asterisk 6501
signalling = fxo_ks
group =
hasagent = yes
hasdirectory = yes
hasiax = yes
hasmanager = no
hassip = yes
hasvoicemail = yes
host = dynamic
mailbox = 6501
secret = 6501
threewaycalling = no
vmsecret = 1234
registeriax = yes
registersip = yes
macaddress = 0004f21476c8
autoprov = yes
label = 6501
canreinvite = no
nat = no
dtmfmode = rfc2833
```

2.3 Digium-Asterisk Configuration *(continued)*

Provisioning Dumps on User Configuration
(continued)

This section contains the configuration file of Digium Trunk Group.

```
[Digium_Trk1]
allow = all
context = DID_Digium_Trk1
dialformat = ${EXTEN:1}
hasexten = no
hasiax = no
hassip = yes
host = 205.158.163.231
port = 5060
registeriax = no
registersip = no
trunkname = Custom-XO
trunkstyle = customvoip
username = Cusom-XO
fromuser = 4693876500
```

This section contains the configuration file of user extension 6502.

```
[6502]
callwaiting = no
cid_number = 4693876502
fullname = Asterisk 6502
  as agent = fxo_ks
rxgain = 0.0
txgain = 0.0
group =
  as agent = yes
hasdirectory = yes
hasiax = yes
hasmanager = no
hassip = yes
hasvoicemail = yes
host = dynamic
mailbox = 6502
secret = 6502
threewaycalling = no
zapchan = 1
registeriax = yes
registersip = yes
autoprov = no
canreinvite = no
nat = no
dtmfmode = rfc2833
```

2.3 Digium-Asterisk Configuration *(continued)*

Provisioning Dumps on Phone Configuration

This section will contain the Polycom SIP phone configuration file of the:

```
[general]
;serveraddr=192.168.1.1 ; Address to send to the phone to use as server address.
serveriface=eth1       ; Same as above, except an ethernet interface.  Useful f
or when the interface uses DHCP.
                       ; There is no default for either of the above, and only
one should be set.
;serverport=5060       ; Port to send to the phone to use as server port.  Defa
ult is 5060.
```

Dumps on Voicemail Configuration

This section contains the default configuration on Digium Voicemail, and how to configure the optional parts on the Voicemail.

```
;! Automatically generated configuration file
;! Filename: voicemail.conf (/etc/asterisk/voicemail.conf)
;! Generator: Manager
;! Creation Date: Mon Aug 27 21:42:27 2007
;!
;
; Voicemail Configuration
;
;
; NOTE: Asterisk has to edit this file to change a user's password.  This does
; note currently work with the "#include <file>" directive for Asterisk
; configuration files.  Do not use it with this configuration file.
;
[general]
; Default formats for writing Voicemail
;format=g723sf|wav49|wav
format = ulaw
;
; WARNING:
; If you change the list of formats that you record voicemail in
; when you have mailboxes that contain messages, you _MUST_ absolutely
; manually go through those mailboxes and convert/delete/add the
; the message files so that they appear to have been stored using
; your new format list. If you don't do this, very unpleasant
; things may happen to your users while they are retrieving and
; manipulating their voicemail.
;
; In other words: don't change the format list on a production system
; unless you are _VERY_  sure that you know what you are doing and are
; prepared for the consequences.
;
```

2.3 Digium-Asterisk Configuration *(continued)*

Dumps on Voicemail Configuration (continued)

```
; Who the e-mail notification should appear to come from
serveremail = asterisk
;serveremail=asterisk@linux-support.net
; Should the email contain the voicemail as an attachment
attach = yes
maxmsg = 25
maxmessage = 120
; Minimum length of a voicemail message in seconds for the message to be kept
; The default is no minimum.
minmessage = 0
; Maximum length of greetings in seconds
;maxgreet=60
; How many miliseconds to skip forward/back when rew/ff in message playback
skipms = 3000
; How many seconds of silence before we end the recording
maxsilence = 10
; Silence threshold (what we consider silence, the lower, the more sensitive)
silencethreshold = 128
; Max number of failed login attempts
maxlogins = 3
; If you need to have an external program, i.e. /usr/bin/myapp called when a
; voicemail is left, delivered, or your voicemailbox is checked, uncomment
; this:
;externnotify=/usr/bin/myapp
; If you need to have an external program, i.e. /usr/bin/myapp called when a
; voicemail password is changed, uncomment this:
;externpass=/usr/bin/myapp
; For the directory, you can override the intro file if you want
;directoryintro=dir-intro
; The character set for voicemail messages can be specified here
;charset=ISO-8859-1
; The ADSI feature descriptor number to download to
;adsifdn=0000000F
; The ADSI security lock code
;adsisec=9BDBF7AC
; The ADSI voicemail application version number.
;adsiver=1
; Skip the "[PBX]:" string from the message title
;pbxskip=yes
; Change the From: string
;fromstring=The Asterisk PBX
; Permit finding entries for forward/compose from the directory
;usedirectory=yes
;
```

▶

2.3 Digium-Asterisk Configuration *(continued)*

Dumps on Voicemail Configuration (continued)

```
; Change the from, body and/or subject, variables:
;     VM_NAME, VM_DUR, VM_MSGNUM, VM_MAILBOX, VM_CALLERID,
VM_CIDNUM,
;     VM_CIDNAME, VM_DATE
;
; Note: The emailbody config row can only be up to 512 characters due to a
;       limitation in the Asterisk configuration subsystem.
;emailsubject=[PBX]: New message ${VM_MSGNUM} in mailbox ${VM_MAILBOX}
; The following definition is very close to the default, but the default shows
; just the CIDNAME, if it is not null, otherise just the CIDNUM, or "an unknown
; caller", if they are both null.
;emailbody=Dear ${VM_NAME}:\n\n\tjust wanted to let you know you were just
left a ${VM_DUR} long message (number ${VM_MSGNUM})\nin mailbox
${VM_MAILBOX} from ${
VM_CALLERID}, on ${VM_DATE}, so you might\nwant to check it when you get
a chanc
e.  Thanks!\n\n\t\t\t\t--Asterisk\n
;
; You can also change the Pager From: string, the pager body and/or subject.
; The above defined variables also can be used here
;pagerfromstring=The Asterisk PBX
;pagersubject=New VM
;pagerbody=New ${VM_DUR} long msg in box ${VM_MAILBOX}\nfrom
${VM_CALLERID}, on
${VM_DATE}
;
; Set the date format on outgoing mails. Valid arguments can be found on the
; strftime(3) man page
;
; Default
emaildateformat = %A, %B %d, %Y at %r
; 24h date format
;emaildateformat=%A, %d %B %Y at %H:%M:%S
;
; You can override the default program to send e-mail if you wish, too
;
mailcmd = /bin/ssmtp
;
; Users may be located in different timezones, or may have different
; message announcements for their introductory message when they enter
; the voicemail system. Set the message and the timezone each user
; hears here. Set the user into one of these zones with the tz= attribute
```

2.3 Digium-Asterisk Configuration *(continued)*

Dumps on Voicemail Configuration (continued)

```
; in the options field of the mailbox. Of course, language substitution
; still applies here so you may have several directory trees that have
; alternate language choices.
;
; Look in /usr/share/zoneinfo/ for names of timezones.
; Look at the manual page for strftime for a quick tutorial on how the
; variable substitution is done on the values below.
;
; Supported values:
; 'filename'    filename of a soundfile (single ticks around the filename
;               required)
; ${VAR}        variable substitution
; A or a        Day of week (Saturday, Sunday, ...)
; B or b or h   Month name (January, February, ...)
; d or e        numeric day of month (first, second, ..., thirty-first)
; Y             Year
; I or l        Hour, 12 hour clock
; H             Hour, 24 hour clock (single digit hours preceded by "oh")
; k             Hour, 24 hour clock (single digit hours NOT preceded by "oh")
; M             Minute, with 00 pronounced as "o'clock"
; N             Minute, with 00 pronounced as "hundred" (US military time)
; P or p        AM or PM
; Q             "today", "yesterday" or ABdY
;               (*note: not standard strftime value)
; q             "" (for today), "yesterday", weekday, or ABdY
;               (*note: not standard strftime value)
; R             24 hour time, including minute
;
;
;
; Each mailbox is listed in the form
<mailbox>=<password>,<name>,<email>,<pager_
email>,<options>
; if the e-mail is specified, a message will be sent when a message is
; received, to the given mailbox. If pager is specified, a message will be
; sent there as well. If the password is prefixed by '-', then it is
; considered to be unchangable.
;
; Advanced options example is extension 4069
; NOTE: All options can be expressed globally in the general section, and
; overriden in the per-mailbox settings, unless listed otherwise.
;
```

2.3 Digium-Asterisk Configuration *(continued)*

Dumps on Voicemail Configuration (continued)

```
; tz=central          ; Timezone from zonemessages above.  Irrelevant if envel
ope=no.
; attach=yes          ; Attach the voicemail to the notification email *NOT* t
he pager email
; saycid=yes          ; Say the caller id information before the message. If n
ot described,
;     or set to no, it will be in the envelope
; cidinternalcontexts=intern    ; Internal Context for Name Playback instead of
extension digits when saying caller id.
; sayduration=no      ; Turn on/off the duration information before the messag
e. [ON by default]
; saydurationm=2      ; Specify the minimum duration to say. Default is 2 minu
tes
; dialout=fromvm      ; Context to dial out from [option 4 from the advanced m
enu]
;     if not listed, dialing out will not be permitted
sendvoicemail = yes  ; Context to Send voicemail from [option 5 from the
advance
d menu]
maxgreet = 60
; if not listed, sending messages from inside voicemail will not be
; permitted
; searchcontexts=yes    ; Current default behavior is to search only the default
 context
; if one is not specified.  The older behavior was to search all contexts.
; This option restores the old behavior [DEFAULT=no]
; callback=fromvm       ; Context to call back from
;     if not listed, calling the sender back will not be permitted
; review=yes            ; Allow sender to review/rerecord their message before s
aving it [OFF by default
; operator=yes          ; Allow sender to hit 0 before/after/during  leaving a v
oicemail to
;     reach an operator  [OFF by default]
; envelope=no           ; Turn on/off envelope playback before message playback.
 [ON by default]
;     This does NOT affect option 3,3 from the advanced options menu
; delete=yes            ; After notification, the voicemail is deleted from the
server. [per-mailbox only]
;     This is intended for use with users who wish to receive their voicemail ON
LY by email.
; nextaftercmd=yes      ; Skips to the next message after hitting 7 or 9 to dele
te/save current message.
;     [global option only at this time]
; forcename=yes         ; Forces a new user to record their name.  A new user is
```

2.3 Digium-Asterisk Configuration *(continued)*

Dumps on Voicemail Configuration (continued)

```
;     determined by the password being the same as
;     the mailbox number.  The default is "no".
; forcegreetings=no     ; This is the same as forcename, except for recording
;     greetings.  The default is "no".
; hidefromdir=yes       ; Hide this mailbox from the directory produced by app_d
irectory
;     The default is "no".
[zonemessages]
eastern = America/New_York|'vm-received' Q 'digits/at' IMp
central = America/Chicago|'vm-received' Q 'digits/at' IMp
central24 = America/Chicago|'vm-received' q 'digits/at' H N 'hours'
military = Zulu|'vm-received' q 'digits/at' H N 'hours' 'phonetic/z_p'

[default]
; Define maximum number of messages per folder for partcular context.
;maxmsg=50
1234 => 4242,Example Mailbox,root@localhost
;4200 => 9855,Mark Spencer,markster@linux-
support.net,mypager@digium.com,attach=
no|serveremail=myaddy@digium.com|tz=central|maxmsg=10
;4300 => 3456,Ben Rigas,ben@american-computer.net
;4310 => -5432,Sales,sales@marko.net
;4069 => 6522,Matt
Brooks,matt@marko.net,,|tz=central|attach=yes|saycid=yes|dial
out=fromvm|callback=fromvm|review=yes|operator=yes|envelope=yes|sayduration=yes|
saydurationm=1
;4073 => 1099,Bianca Paige,bianca@biancapaige.com,,delete=1
;4110 => 3443,Rob Flynn,rflynn@blueridge.net
;
; Mailboxes may be organized into multiple contexts for
; voicemail virtualhosting
;
[other]
;The intro can be customized on a per-context basis
;directoryintro=dir-company2
1234 => 5678,Company2 User,root@localhost
```

3. Testing of Digium Asterisk Appliance IP PBX, Software Version 1.1.1

3.1 Software and Hardware Versions Tested

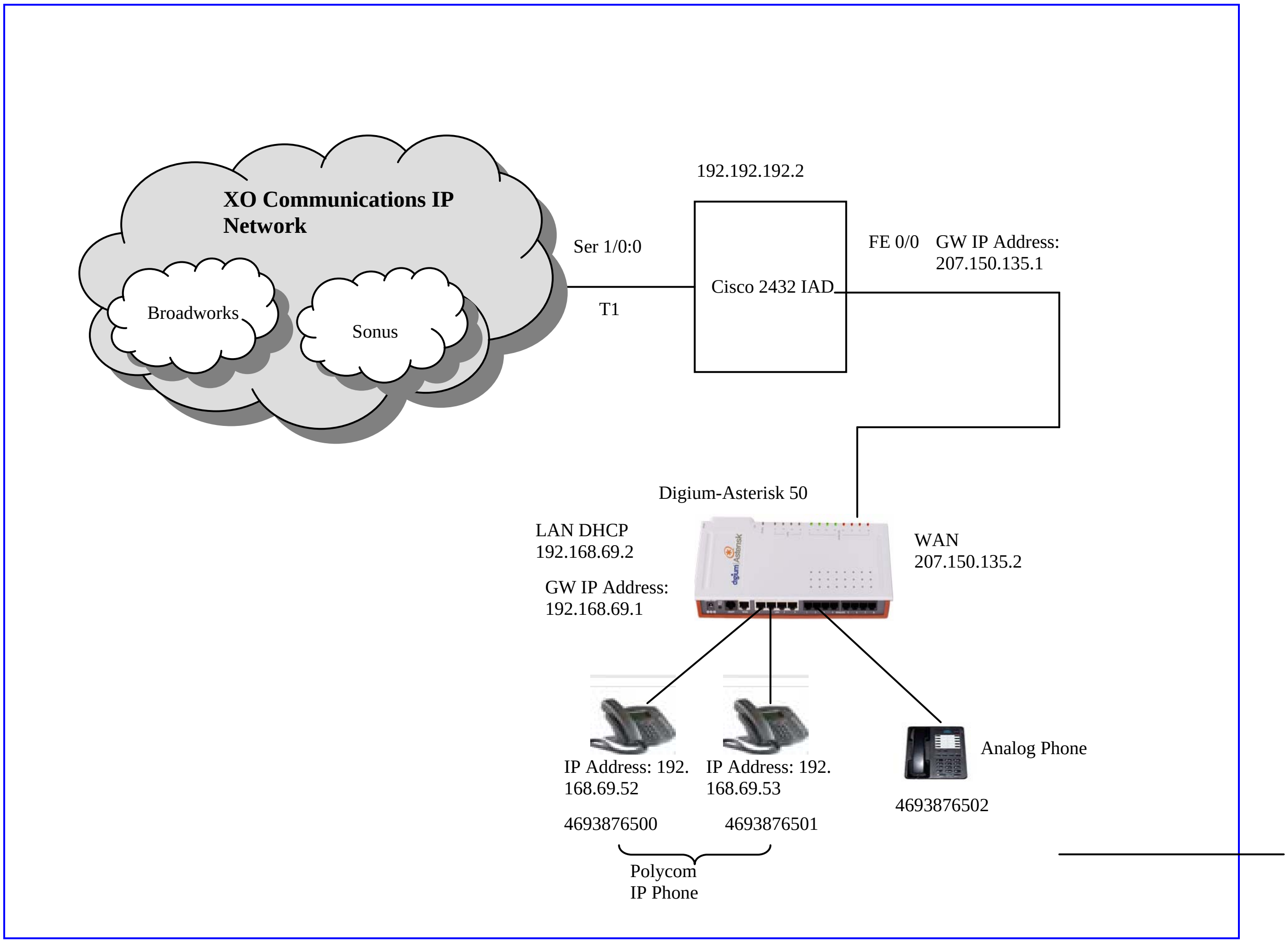
Cisco 2432 IAD CPE used as a router
Cisco 2432 Software Version: Cisco IOS Software, 2400 Software (C2430-IS-M), Version 12.4(7e), RELEASE SOFTWARE (fc5)
System image file is "flash:c2430-is-mz.124-7e.bin"
Cisco IAD2432 (R527x) processor (revision 4.1) with 119808K/11264K bytes of memory.
Processor board ID FHK1012F04P
R527x CPU at 225MHz, Implementation 40, Rev 3.1
DRAM configuration is 64 bits wide with parity disabled.
63K bytes of non-volatile configuration memory.
62720K bytes of ATA System CompactFlash (Read/Write)
0K bytes of ATA Slot0 CompactFlash (Read/Write)
Slot 0: C2400 Mother board 24FXS-2F-2FE (3 onboard DSPs) Port adapter, 33 ports
Version Identifier : V01
WIC Slot 0:
T1 (2 port) Multi-Flex Trunk WAN daughter card
Hardware revision 1.0 Board revision B0

Digium Asterisk Appliance 50 (AA50) 50
a. hardware No: S844i Rev: B1
b. The software: OS Version: Linux asteriskpbx 2.6.16.27sx00i-1.1.1
Asterisk Build: Asterisk Autotag for sx00i-1.1.1

Polycom SIP Phone & Analog Phone
Polycom IP 430 release 2.2.1 revision C
LAN Port Mode: 100FD

3.2 Diagram of Lab Test Set-Up for Digium Asterisk Appliance IP PBX, Software Version 1.1.1

The following diagram is used during lab testing to depict the configuration of Digium Asterisk Appliance 50 (AA50) deployed with Cisco2432 IAD CPE:



3.3 Digium-Asterisk Configuration for Software Version 1.1.1

This section contains GUI and dump configuration on phone provisioning, service provider, users, voicemail and sip configuration.

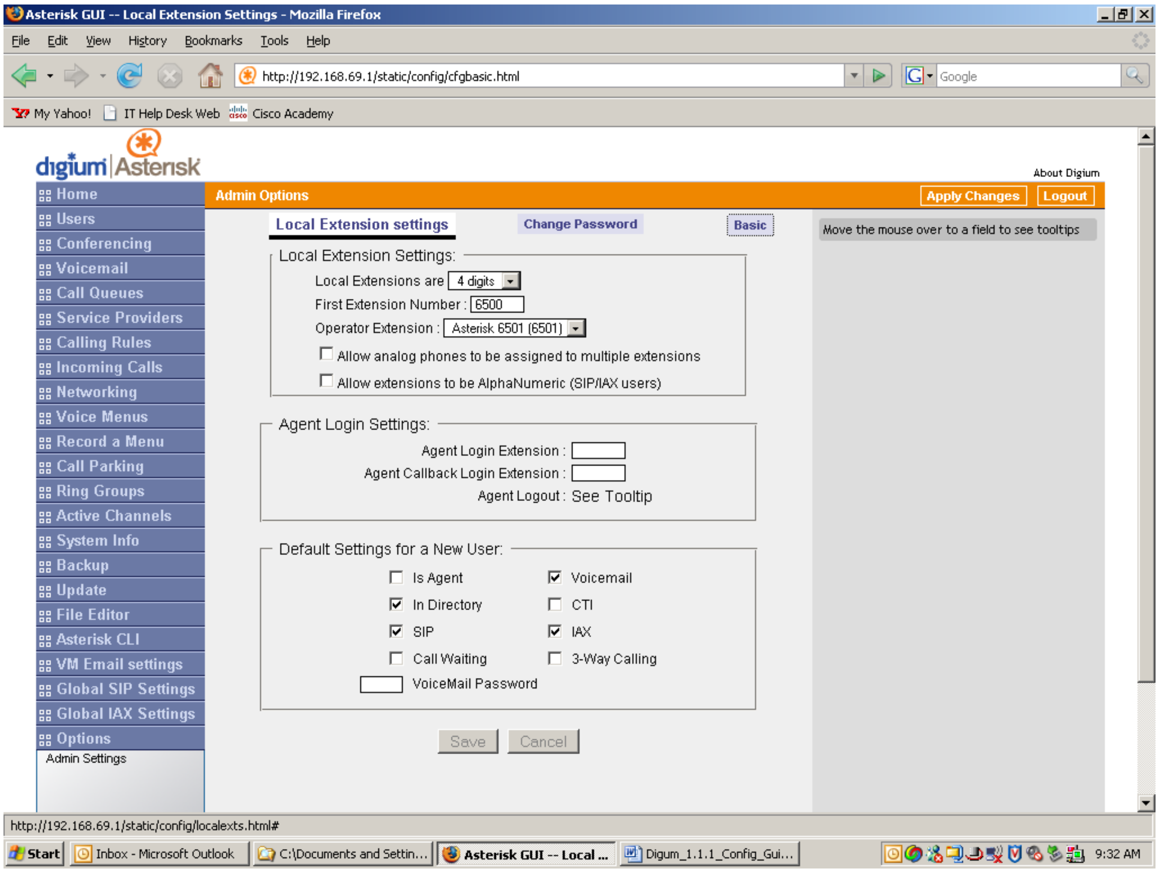


Figure 1: Show the Admin Settings

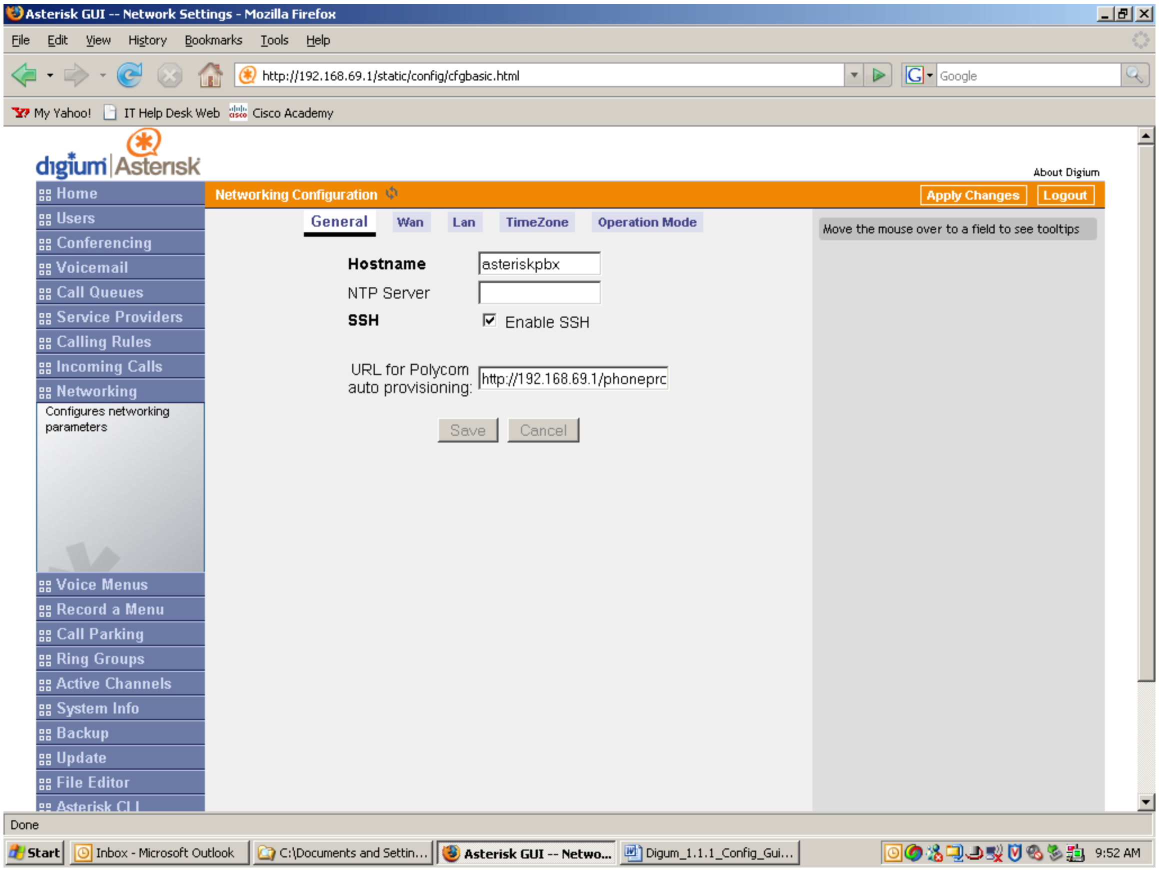


Figure 2: Show Networking Configuration

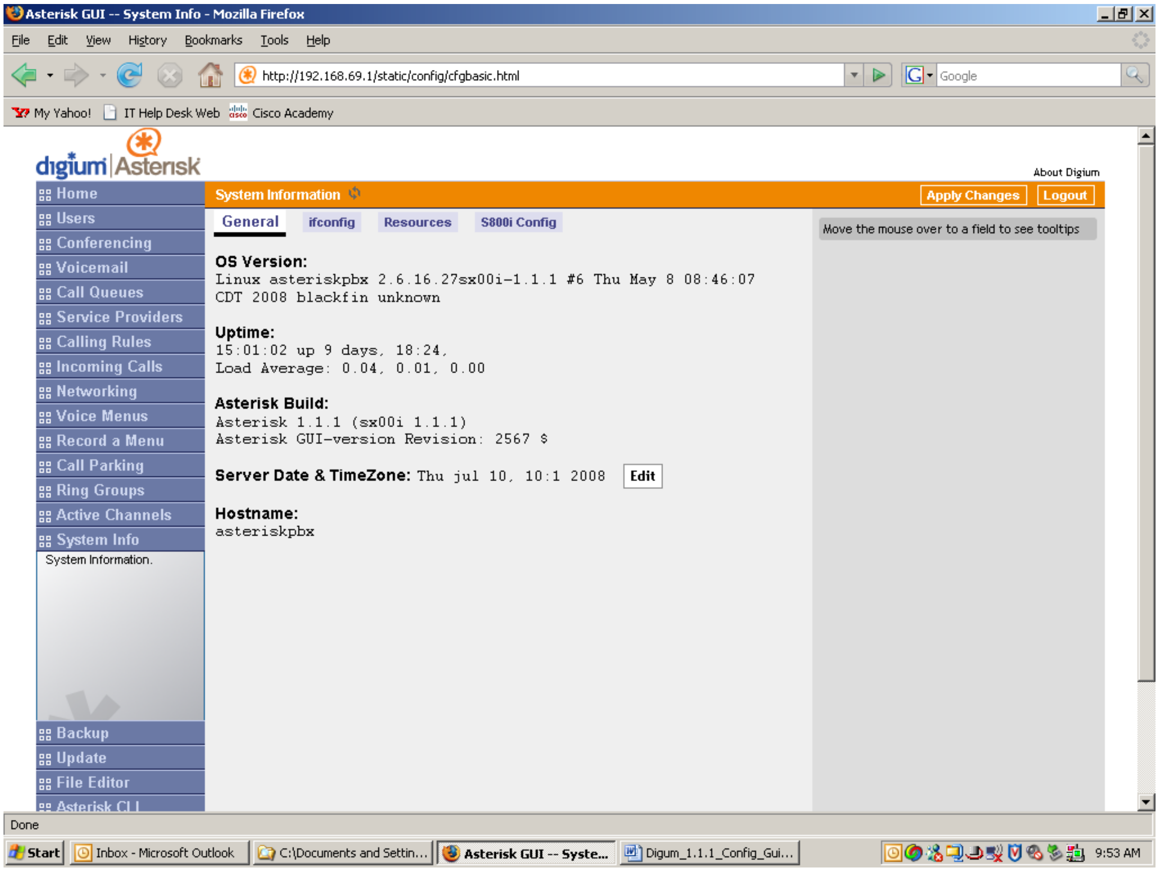


Figure 3: Show System Info.

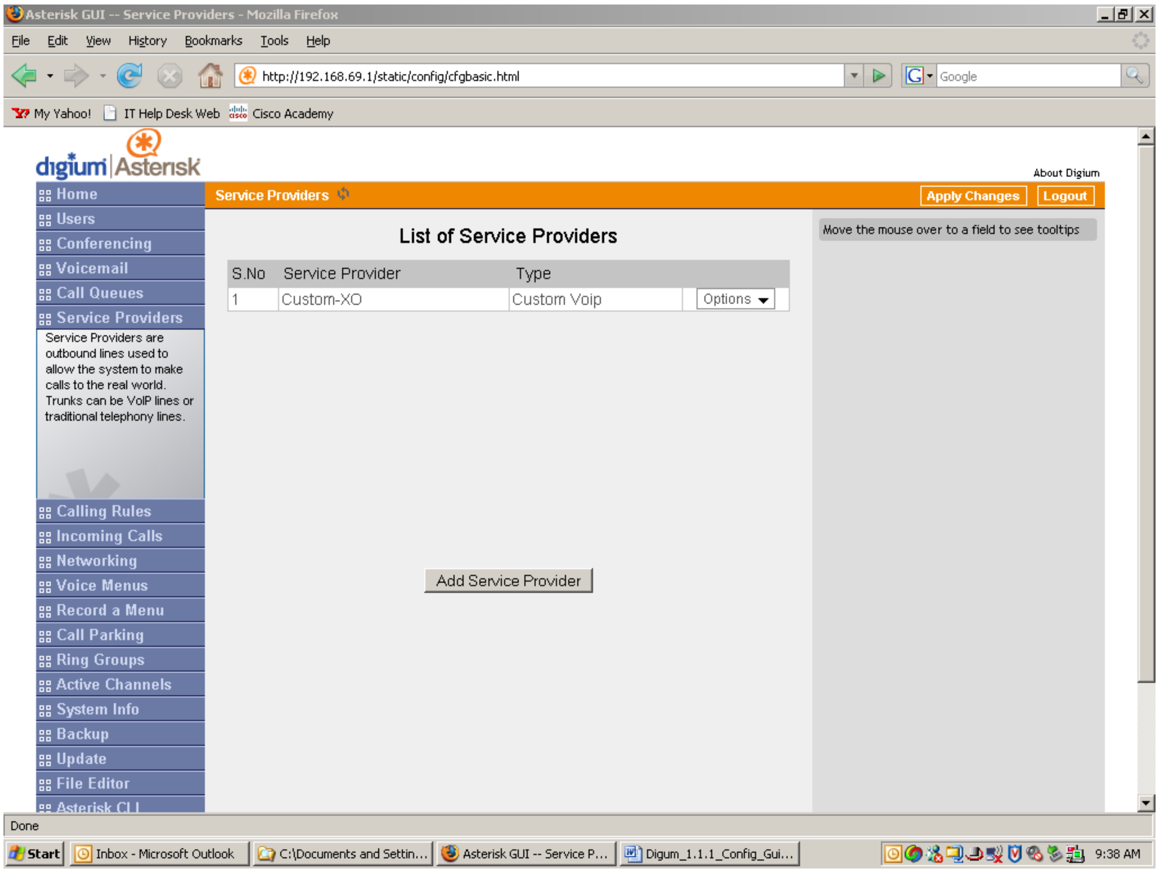


Figure 4: Show the configuration of the Service Provider

The Digium-Asterisk uses G711 ulaw

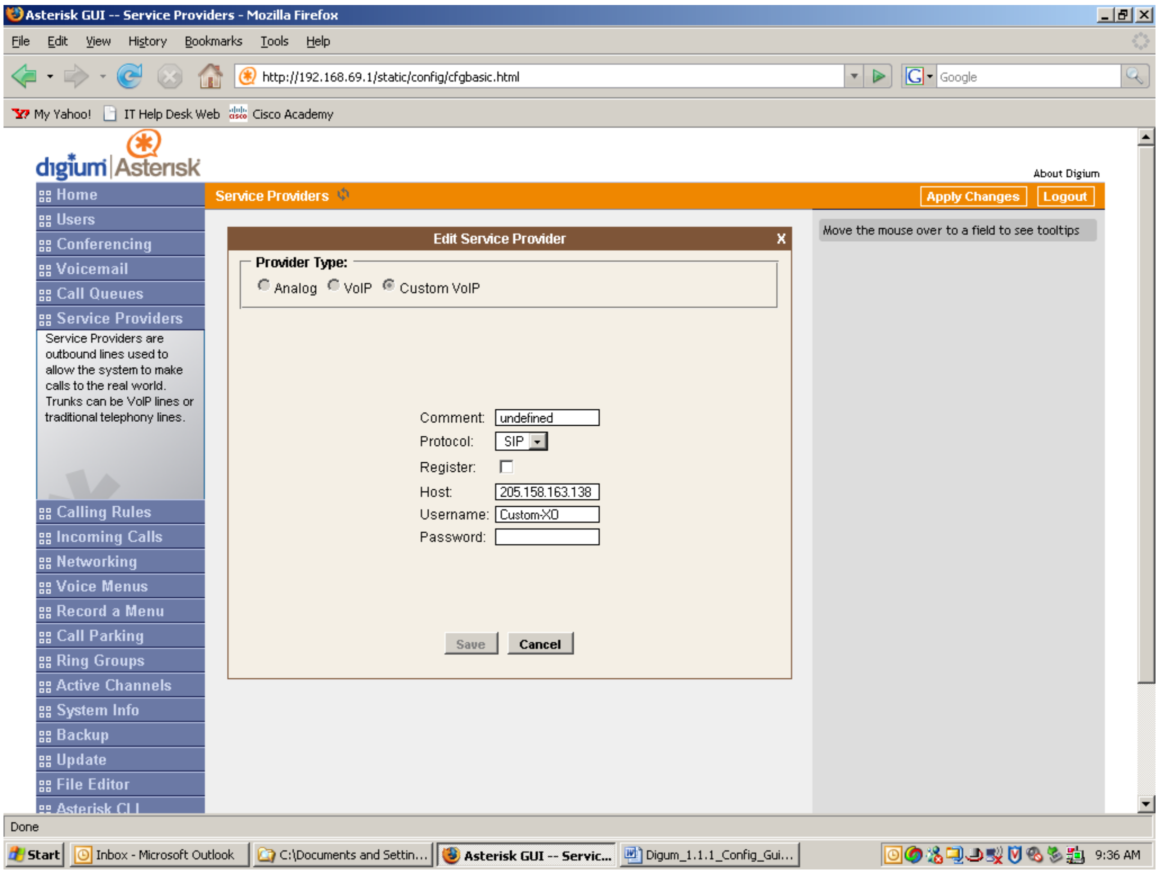


Figure 5: Show how to edit a Service Provider information

The SIP Protocol is used. The register box is not checked since it is static registration.

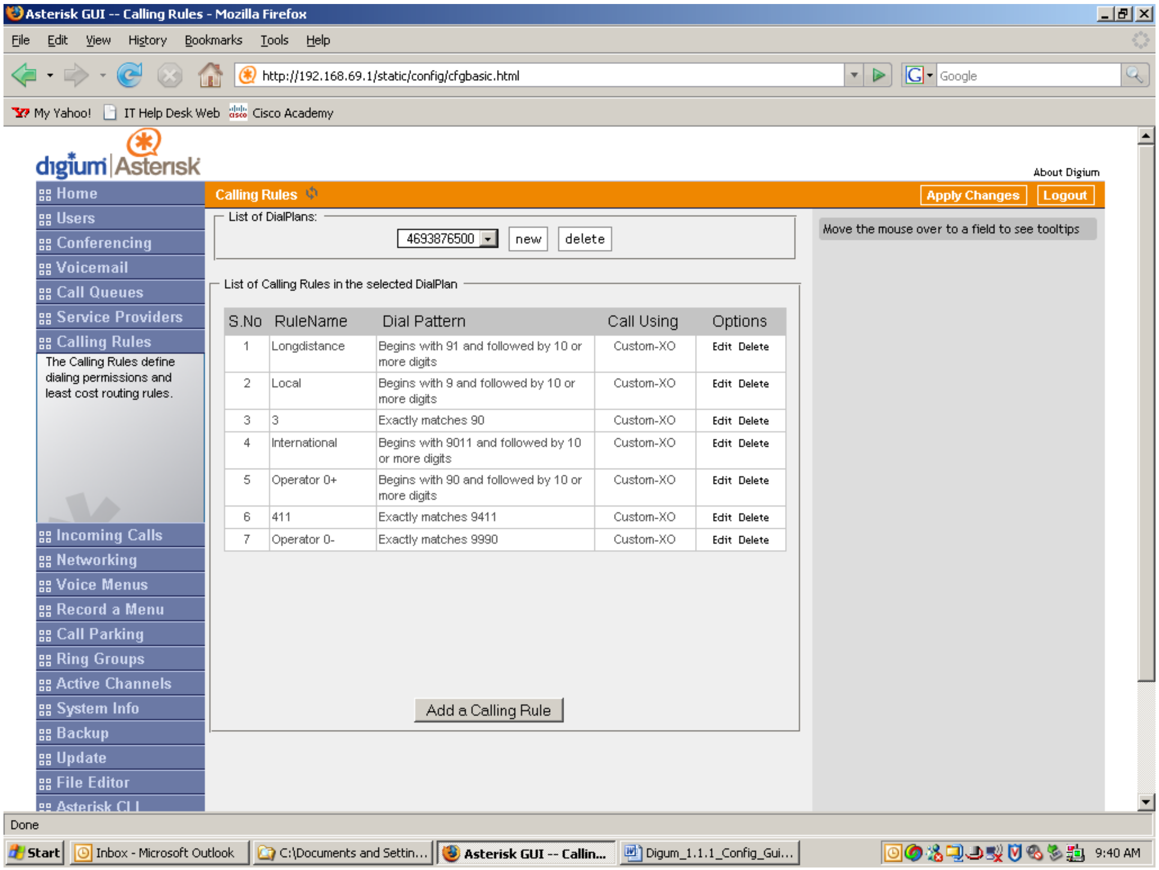


Figure 6: Show the configuration of the Outbound Calling Rules

Local number are begun with 9 and Long distance with a 91.

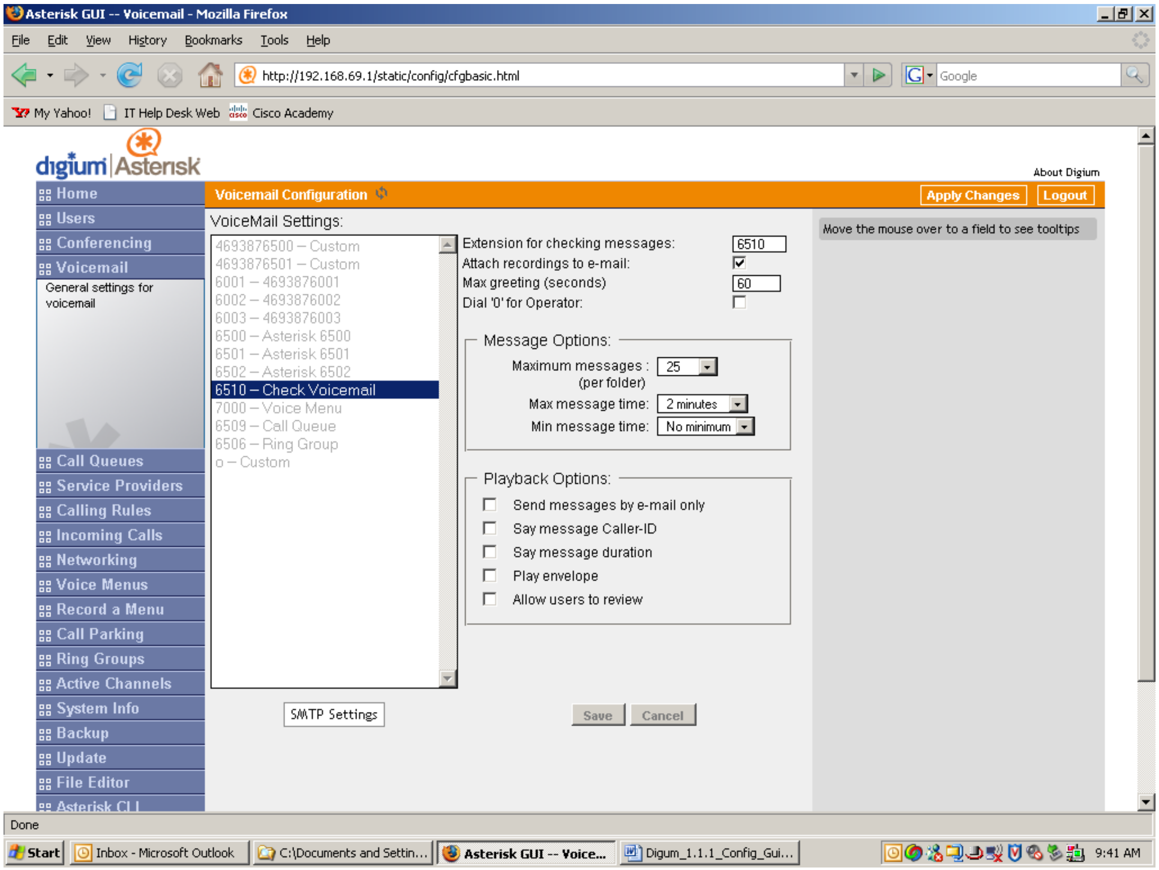


Figure 7: Show the configuration of VoiceMail.

The Attach recordings to email is selected. Maximum messages per folder and message time can be selected the drop down box.

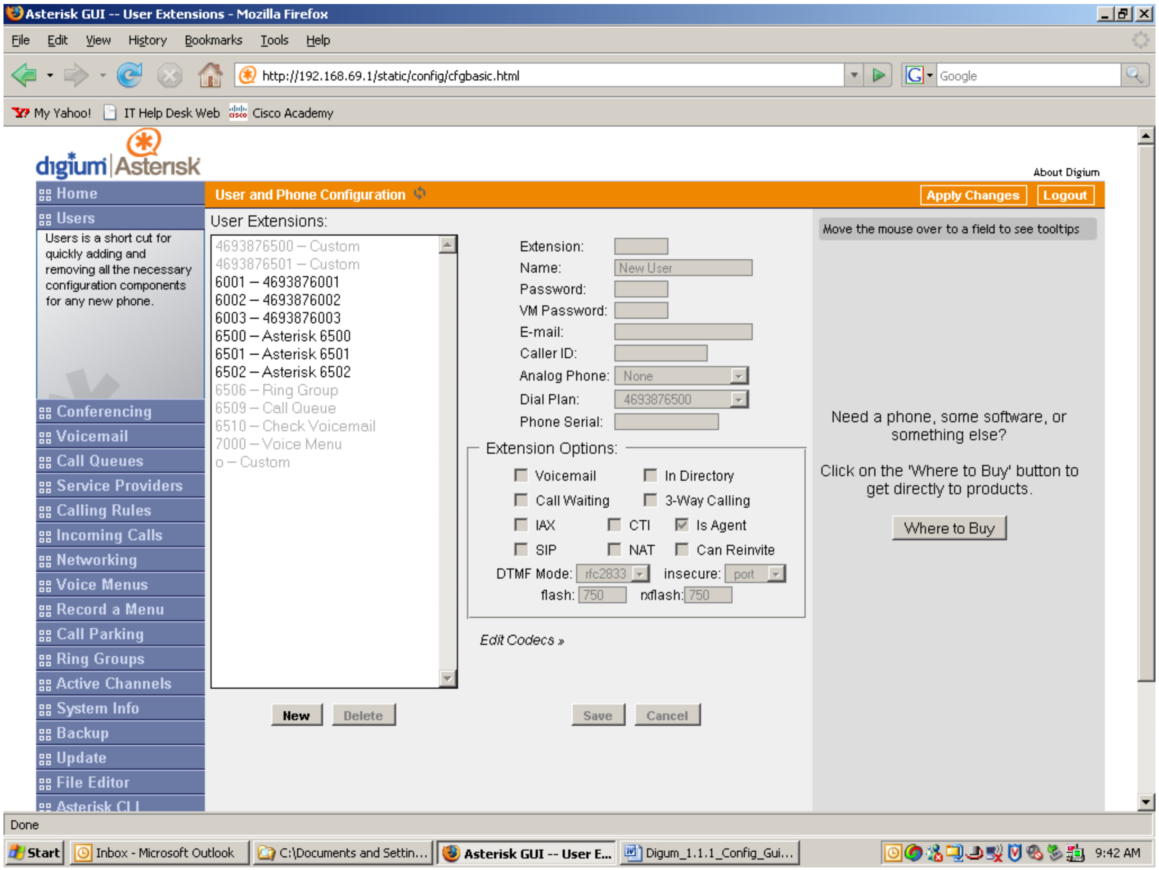


Figure 8: Show the configuration of the User Extensions

Each user can be edited in the Edit option.

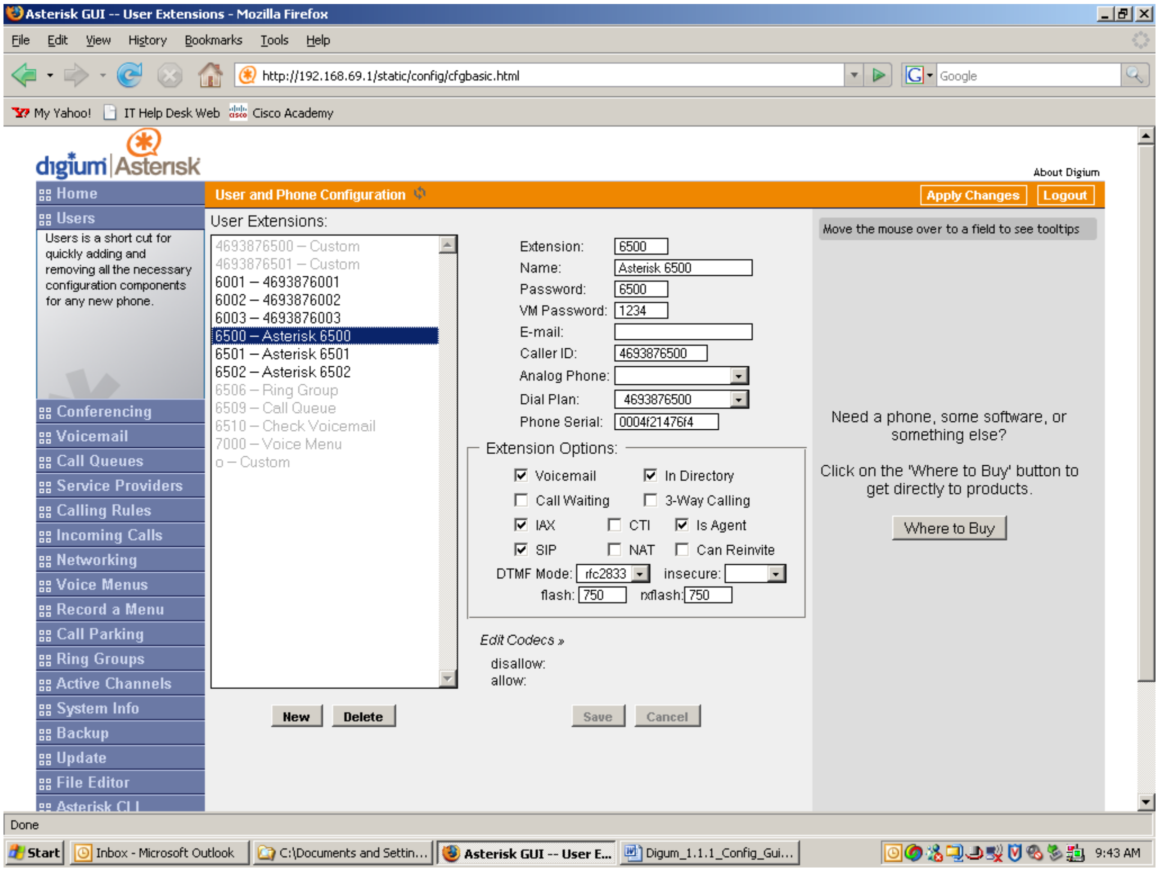


Figure 9: Show how to edit a user information

The Dial Plan is selected using the user extension. The Phone Serial is in the back of Polycom SIP Phone.

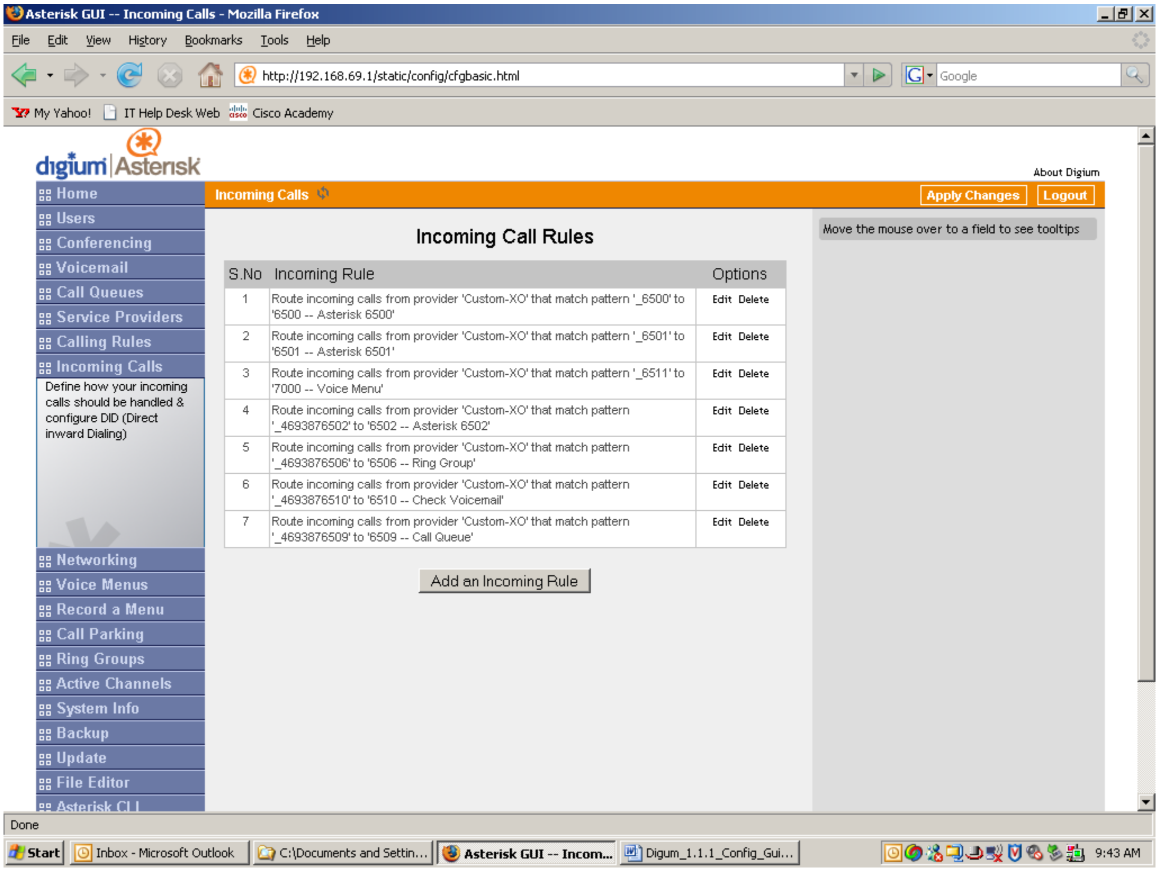


Figure 10: Show the configuration of the Incoming Calls for each user extension

Each user extension is configured to have its own rule for incoming call.

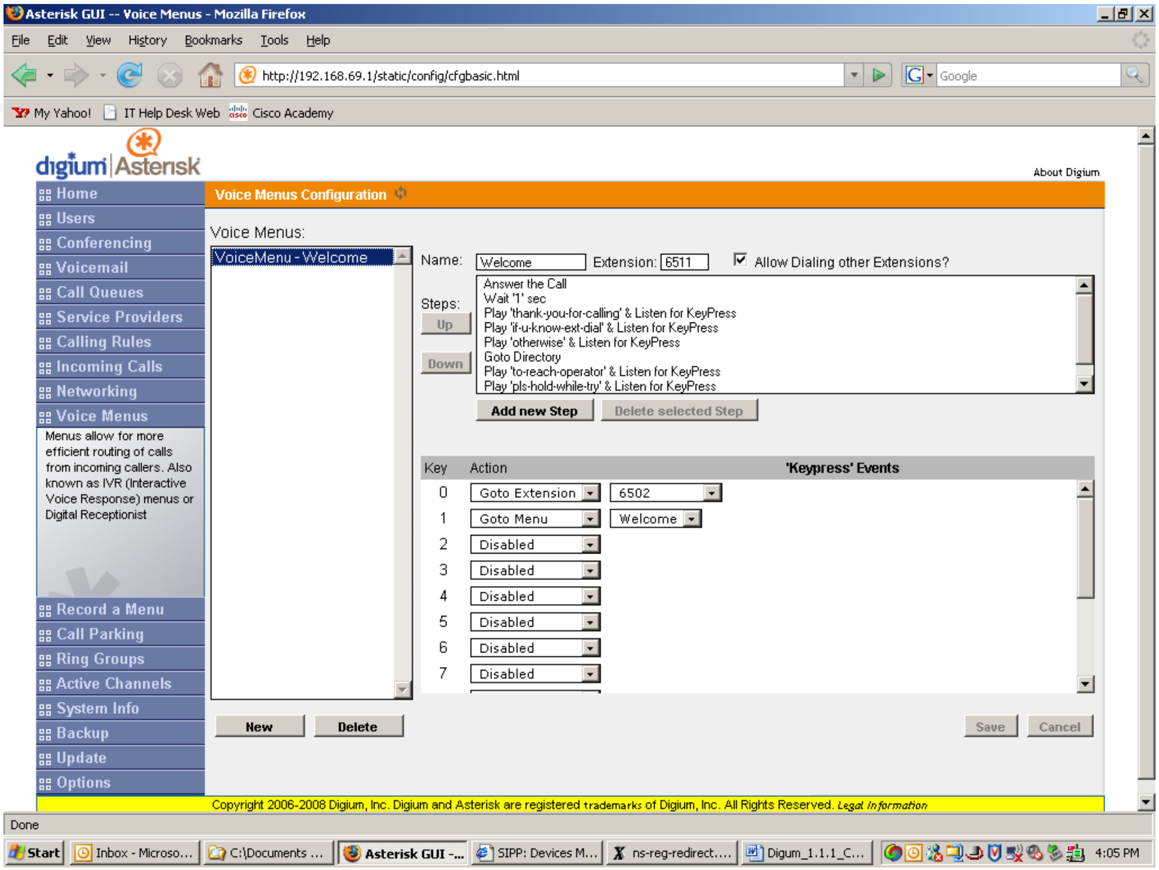


Figure 11: Show the configuration of the Auto Attendant

The new step in the greeting menu can be added in the Add a new Step drop down box.

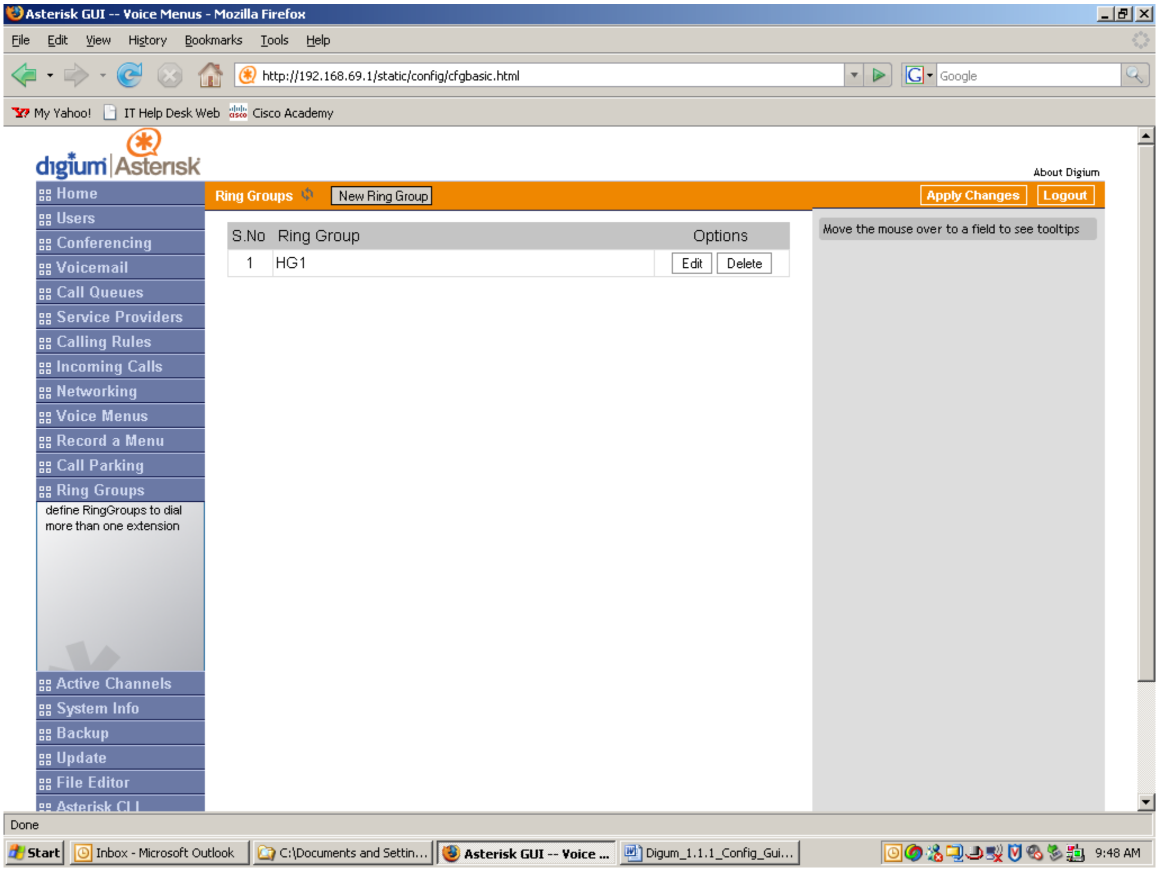
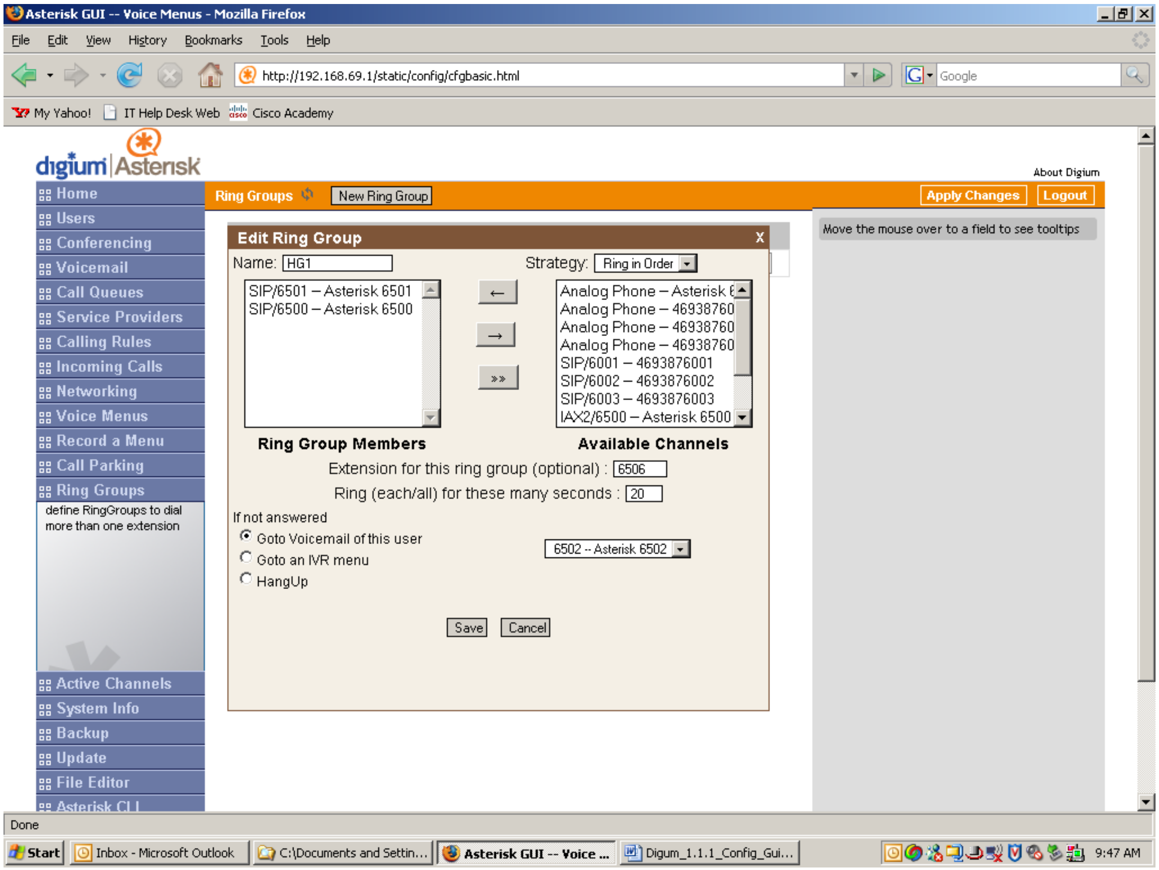


Figure 12: Show the configuration of the Hunt Group

The users are added into the Add Ring Group. It has two Strategies: Ring in Order: will ring the first agent in first list, if it's not answered in 4 rings, it will ring the second user. Ring All option: all the users in the Add Ring Group will ring simultaneously.


Asterisk GUI -- Voice Menus - Mozilla Firefox
File Edit View History Bookmarks Tools Help
http://192.168.69.1/static/config/cfgbasic.html
My Yahoo! IT Help Desk Web Cisco Academy
digium Asterisk
About Digium
Home
Users
Conferencing
Voicemail
Call Queues
Service Providers
Calling Rules
Incoming Calls
Networking
Voice Menus
Record a Menu
Call Parking
Ring Groups
define RingGroups to dial more than one extension
Active Channels
System Info
Backup
Update
File Editor
Asterisk CLI
Ring Groups
New Ring Group
Apply Changes
Logout
Move the mouse over to a field to see tooltips
Edit Ring Group
Name: HG1
Strategy: Ring in Order
SIP/6501 – Asterisk 6501
SIP/6500 – Asterisk 6500
Analog Phone – Asterisk 6
Analog Phone – 46938760
Analog Phone – 46938760
Analog Phone – 46938760
SIP/6001 – 4693876001
SIP/6002 – 4693876002
SIP/6003 – 4693876003
IAX2/6500 – Asterisk 6500
Ring Group Members
Available Channels
Extension for this ring group (optional) : 6506
Ring (each/all) for these many seconds : 20
If not answered
Goto Voicemail of this user
Goto an IVR menu
HangUp
6502 -- Asterisk 6502
Save
Cancel
Done
Start
Inbox - Microsoft Outlook
C:\Documents and Settin...
Asterisk GUI -- Voice ...
Digum_1.1.1_Config_Gui...
9:47 AM

Provisioning Dumps on SIP configuration:

This is the automatically generated configuration file in Digium appliance. This file will contain all the default SIP configuration information.

```
;! Filename: sip.conf (/etc/asterisk/sip.conf)
;! Generator: Manager
;! Creation Date: Tue Sep 11 19:48:47 2007
;!
[general]
context = default  ; Default context for incoming calls
;allowguest=no                  ; Allow or reject guest calls (default is yes, t
his can also be set to 'osp'
; if asterisk was compiled with OSP support.
;realm=mydomain.tld             ; Realm for digest authentication
; defaults to "asterisk"
; Realms MUST be globally unique according to RFC 3261
; Set this to your host name or domain name
bindport = 5060  ; UDP Port to bind to (SIP standard port is 5060)
bindaddr = 0.0.0.0  ; IP address to bind to (0.0.0.0 binds to all)
srvlookup = yes  ; Enable DNS SRV lookups on outbound calls
allowexternaldomains = no
allowexternalinvites = no
allowguest = yes
allowoverlap = yes
allowsubscribe = no
allowtransfer = yes
alwaysauthreject = no
autodomain = no
callevents = no
compactheaders = no
dumphistory = yes
g726nonstandard = no
ignoreregexpire = no
jbenable = no
jbforce = no
jblog = no
maxcallbitrate = 384
maxexpiry = 3600
minexpiry = 60
notifyringing = no
pedantic = no
promiscredir = no
recordhistory = yes
relaxdtmf = no
rtcachefriends = no
rtsavesysname = no
rtupdate = no
sendrpid = no
sipdebug = yes
t1min = 100
t38pt_udptl = no
progressinband = no
```

trustrpid = no
usereqphone = no
videosupport = no
; Note: Asterisk only uses the first host
; in SRV records
; Disabling DNS SRV lookups disables the
; ability to place SIP calls based on domain
; names to some other SIP users on the Internet
;domain=mydomain.tld ; Set default domain for this host
; If configured, Asterisk will only allow
; INVITE and REFER to non-local domains
; Use "sip show domains" to list local domains
;domain=mydomain.tld,mydomain-incoming
; Add domain and configure incoming context
; for external calls to this domain
;domain=1.2.3.4 ; Add IP address as local domain
;

This section contains the registration information

;allowexternalinvites=no ; Disable INVITE and REFER to non-local domains
; Default is yes
;autodomain=yes ; Turn this on to have Asterisk add local host
; name and local IP to domain list.
;pedantic=yes ; Enable slow, pedantic checking for Pingtel
; and multiline formatted headers for strict
; SIP compatibility (defaults to "no")
;tos=184 ; Set IP QoS to either a keyword or numeric val
;tos=lowdelay ; lowdelay,throughput,reliability,mincost,none
;maxexpiry=3600 ; Max length of incoming registration we allow
;defaultexpiry=120 ; Default length of incoming/outoing registratio
n
;notifymimetype=text/plain ; Allow overriding of mime type in MWI NOTIFY
;checkmwi=10 ; Default time between mailbox checks for peers
;vmexten=voicemail ; dialplan extension to reach mailbox sets the
; Message-Account in the MWI notify message
; defaults to "asterisk"
;videosupport=yes ; Turn on support for SIP video
;recordhistory=yes ; Record SIP history by default
; (see sip history / sip no history)
;disallow=all ; First disallow all codecs
;allow=ulaw ; Allow codecs in order of preference
;allow=ilbc ;
;musicclass=default ; Sets the default music on hold class for all S
IP calls
; This may also be set for individual users/peers
;language=en ; Default language setting for all users/peers
; This may also be set for individual users/peers
;relaxdtmf=yes ; Relax dtmf handling
;rtptimeout=60 ; Terminate call if 60 seconds of no RTP activit
y
; when we're not on hold
;rtpholdtimeout=300 ; Terminate call if 300 seconds of no RTP activi
ty
; when we're on hold (must be > rtptimeout)
;trustrpid = no ; If Remote-Party-ID should be trusted
;sendrpid = yes ; If Remote-Party-ID should be sent

;progressinband=never ; If we should generate in-band ringing always
; use 'never' to never use in-band signalling, even in cases
; where some buggy devices might not render it
;useragent=Asterisk PBX ; Allows you to change the user agent string
;promiscredir = no ; If yes, allows 302 or REDIR to non-local SIP a
ddress
; Note that promiscredir when redirects are made to the
; local system will cause loops since SIP is incapable
; of performing a "hairpin" call.
;usereqphone = no ; If yes, ";user=phone" is added to uri that con
tains
; a valid phone number

This section contains the DTMF information on Digium

;dtmfmode = rfc2833 ; Set default dtmfmode for sending DTMF. Default
: rfc2833
; Other options:
; info : SIP INFO messages
; inband : Inband audio (requires 64 kbit codec -alaw, ulaw)
; auto : Use rfc2833 if offered, inband otherwise
;compactheaders = yes ; send compact sip headers.
;sipdebug = yes ; Turn on SIP debugging by default, from
; the moment the channel loads this configuration
;subscribecontext = default ; Set a specific context for SUBSCRIBE requests
; Useful to limit subscriptions to local extensions
; Settable per peer/user also
;notifyringing = yes ; Notify subscriptions on RINGING state

Provisioning Dumps on Users Configuration:

;! Automatically generated configuration file
;! Filename: users.conf (/etc/asterisk/users.conf)
;! Generator: Manager
;! Creation Date: Wed Sep 19 15:01:53 2007
;!
[general]
trunkstyle = voip
fullname = New User
phone = none

- This section contains the configuration file of user extension 6500:

userbase = 6500
hasvoicemail = yes
hasdirectory = yes
hassip = yes
hasiax = yes
hasmanager = no
callwaiting = no
threewaycalling = no
localextenlength = 4
switchtype = national

usecallerid = yes
hidecallerid = no
usecallingpres = yes
canpark = yes
cancallforward = yes
callreturn = yes
echocancel = yes
echocancelwhenbridged = yes
rxgain = 0.0
txgain = 0.0
immediate = no
callwaitingcallerid = yes
transfer = yes
allow_aliasextns = no
allow_an_extns = no
hasagent = no

[6500]
callwaiting = no
cid_number = 4693876500
context = numberplan-custom-1
fullname = Asterisk 6500
signalling = fxo_ks
group =
hasagent = yes
hasdirectory = yes
hasiax = yes
hasmanager = no
hassip = yes
hasvoicemail = yes
host = dynamic
mailbox = 6500
secret = 6500
threewaycalling = no
vmsecret = 1234
registeriax = yes
registersip = yes
autoprov = yes
canreinvite = no
nat = no
dtmfmode = rfc2833
macaddress = 0004f21476f4
label = 6500

- This section contains the configuration file of user extension 6501:

[6501]
callwaiting = no
cid_number = 4693876501
context = numberplan-custom-2
fullname = Asterisk 6501
signalling = fxo_ks
group =
hasagent = yes
hasdirectory = yes
hasiax = yes

hasmanager = no
hassip = yes
hasvoicemail = yes
host = dynamic
mailbox = 6501
secret = 6501
threewaycalling = no
vmsecret = 1234
registeriax = yes
registersip = yes
macaddress = 0004f21476c8
autoprov = yes
label = 6501
canreinvite = no
nat = no
dtmfmode = rfc2833

- This section contains the configuration file of Digium Trunk Group

[Digium_Trk1]
allow = all
context = DID_Digium_Trk1
dialformat = ${EXTEN:1}
hasexten = no
hasiax = no
hassip = yes
host = 205.158.163.231
port = 5060
registeriax = no
registersip = no
trunkname = Custom-XO
trunkstyle = customvoip
username = Cusom-XO
fromuser = 4693876500

- This section contains the configuration file of user extension 6502

[6502]
callwaiting = no
cid_number = 4693876502
fullname = Asterisk 6502
  as agent = fxo_ks
rxgain = 0.0
txgain = 0.0
group =
  as agent = yes
hasdirectory = yes
hasiax = yes
hasmanager = no
hassip = yes
hasvoicemail = yes
host = dynamic
mailbox = 6502
secret = 6502
threewaycalling = no
zapchan = 1

```
registeriax = yes
registersip = yes
autoprov = no
canreinvite = no
nat = no
dtmfmode = rfc2833
```

Provisioning Dumps on Phone Configuration:

This section will contain the Polycom SIP phone configuration file of the:

```
[general]
;serveraddr=192.168.1.1 ; Address to send to the phone to use as server address.
serveriface=eth1        ; Same as above, except an ethernet interface.  Useful f
or when the interface uses DHCP.
                        ; There is no default for either of the above, and only
one should be set.
;serverport=5060        ; Port to send to the phone to use as server port.  Defa
ult is 5060.
```

Dumps on Voicemail Configuration:

- This section contains the default configuration on Digium Voicemail, and how to configure the optional parts on the Voicemail

```
;! Automatically generated configuration file
;! Filename: voicemail.conf (/etc/asterisk/voicemail.conf)
;! Generator: Manager
;! Creation Date: Mon Aug 27 21:42:27 2007
;!
;
; Voicemail Configuration
;
;
; NOTE: Asterisk has to edit this file to change a user's password.  This does
; note currently work with the "#include <file>" directive for Asterisk
; configuration files.  Do not use it with this configuration file.
;
[general]
; Default formats for writing Voicemail
;format=g723sf|wav49|wav
format = ulaw
;
; WARNING:
; If you change the list of formats that you record voicemail in
; when you have mailboxes that contain messages, you _MUST_ absolutely
; manually go through those mailboxes and convert/delete/add the
; the message files so that they appear to have been stored using
; your new format list. If you don't do this, very unpleasant
; things may happen to your users while they are retrieving and
; manipulating their voicemail.
;
; In other words: don't change the format list on a production system
; unless you are _VERY_  sure that you know what you are doing and are
; prepared for the consequences.
;
; Who the e-mail notification should appear to come from
```

```
serveremail = asterisk
;serveremail=asterisk@linux-support.net
; Should the email contain the voicemail as an attachment
attach = yes
maxmsg = 25
maxmessage = 120
; Minimum length of a voicemail message in seconds for the message to be kept
; The default is no minimum.
minmessage = 0
; Maximum length of greetings in seconds
;maxgreet=60
; How many miliseconds to skip forward/back when rew/ff in message playback
skipms = 3000
; How many seconds of silence before we end the recording
maxsilence = 10
; Silence threshold (what we consider silence, the lower, the more sensitive)
silencethreshold = 128
; Max number of failed login attempts
maxlogins = 3
; If you need to have an external program, i.e. /usr/bin/myapp called when a
; voicemail is left, delivered, or your voicemailbox is checked, uncomment
; this:
;externnotify=/usr/bin/myapp
; If you need to have an external program, i.e. /usr/bin/myapp called when a
; voicemail password is changed, uncomment this:
;externpass=/usr/bin/myapp
; For the directory, you can override the intro file if you want
;directoryintro=dir-intro
; The character set for voicemail messages can be specified here
;charset=ISO-8859-1
; The ADSI feature descriptor number to download to
;adsifdn=0000000F
; The ADSI security lock code
;adsisec=9BDBF7AC
; The ADSI voicemail application version number.
;adsiver=1
; Skip the "[PBX]:" string from the message title
;pbxskip=yes
; Change the From: string
;fromstring=The Asterisk PBX
; Permit finding entries for forward/compose from the directory
;usedirectory=yes
;
; Change the from, body and/or subject, variables:
;     VM_NAME, VM_DUR, VM_MSGNUM, VM_MAILBOX, VM_CALLERID, VM_CIDNUM,
;     VM_CIDNAME, VM_DATE
;
; Note: The emailbody config row can only be up to 512 characters due to a
;       limitation in the Asterisk configuration subsystem.
;emailsubject=[PBX]: New message ${VM_MSGNUM} in mailbox ${VM_MAILBOX}
; The following definition is very close to the default, but the default shows
; just the CIDNAME, if it is not null, otherise just the CIDNUM, or "an unknown
; caller", if they are both null.
;emailbody=Dear ${VM_NAME}:\n\n\tjust wanted to let you know you were just left
a ${VM_DUR} long message (number ${VM_MSGNUM})\nin mailbox ${VM_MAILBOX} from ${
VM_CALLERID}, on ${VM_DATE}, so you might\nwant to check it when you get a chanc
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e.  Thanks!\n\n\t\t\t\t--Asterisk\n
;
; You can also change the Pager From: string, the pager body and/or subject.
; The above defined variables also can be used here
;pagerfromstring=The Asterisk PBX
;pagersubject=New VM
;pagerbody=New ${VM_DUR} long msg in box ${VM_MAILBOX}\nfrom ${VM_CALLERID}, on
${VM_DATE}
;
; Set the date format on outgoing mails. Valid arguments can be found on the
; strftime(3) man page
;
; Default
emaildateformat = %A, %B %d, %Y at %r
; 24h date format
;emaildateformat=%A, %d %B %Y at %H:%M:%S
;
; You can override the default program to send e-mail if you wish, too
;
mailcmd = /bin/ssmtp
;
; Users may be located in different timezones, or may have different
; message announcements for their introductory message when they enter
; the voicemail system. Set the message and the timezone each user
; hears here. Set the user into one of these zones with the tz= attribute
; in the options field of the mailbox. Of course, language substitution
; still applies here so you may have several directory trees that have
; alternate language choices.
;
; Look in /usr/share/zoneinfo/ for names of timezones.
; Look at the manual page for strftime for a quick tutorial on how the
; variable substitution is done on the values below.
;
; Supported values:
; 'filename'    filename of a soundfile (single ticks around the filename
;               required)
; ${VAR}        variable substitution
; A or a        Day of week (Saturday, Sunday, ...)
; B or b or h   Month name (January, February, ...)
; d or e        numeric day of month (first, second, ..., thirty-first)
; Y             Year
; I or l        Hour, 12 hour clock
; H             Hour, 24 hour clock (single digit hours preceded by "oh")
; k             Hour, 24 hour clock (single digit hours NOT preceded by "oh")
; M             Minute, with 00 pronounced as "o'clock"
; N             Minute, with 00 pronounced as "hundred" (US military time)
; P or p        AM or PM
; Q             "today", "yesterday" or ABdY
;               (*note: not standard strftime value)
; q             "" (for today), "yesterday", weekday, or ABdY
;               (*note: not standard strftime value)
; R             24 hour time, including minute
;
;
;
; Each mailbox is listed in the form <mailbox>=<password>,<name>,<email>,<pager_
```

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email>,<options>
; if the e-mail is specified, a message will be sent when a message is
; received, to the given mailbox. If pager is specified, a message will be
; sent there as well. If the password is prefixed by '-', then it is
; considered to be unchangable.
;
; Advanced options example is extension 4069
; NOTE: All options can be expressed globally in the general section, and
; overriden in the per-mailbox settings, unless listed otherwise.
;
; tz=central          ; Timezone from zonemessages above.  Irrelevant if envel
ope=no.
; attach=yes          ; Attach the voicemail to the notification email *NOT* t
he pager email
; saycid=yes          ; Say the caller id information before the message. If n
ot described,
;     or set to no, it will be in the envelope
; cidinternalcontexts=intern    ; Internal Context for Name Playback instead of
extension digits when saying caller id.
; sayduration=no       ; Turn on/off the duration information before the messag
e. [ON by default]
; saydurationm=2       ; Specify the minimum duration to say. Default is 2 minu
tes
; dialout=fromvm       ; Context to dial out from [option 4 from the advanced m
enu]
;     if not listed, dialing out will not be permitted
sendvoicemail = yes  ; Context to Send voicemail from [option 5 from the advance
d menu]
maxgreet = 60
; if not listed, sending messages from inside voicemail will not be
; permitted
; searchcontexts=yes    ; Current default behavior is to search only the default
 context
; if one is not specified.  The older behavior was to search all contexts.
; This option restores the old behavior [DEFAULT=no]
; callback=fromvm       ; Context to call back from
;     if not listed, calling the sender back will not be permitted
; review=yes           ; Allow sender to review/rerecord their message before s
aving it [OFF by default
; operator=yes         ; Allow sender to hit 0 before/after/during  leaving a v
oicemail to
;     reach an operator  [OFF by default]
; envelope=no          ; Turn on/off envelope playback before message playback.
 [ON by default]
;     This does NOT affect option 3,3 from the advanced options menu
; delete=yes           ; After notification, the voicemail is deleted from the
server. [per-mailbox only]
;     This is intended for use with users who wish to receive their voicemail ON
LY by email.
; nextaftercmd=yes      ; Skips to the next message after hitting 7 or 9 to dele
te/save current message.
;     [global option only at this time]
; forcename=yes        ; Forces a new user to record their name.  A new user is
;     determined by the password being the same as
;     the mailbox number.  The default is "no".
; forcegreetings=no     ; This is the same as forcename, except for recording
```

```
;     greetings.  The default is "no".
; hidefromdir=yes       ; Hide this mailbox from the directory produced by app_d
irectory
;     The default is "no".
[zonemessages]
eastern = America/New_York|'vm-received' Q 'digits/at' IMp
central = America/Chicago|'vm-received' Q 'digits/at' IMp
central24 = America/Chicago|'vm-received' q 'digits/at' H N 'hours'
military = Zulu|'vm-received' q 'digits/at' H N 'hours' 'phonetic/z_p'

[default]
; Define maximum number of messages per folder for particular context.
;maxmsg=50
1234 => 4242,Example Mailbox,root@localhost
;4200 => 9855,Mark Spencer,markster@linux-support.net,mypager@digium.com,attach=
no|serveremail=myaddy@digium.com|tz=central|maxmsg=10
;4300 => 3456,Ben Rigas,ben@american-computer.net
;4310 => -5432,Sales,sales@marko.net
;4069 => 6522,Matt Brooks,matt@marko.net,,|tz=central|attach=yes|saycid=yes|dial
out=fromvm|callback=fromvm|review=yes|operator=yes|envelope=yes|sayduration=yes|
saydurationm=1
;4073 => 1099,Bianca Paige,bianca@biancapaige.com,,delete=1
;4110 => 3443,Rob Flynn,rflynn@blueridge.net
;
; Mailboxes may be organized into multiple contexts for
; voicemail virtualhosting
;
[other]
;The intro can be customized on a per-context basis
;directoryintro=dir-company2
1234 => 5678,Company2 User,root@localhost
```

For Further Information

Digium-Asterisk User Manual
https://www.digium.com/en/supportcenter/documentation/viewdocs/AA50

Polycom Phone User Manual
http://www.polycom.com/usa/en/support/support.html