

ESTADO DE ALAGOAS



Introdução à voz sobre IP e Asterisk

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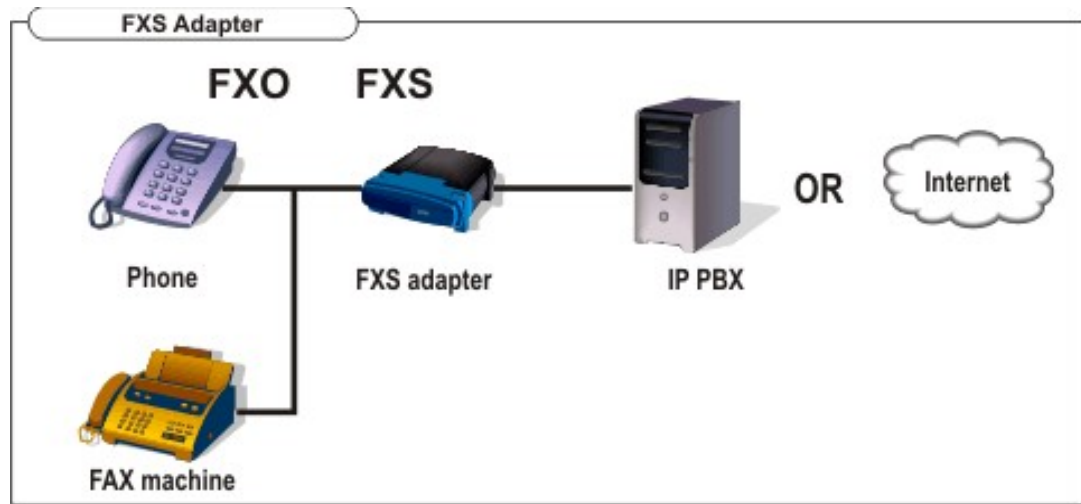
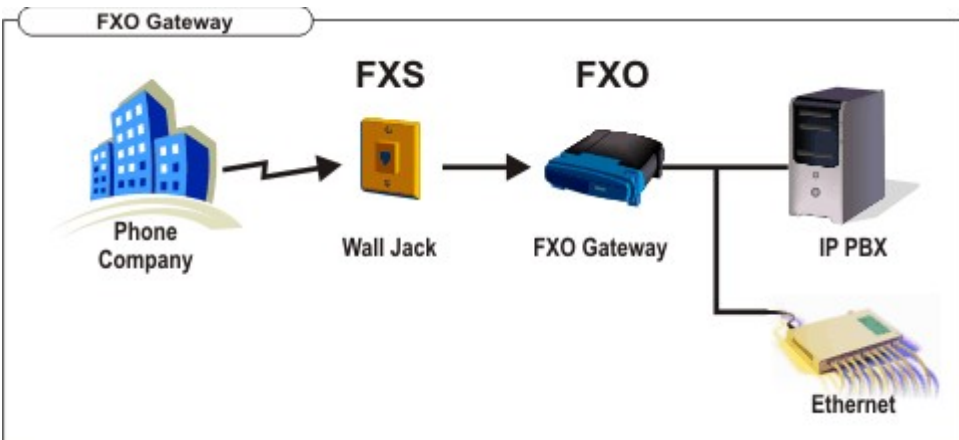
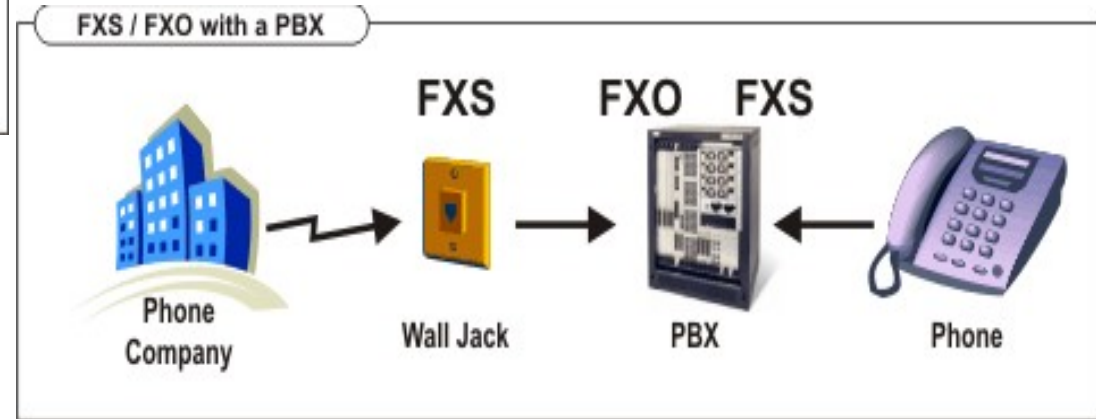
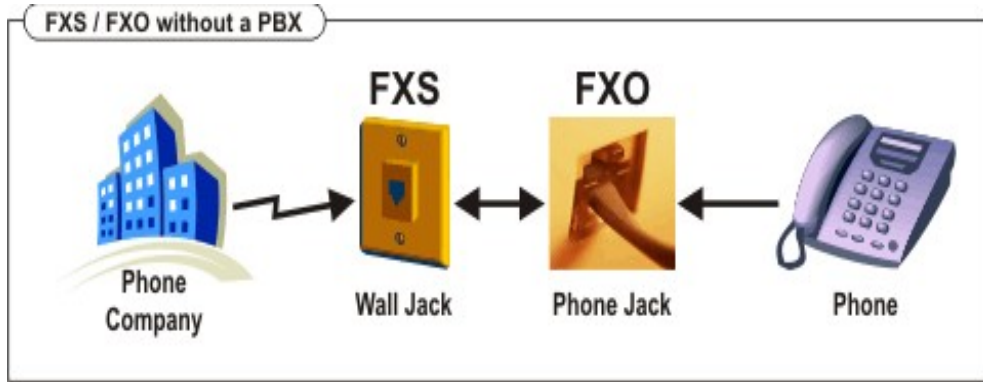
# Sumário

- x Canais FXS e FXO;
- x Troncos T1, E1 e sinalização;
- x Instalação Asterisk 1.4.24
- x Criando ramais SIP;
- x Criando ramais DGV;
- x Criando ramais KHOMP;
- x Criando um plano de discagem;
- x Utilizando Asterisk/CLI;
- x Instalando clientes SIP (ekiga, x-lite);
- x Roteiro de atividades;

# FXO e FXS

- × Interfaces FX (Foreign eXchange)
  - × Interfaces analógicas. É um termo aplicado a troncos com acesso à rede pública (PSTN);
- × FXO (Foreign eXchange Office)
  - × Utilizado para comunicação com a central pública ou uma porta de ramal de um PABX. **A linha telefônica oriunda da PSTN é conectada a uma porta FXO da central;**
- × FXS (Foreign eXchange Station)
  - × Utilizado para conectar dispositivos básicos: telefones, fax e etc. **É na porta FXS da central que se coloca o telefone comun;**

# FXO e FXS



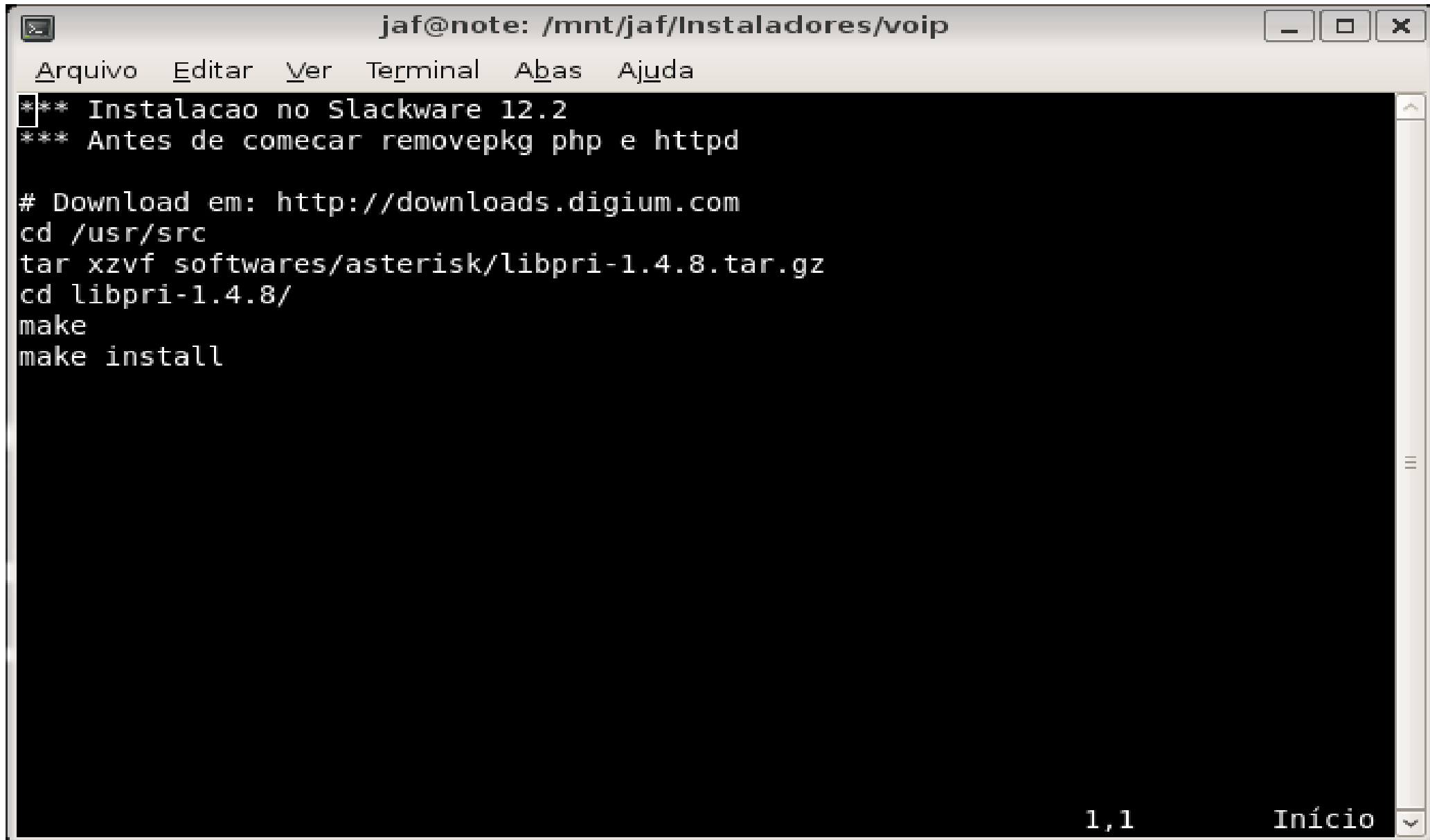
# Troncos digitais T1, E1

- x Sinalização:
  - x MFC/R2 – Muito utilizado na América Latina e Ásia. Sinalização associada ao canal (CAS-Channel Associated Signaling). Amplamente utilizado no Brasil;
  - x ISDN – Rede Digital de Serviços Integrados. SMS e outros foram desenvolvidos inicialmente para esse tipo de sinalização.



# Instalação Asterisk 1.4.24

## Contexto DigiVoice

A terminal window titled 'jaf@note: /mnt/jaf/Instaladores/voip' with a menu bar containing 'Arquivo', 'Editar', 'Ver', 'Terminal', 'Abas', and 'Ajuda'. The terminal content shows instructions for installing Asterisk on Slackware 12.2, including removing php and httpd, downloading from digium.com, and running 'make' and 'make install'. The status bar at the bottom right shows '1,1' and 'Início'.

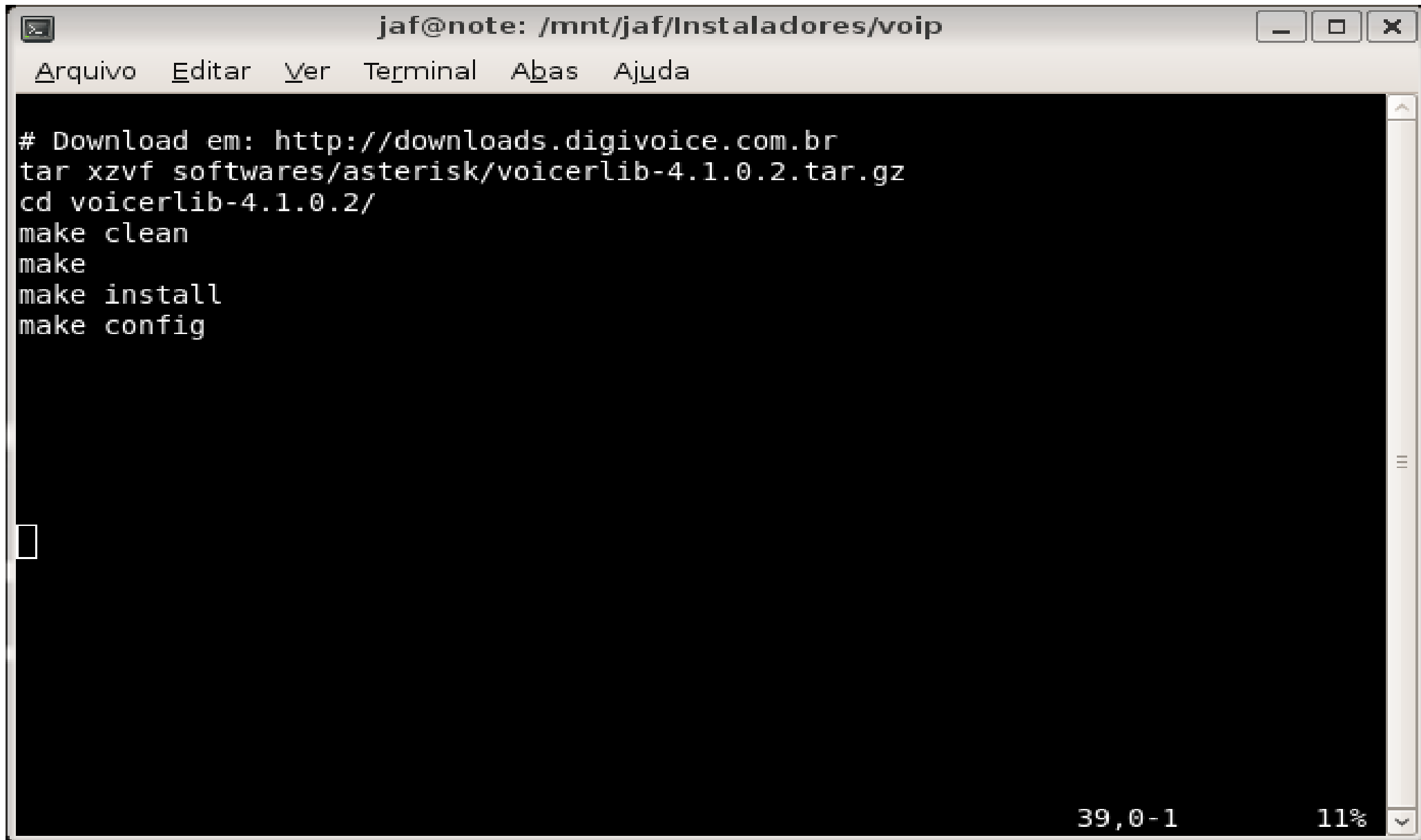
```
jaf@note: /mnt/jaf/Instaladores/voip
Arquivo  Editar  Ver  Terminal  Abas  Ajuda
*** Instalacao no Slackware 12.2
*** Antes de comecar removepkg php e httpd

# Download em: http://downloads.digium.com
cd /usr/src
tar xzvf softwares/asterisk/libpri-1.4.8.tar.gz
cd libpri-1.4.8/
make
make install

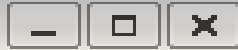
1,1  Início
```

# Instalação Asterisk 1.4.24

## Contexto DigiVoice

A terminal window titled 'jaf@note: /mnt/jaf/Instaladores/voip' with a menu bar containing 'Arquivo', 'Editar', 'Ver', 'Terminal', 'Abas', and 'Ajuda'. The terminal displays a series of commands for installing Asterisk voicemail components. The commands are: '# Download em: http://downloads.digivoice.com.br', 'tar xzvf softwares/asterisk/voicerlib-4.1.0.2.tar.gz', 'cd voicerlib-4.1.0.2/', 'make clean', 'make', 'make install', and 'make config'. The terminal shows a cursor at the end of the last command. The status bar at the bottom right indicates '39,0-1' and '11%'.

jaf@note: /mnt/jaf/Instaladores/voip



Arquivo Editar Ver Terminal Abas Ajuda

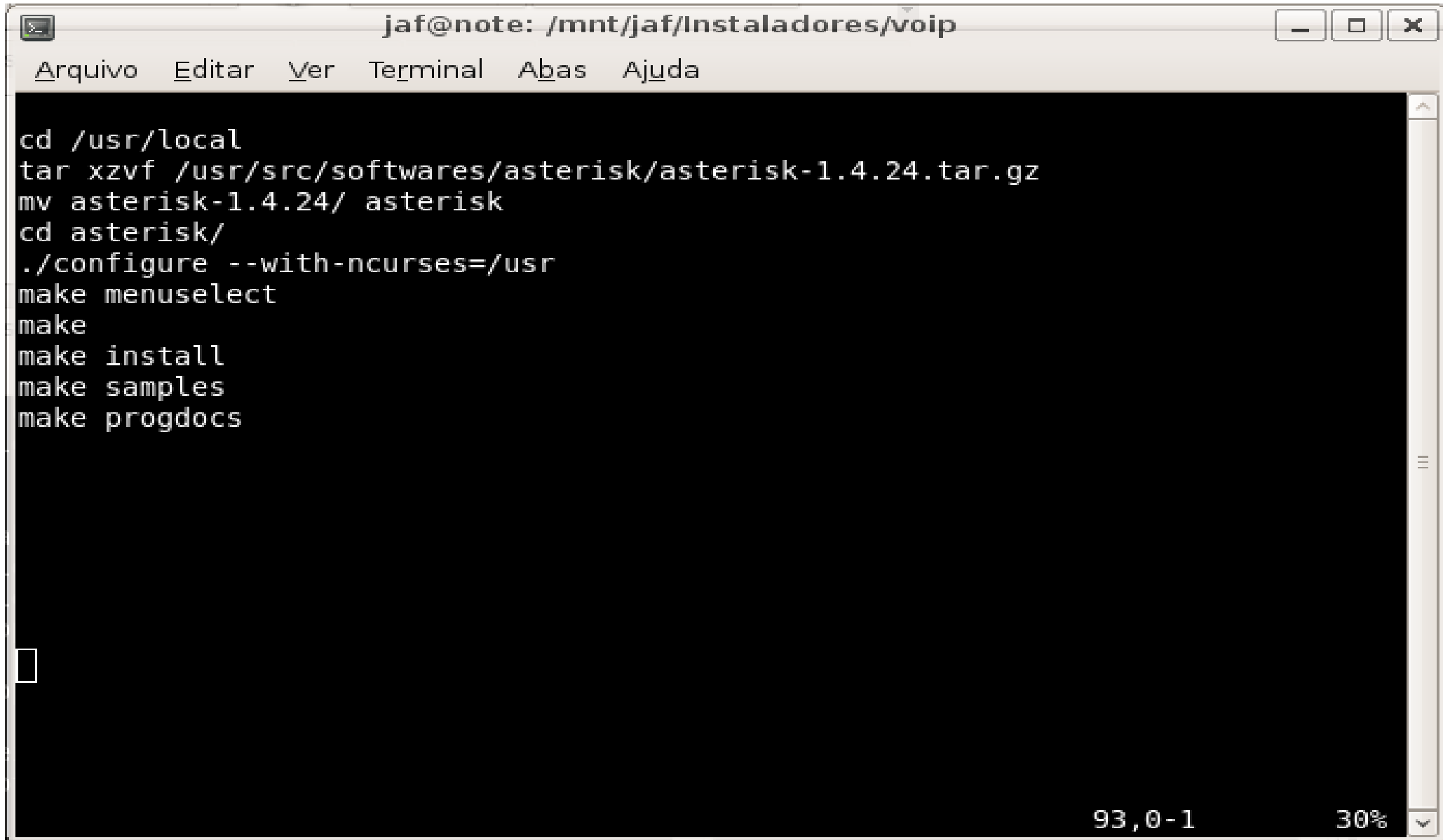
```
# Download em: http://downloads.digivoice.com.br
tar xzvf softwares/asterisk/voicerlib-4.1.0.2.tar.gz
cd voicerlib-4.1.0.2/
make clean
make
make install
make config
```

39,0-1

11%

# Instalação Asterisk 1.4.24

## Contexto DigiVoice



A terminal window titled "jaf@note: /mnt/jaf/Instaladores/voip" with a menu bar containing "Arquivo", "Editar", "Ver", "Terminal", "Abas", and "Ajuda". The terminal displays the following commands and their output:

```
cd /usr/local
tar xzvf /usr/src/softwares/asterisk/asterisk-1.4.24.tar.gz
mv asterisk-1.4.24/ asterisk
cd asterisk/
./configure --with-ncurses=/usr
make menuselect
make
make install
make samples
make progdocs
```

The terminal status bar at the bottom right shows "93,0-1" and "30%".



# Instalação Asterisk 1.4.24

## Contexto DigiVoice

sssoal espera eh a pratic jaf@note: /mnt/jaf/Instaladores/voip

Arquivo Editar Ver Terminal Abas Ajuda

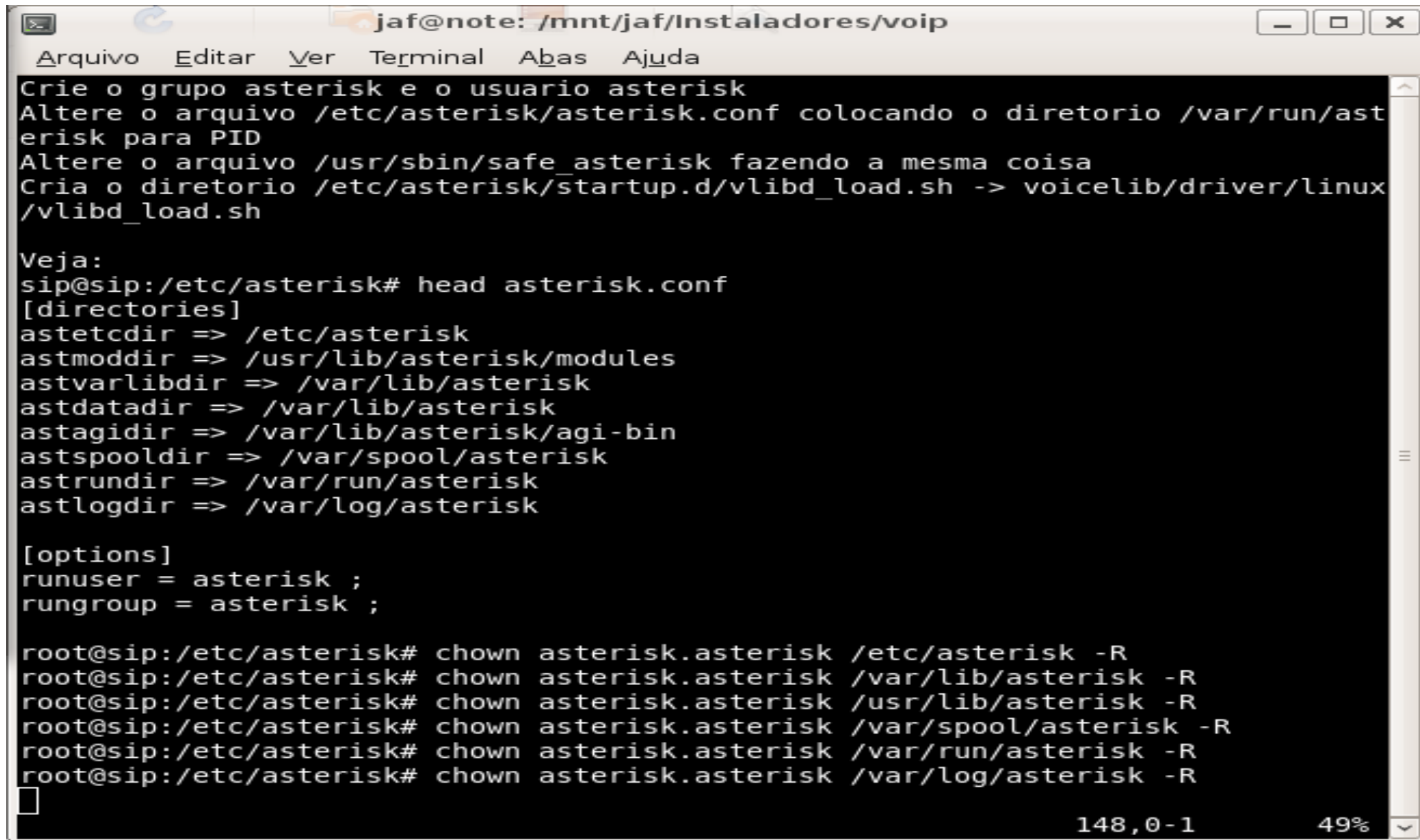
```
cd /usr/src
ln -s /usr/local/asterisk asterisk
tar xzvf softwares/asterisk/dgvchannel-1.0.2.tar.gz
cd dgvchannel-1.0.2/
./configure
make clean
make
make install
make install_config
```

98,0-1

39%

# Configurações Asterisk 1.4.24

## Contexto DigiVoice



```
jaf@note: /mnt/jaf/Instaladores/voip
Arquivo  Editar  Ver  Terminal  Abas  Ajuda
Crie o grupo asterisk e o usuario asterisk
Altere o arquivo /etc/asterisk/asterisk.conf colocando o diretorio /var/run/asterisk para PID
Altere o arquivo /usr/sbin/safe_asterisk fazendo a mesma coisa
Cria o diretorio /etc/asterisk/startup.d/vlibd_load.sh -> voicelib/driver/linux/vlibd_load.sh

Veja:
sip@sip:/etc/asterisk# head asterisk.conf
[directories]
astetcdir => /etc/asterisk
astmoddir => /usr/lib/asterisk/modules
astvarlibdir => /var/lib/asterisk
astdatadir => /var/lib/asterisk
astagidir => /var/lib/asterisk/agi-bin
astspooldir => /var/spool/asterisk
astrundir => /var/run/asterisk
astlogdir => /var/log/asterisk

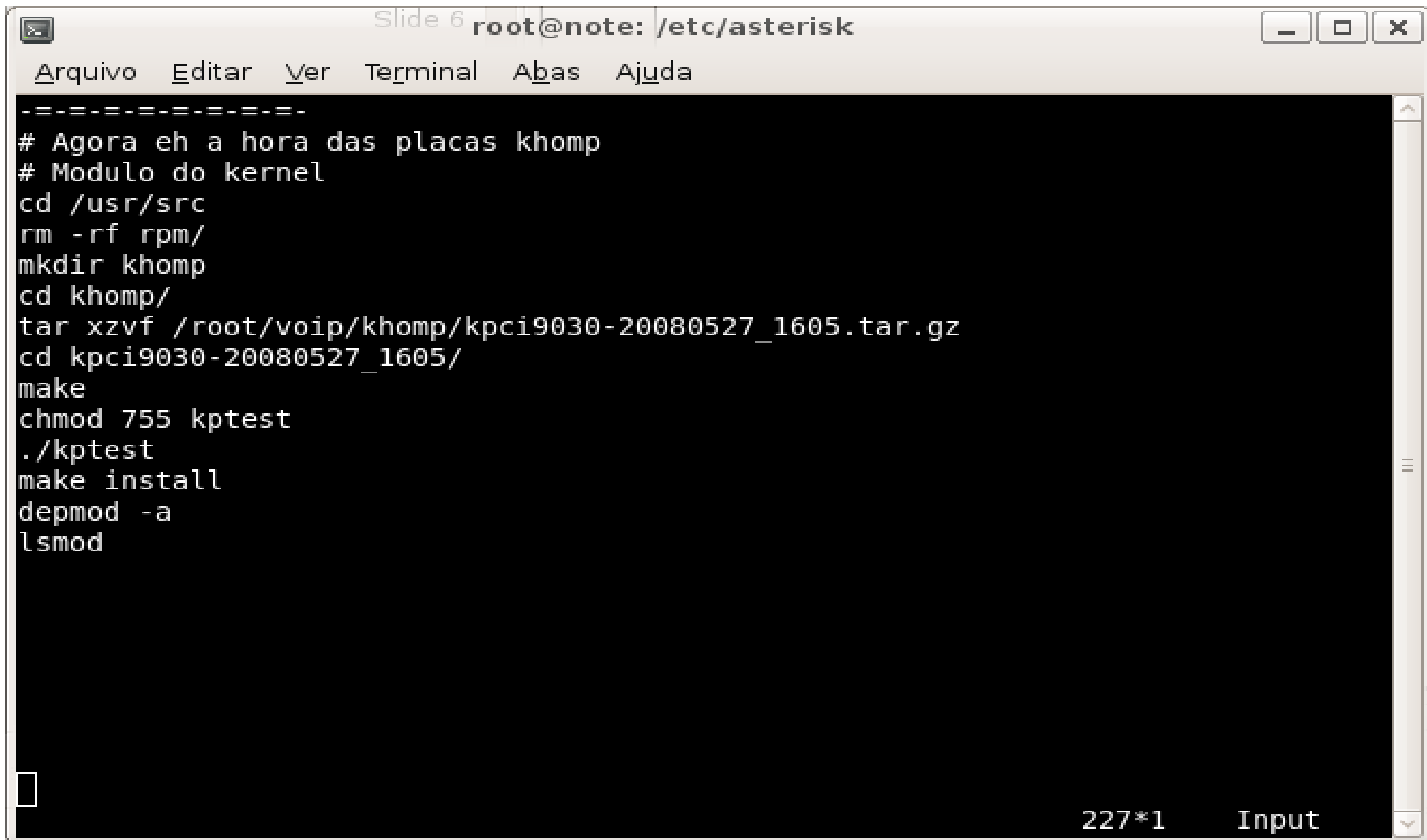
[options]
runuser = asterisk ;
runcgroup = asterisk ;

root@sip:/etc/asterisk# chown asterisk.asterisk /etc/asterisk -R
root@sip:/etc/asterisk# chown asterisk.asterisk /var/lib/asterisk -R
root@sip:/etc/asterisk# chown asterisk.asterisk /usr/lib/asterisk -R
root@sip:/etc/asterisk# chown asterisk.asterisk /var/spool/asterisk -R
root@sip:/etc/asterisk# chown asterisk.asterisk /var/run/asterisk -R
root@sip:/etc/asterisk# chown asterisk.asterisk /var/log/asterisk -R

148,0-1 49%
```

# Configurações Asterisk 1.4.24

## Contexto Khomp



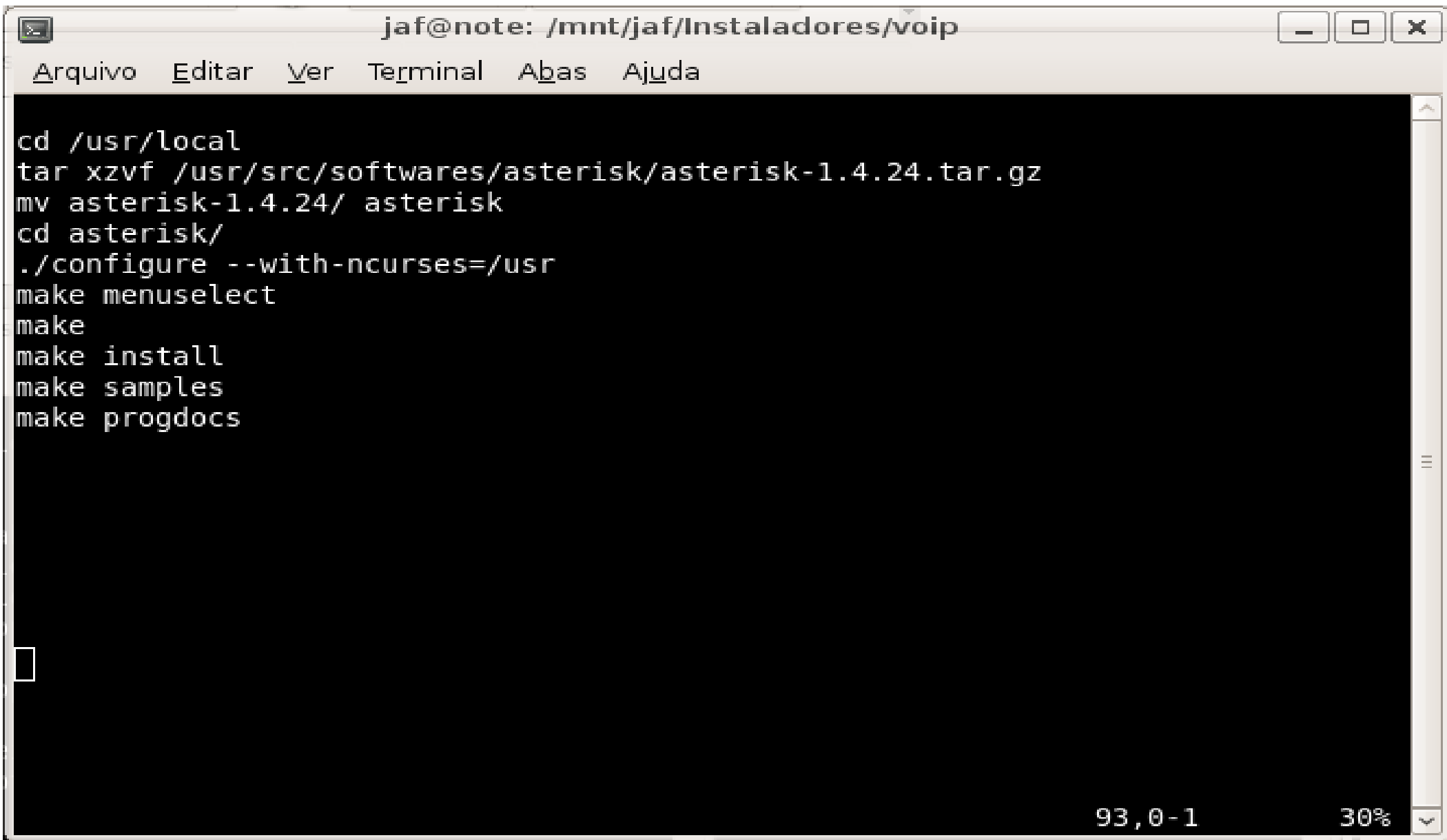
The image shows a terminal window titled "Slide 6 root@note: /etc/asterisk". The window contains a menu bar with "Arquivo", "Editar", "Ver", "Terminal", "Abas", and "Ajuda". The terminal output is as follows:

```
--==--  
# Agora eh a hora das placas khomp  
# Modulo do kernel  
cd /usr/src  
rm -rf rpm/  
mkdir khomp  
cd khomp/  
tar xzvf /root/voip/khomp/kpci9030-20080527_1605.tar.gz  
cd kpci9030-20080527_1605/  
make  
chmod 755 kptest  
./kptest  
make install  
depmod -a  
lsmod
```

At the bottom right of the terminal window, the text "227\*1" and "Input" is visible.

# Configurações Asterisk 1.4.24

## Contexto Khomp



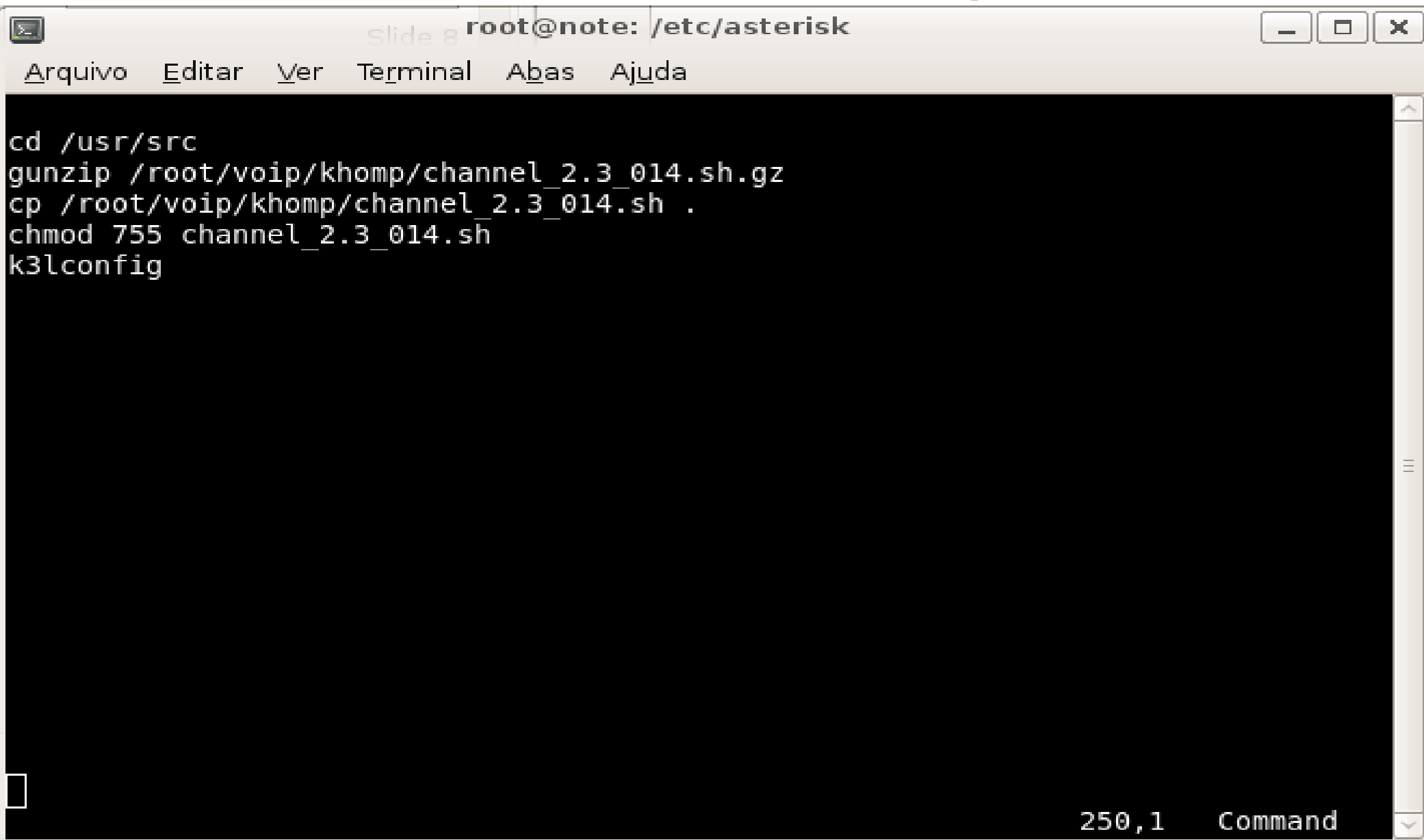
A terminal window titled "jaf@note: /mnt/jaf/Instaladores/voip" with a menu bar containing "Arquivo", "Editar", "Ver", "Terminal", "Abas", and "Ajuda". The terminal displays the following commands and their output:

```
cd /usr/local
tar xzvf /usr/src/softwares/asterisk/asterisk-1.4.24.tar.gz
mv asterisk-1.4.24/ asterisk
cd asterisk/
./configure --with-ncurses=/usr
make menuselect
make
make install
make samples
make progdocs
```

The terminal shows a cursor at the end of the last command. The bottom right corner of the terminal displays "93,0-1" and "30%".

# Configurações Asterisk 1.4.24

## Contexto Khomp



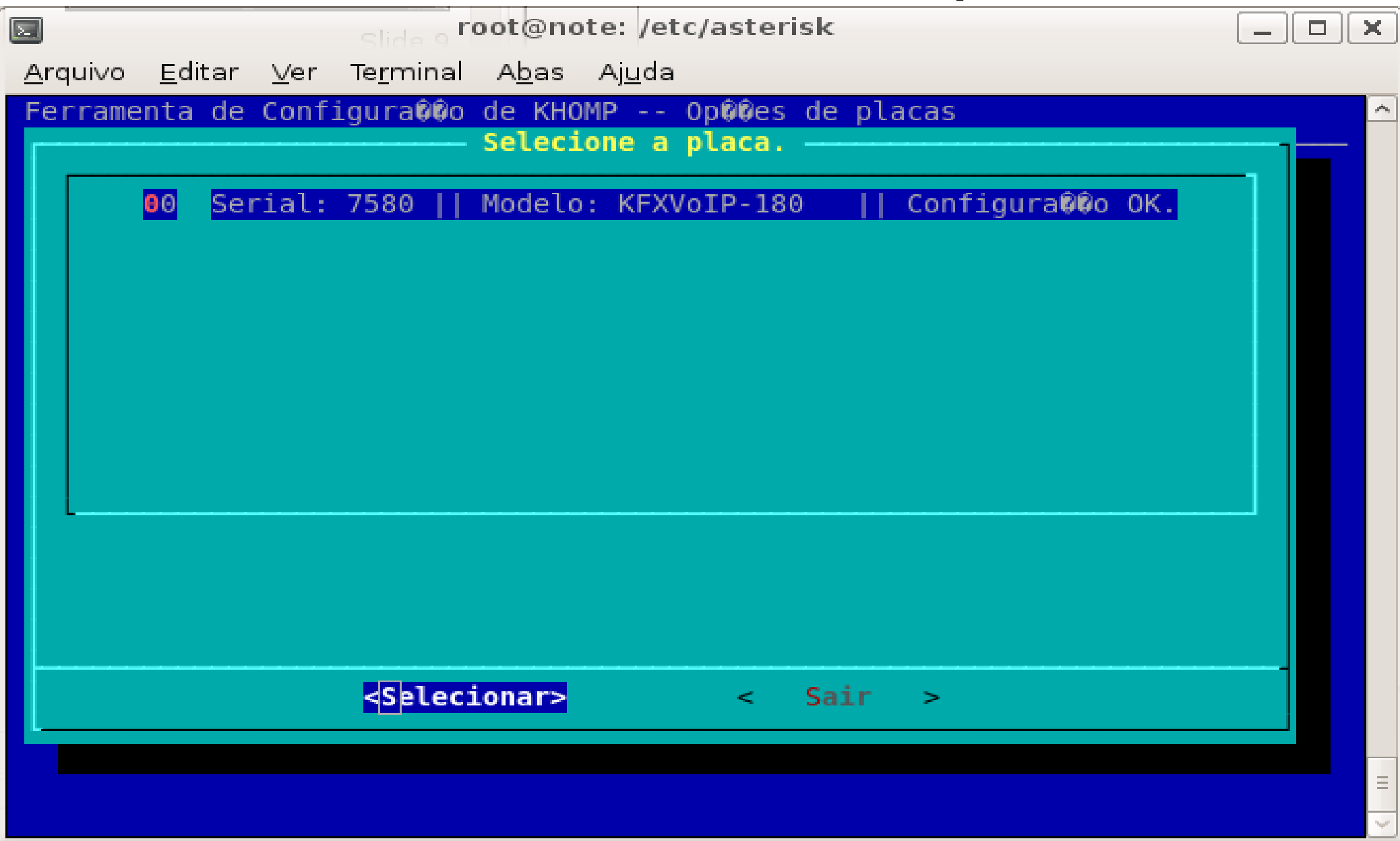
A terminal window titled "root@note: /etc/asterisk" with a menu bar containing "Arquivo", "Editar", "Ver", "Terminal", "Abas", and "Ajuda". The terminal displays the following commands:

```
cd /usr/src
gunzip /root/voip/khomp/channel_2.3_014.sh.gz
cp /root/voip/khomp/channel_2.3_014.sh .
chmod 755 channel_2.3_014.sh
k3lconfig
```

The terminal also shows a status bar at the bottom right with the coordinates "250,1" and the word "Command".

# Configurações Asterisk 1.4.24

## Contexto Khomp



# Configurações ramais SIP /etc/asterisk/sip.conf

Slide 9 root@note: /etc/asterisk

Arquivo Editar Ver Terminal Abas Ajuda

```
[102]
type=friend
secret=102
host=dynamic
dtmfmode=rfc2833
username=102
disallow=all
allow=ulaw
```

```
[103]
type=friend
secret=103
host=dynamic
dtmfmode=rfc2833
username=103
disallow=all
allow=ulaw
```

678\*1

Command

# Configurações troncos SIP /etc/asterisk/sip.conf

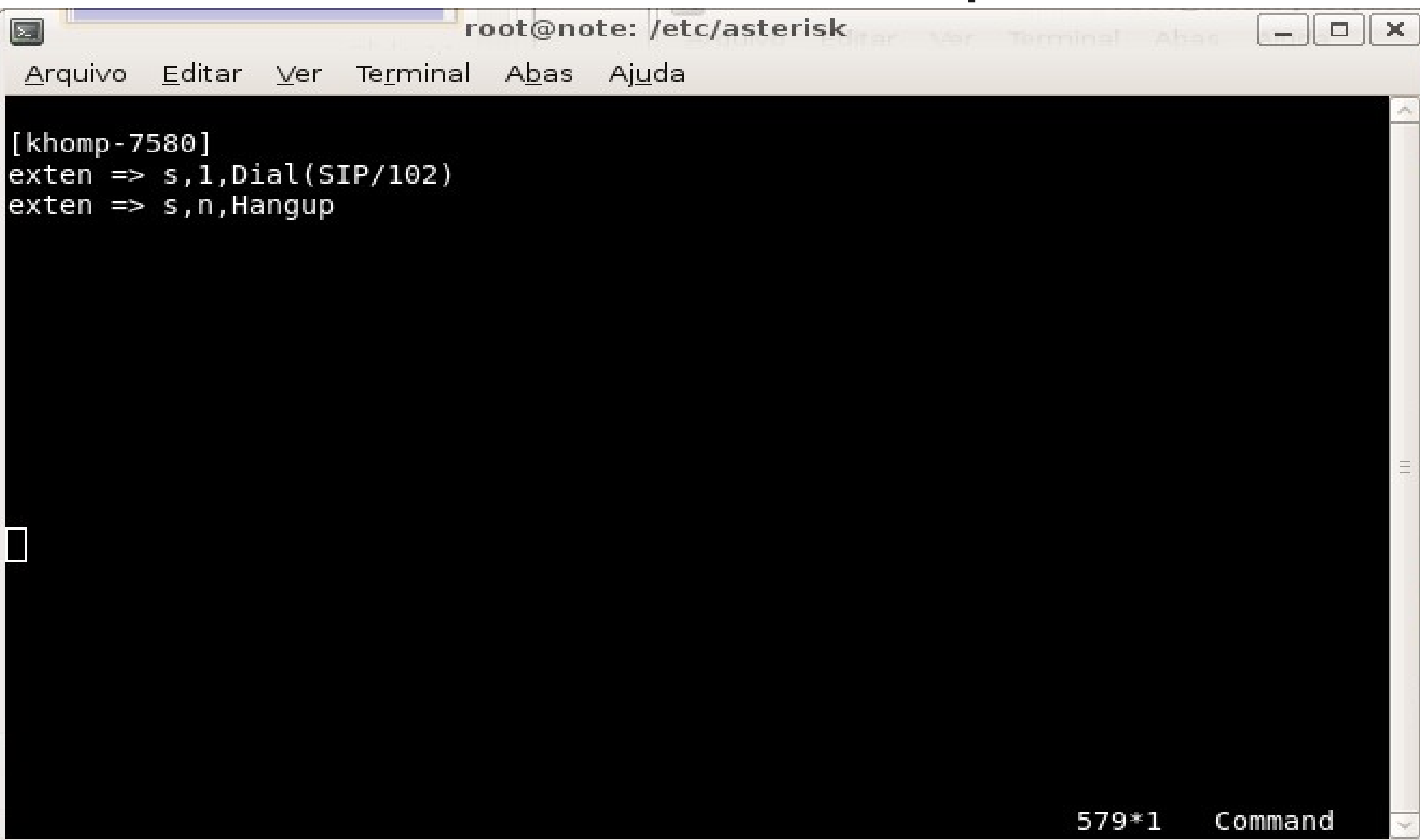
```
[TELLFREE]
username=7309197
type=peer
secret=12345678
qualify=no
port=5060
nat=yes
insecure=very
host=sip.tellfree.net
fromuser=7309197
fromdomain=sip.tellfree.net
dtmfmode=rfc2833
disallow=all
canreinvite=no
aut=md5
allow=g729
allow=ilbc

[VONO]
username=jaf_voip
type=peer
secret=12345678
reinvite=no
qualify=no
port=5060
nat=no
host=vono.net.br
fromuser=jaf_voip
fromdomain=vono.net.br
dtmfmode=rfc2833
domain=vono.net.br
disallow=all
canreinvite=no
allow=ilbc
```

170,0-1 Bot



# Configurações plano discagem Contexto Khomp

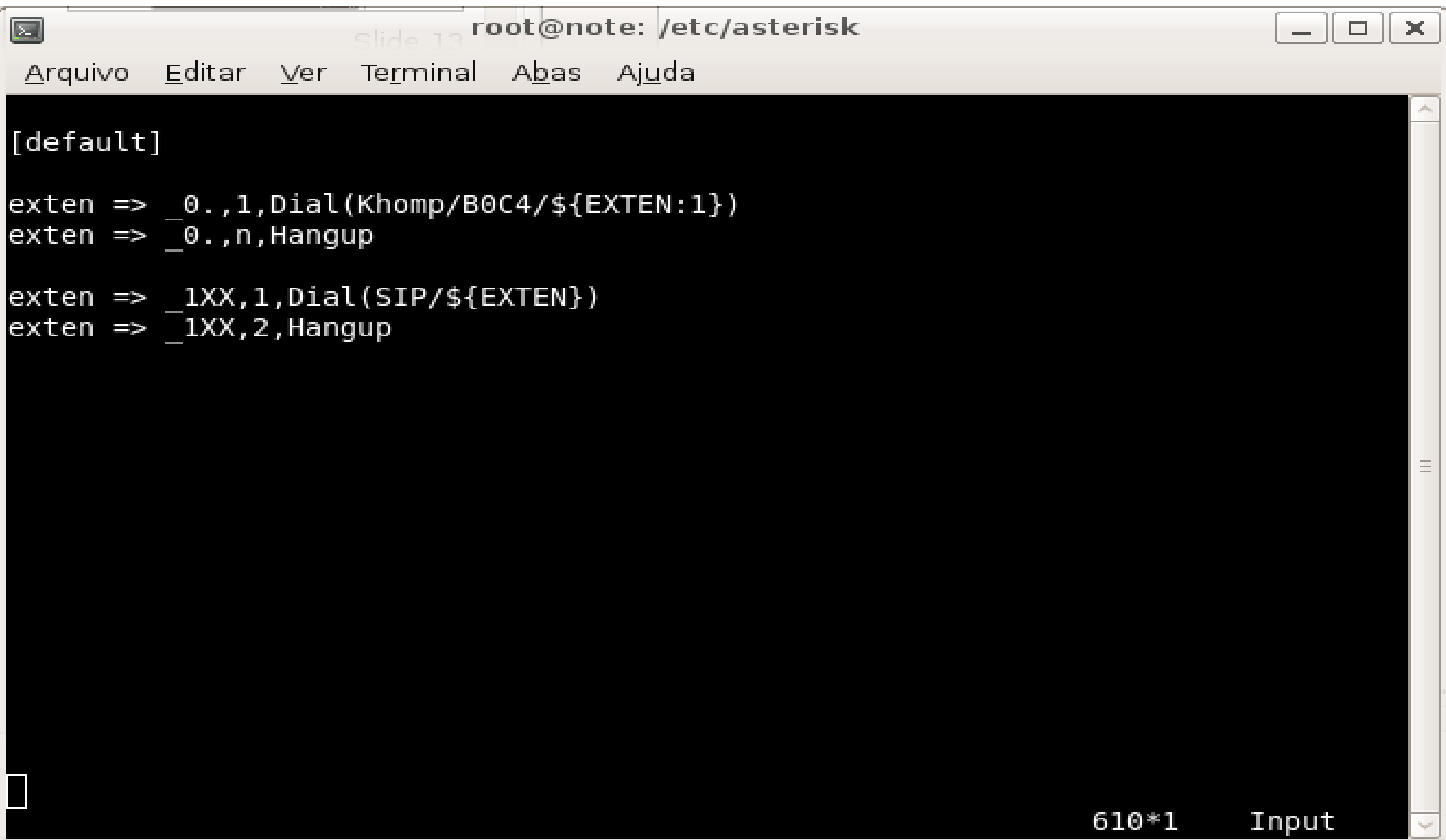


The image shows a terminal window titled "root@note: /etc/asterisk". The window has a menu bar with "Arquivo", "Editar", "Ver", "Terminal", "Abas", and "Ajuda". The terminal content is as follows:

```
[khomp-7580]  
exten => s,1,Dial(SIP/102)  
exten => s,n,Hangup
```

At the bottom right of the terminal window, the text "579\*1 Command" is visible.

# Configurações plano de discagem Contexto Khomp



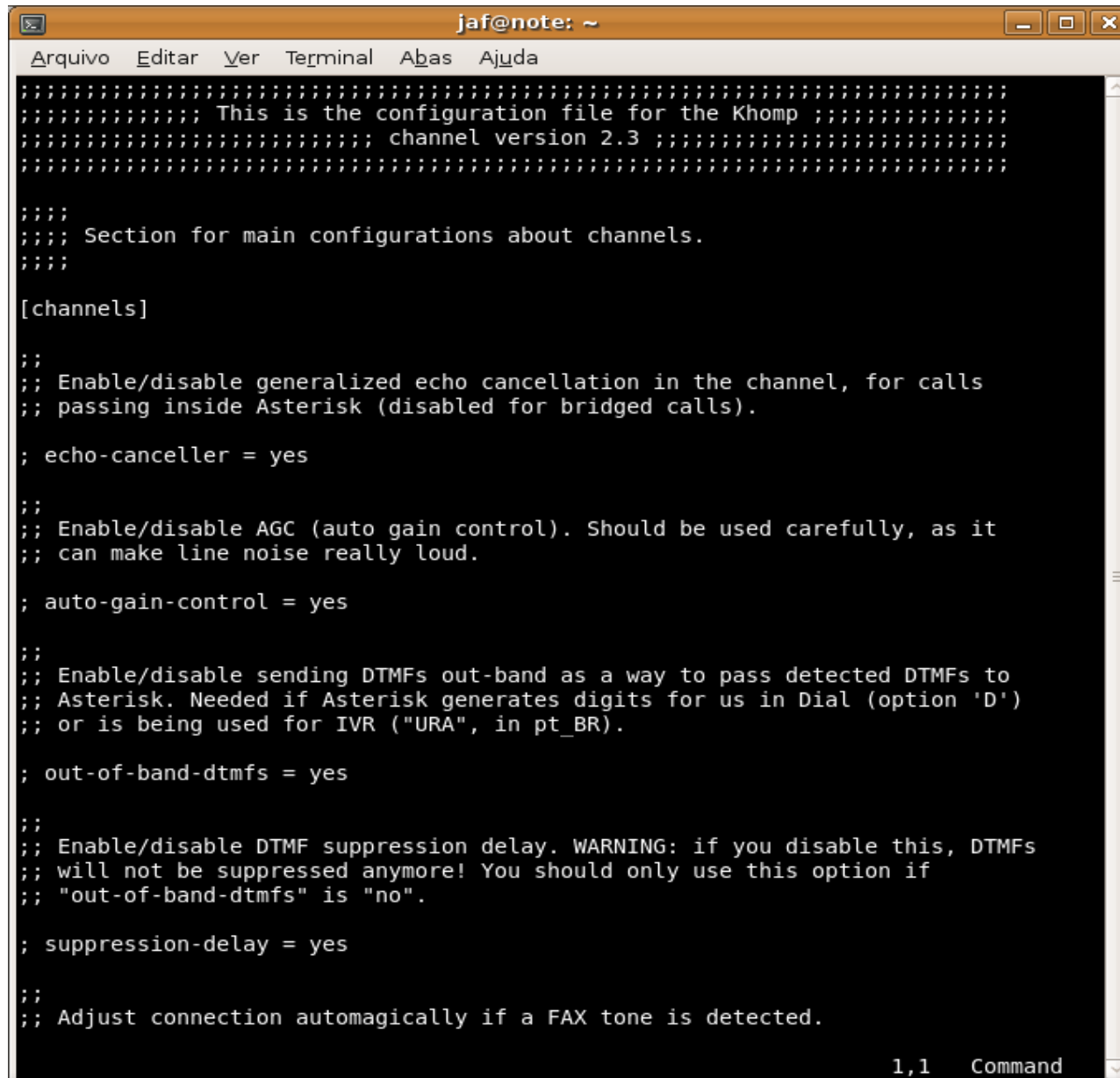
The image shows a terminal window titled "root@note: /etc/asterisk" with a menu bar containing "Arquivo", "Editar", "Ver", "Terminal", "Abas", and "Ajuda". The terminal content displays the configuration for the [default] context in Asterisk, including extension patterns and actions.

```
[default]
exten => _0.,1,Dial(Khomp/B0C4/${EXTEN:1})
exten => _0.,n,Hangup

exten => _1XX,1,Dial(SIP/${EXTEN})
exten => _1XX,2,Hangup
```

At the bottom right of the terminal window, the text "610\*1" and "Input" is visible.

# Arquivo de configuração placas Khomp



```
jaf@note: ~
Arquivo  Editar  Ver  Terminal  Abas  Ajuda
::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::
::::::::::::::::::::::::::::::::::::::::::::; This is the configuration file for the Khomp ;::::::::::::::::::::::::::::::::::::::::::::
::::::::::::::::::::::::::::::::::::::::::::; channel version 2.3 ;::::::::::::::::::::::::::::::::::::::::::::
::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::::

;;;
;;; Section for main configurations about channels.
;;;

[channels]

;;
;; Enable/disable generalized echo cancellation in the channel, for calls
;; passing inside Asterisk (disabled for bridged calls).

; echo-canceller = yes

;;
;; Enable/disable AGC (auto gain control). Should be used carefully, as it
;; can make line noise really loud.

; auto-gain-control = yes

;;
;; Enable/disable sending DTMFs out-band as a way to pass detected DTMFs to
;; Asterisk. Needed if Asterisk generates digits for us in Dial (option 'D')
;; or is being used for IVR ("URA", in pt_BR).

; out-of-band-dtmfs = yes

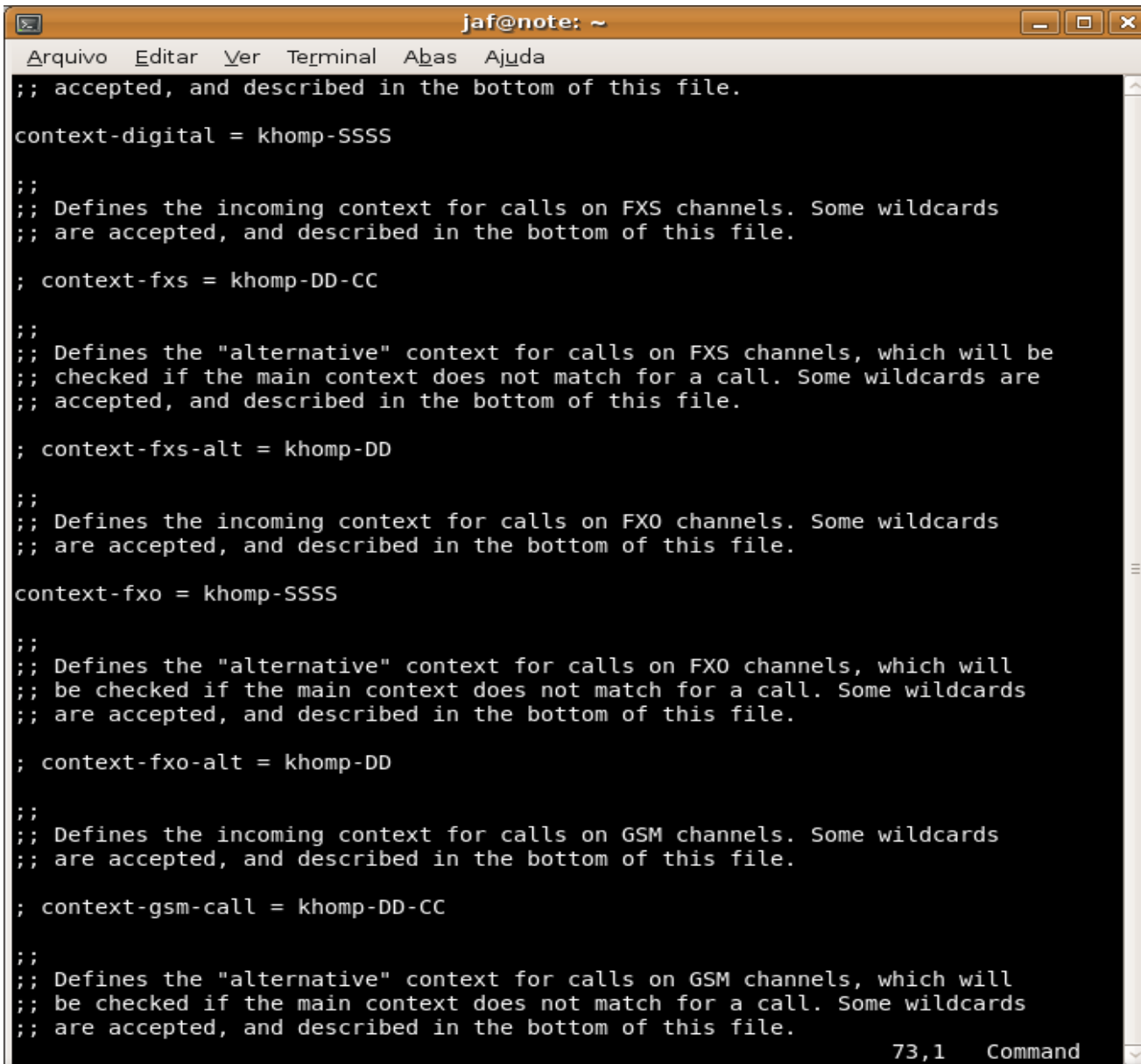
;;
;; Enable/disable DTMF suppression delay. WARNING: if you disable this, DTMFs
;; will not be suppressed anymore! You should only use this option if
;; "out-of-band-dtmfs" is "no".

; suppression-delay = yes

;;
;; Adjust connection automagically if a FAX tone is detected.

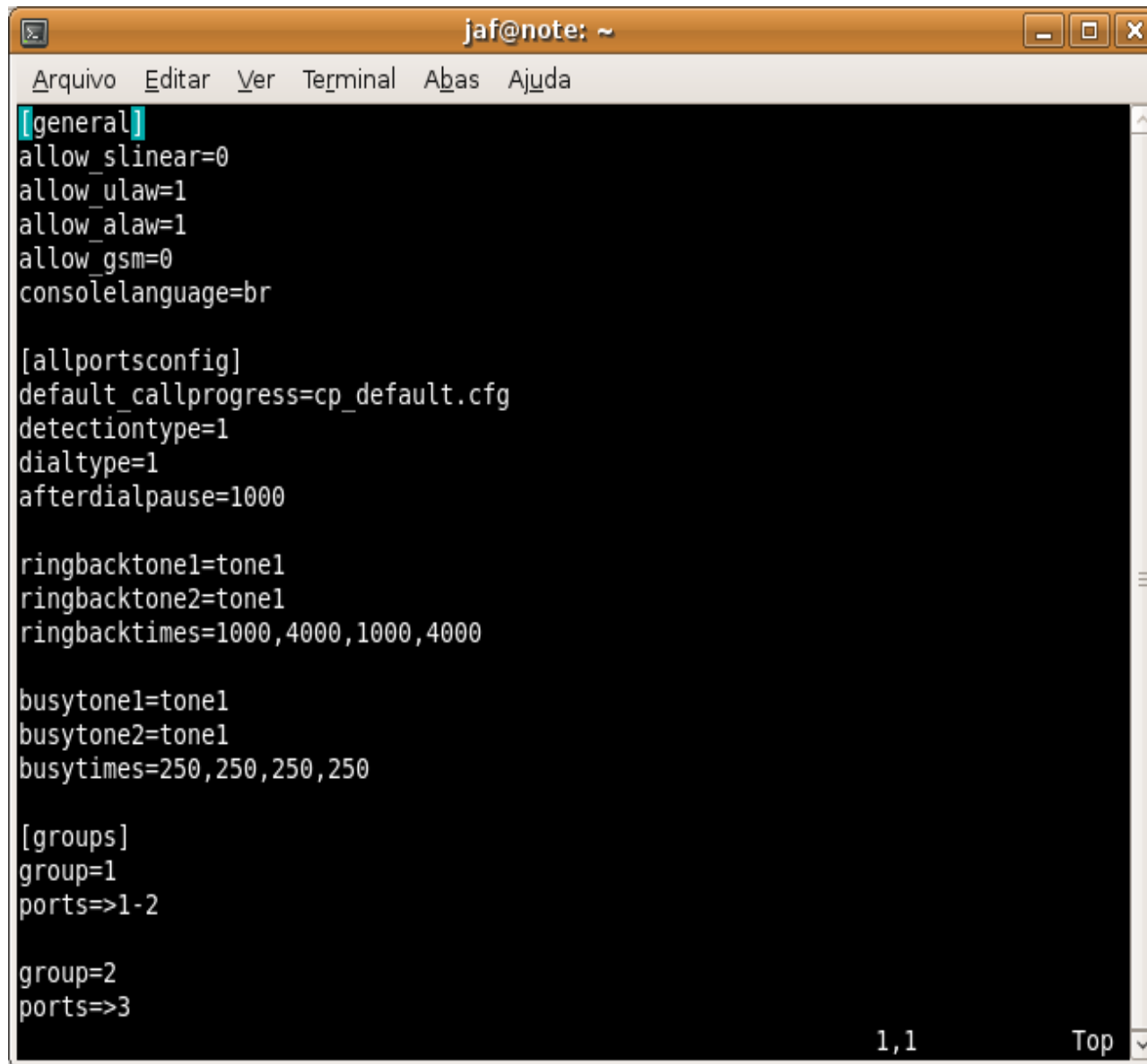
1,1 Command
```

# Arquivos de configuração placas Khomp



```
jaf@note: ~
Arquivo  Editar  Ver  Terminal  Abas  Ajuda
;; accepted, and described in the bottom of this file.
context-digital = khomp-SSSS
;;
;; Defines the incoming context for calls on FXS channels. Some wildcards
;; are accepted, and described in the bottom of this file.
; context-fxs = khomp-DD-CC
;;
;; Defines the "alternative" context for calls on FXS channels, which will be
;; checked if the main context does not match for a call. Some wildcards are
;; accepted, and described in the bottom of this file.
; context-fxs-alt = khomp-DD
;;
;; Defines the incoming context for calls on FXO channels. Some wildcards
;; are accepted, and described in the bottom of this file.
context-fxo = khomp-SSSS
;;
;; Defines the "alternative" context for calls on FXO channels, which will
;; be checked if the main context does not match for a call. Some wildcards
;; are accepted, and described in the bottom of this file.
; context-fxo-alt = khomp-DD
;;
;; Defines the incoming context for calls on GSM channels. Some wildcards
;; are accepted, and described in the bottom of this file.
; context-gsm-call = khomp-DD-CC
;;
;; Defines the "alternative" context for calls on GSM channels, which will
;; be checked if the main context does not match for a call. Some wildcards
;; are accepted, and described in the bottom of this file.
73,1  Command
```

# Arquivo de configuração placas DigiVoice



The image shows a terminal window titled "jaf@note: ~" with a menu bar containing "Arquivo", "Editar", "Ver", "Terminal", "Abas", and "Ajuda". The terminal displays the following configuration text:

```
[general]
allow_slinear=0
allow_ulaw=1
allow_alaw=1
allow_gsm=0
consolelanguage=br

[allportsconfig]
default_callprogress=cp_default.cfg
detectiontype=1
dialtype=1
afterdialpause=1000

ringbacktone1=tone1
ringbacktone2=tone1
ringbacktimes=1000,4000,1000,4000

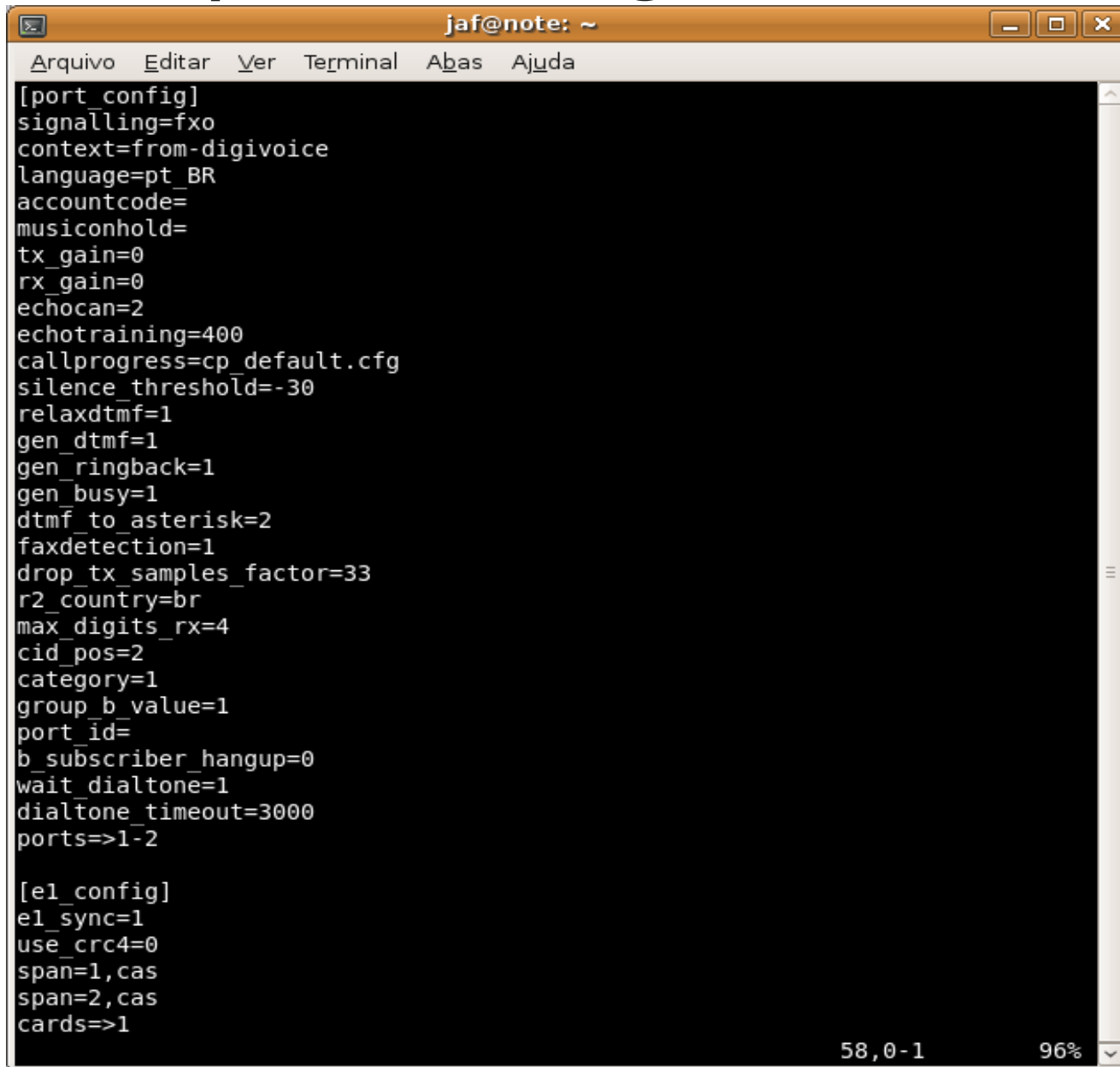
busytone1=tone1
busytone2=tone1
busytimes=250,250,250,250

[groups]
group=1
ports=>1-2

group=2
ports=>3
```

At the bottom right of the terminal, the text "1,1" and "Top" are visible.

# Arquivos de configuração placas DigiVoice



The image shows a terminal window titled "jaf@note: ~" with a menu bar containing "Arquivo", "Editar", "Ver", "Terminal", "Abas", and "Ajuda". The terminal displays two configuration sections: "[port\_config]" and "[e1\_config]".

```
[port_config]
signalling=fxo
context=from-digivoice
language=pt_BR
accountcode=
musiconhold=
tx_gain=0
rx_gain=0
echocan=2
echotraining=400
callprogress=cp_default.cfg
silence_threshold=-30
relaxdtmf=1
gen_dtmf=1
gen_ringback=1
gen_busy=1
dtmf_to_asterisk=2
faxdetection=1
drop_tx_samples_factor=33
r2_country=br
max_digits_rx=4
cid_pos=2
category=1
group_b_value=1
port_id=
b_subscriber_hangup=0
wait_dialtone=1
dialtone_timeout=3000
ports=>1-2

[e1_config]
e1_sync=1
use_crc4=0
span=1,cas
span=2,cas
cards=>1
```

At the bottom right of the terminal, the text "58,0-1" and "96%" is visible.

# Criando um plano de discagem

- × O plano de discagem é considerado por muito a parte mais importante de um sistema Asterisk;
- × O arquivo `/etc/asterisk/extensions.conf` especifica o plano de discagem no Asterisk;
- × O plano de discagem é composto por 4 elementos:
  - × *Contextos;*
  - × *Extensões;*
  - × *Prioridades;*
  - × *Aplicações;*

# Criando um plano de discagem

- x Contextos
  - x Responsável pela organização e escopo do plano de discagem;
  - x Quando uma ligação entra no Asterisk por um canal, ela é processada dentro de um contexto;
  - x Os contexto estão ligados diretamente aos canais;
  - x Exemplificando ...
  - x Temos dois contextos [gerentes] e [adm]. O primeiro é permitido ligações longa distância, enquanto o segundo não;



# Criando um plano de discagem

- x Extensões

- x É uma **instrução** que o Asterisk segue, acionada por uma chamada de entrada ou por dígitos discados em um canal;

- x A declaração de uma extensão possui o seguinte formato:

- x *exten => número,prioridade,aplicação*

- x Exemplo:

- x `exten => 1001,1,Answer( )`

- x `exten => 1001,n,Hangup( )`

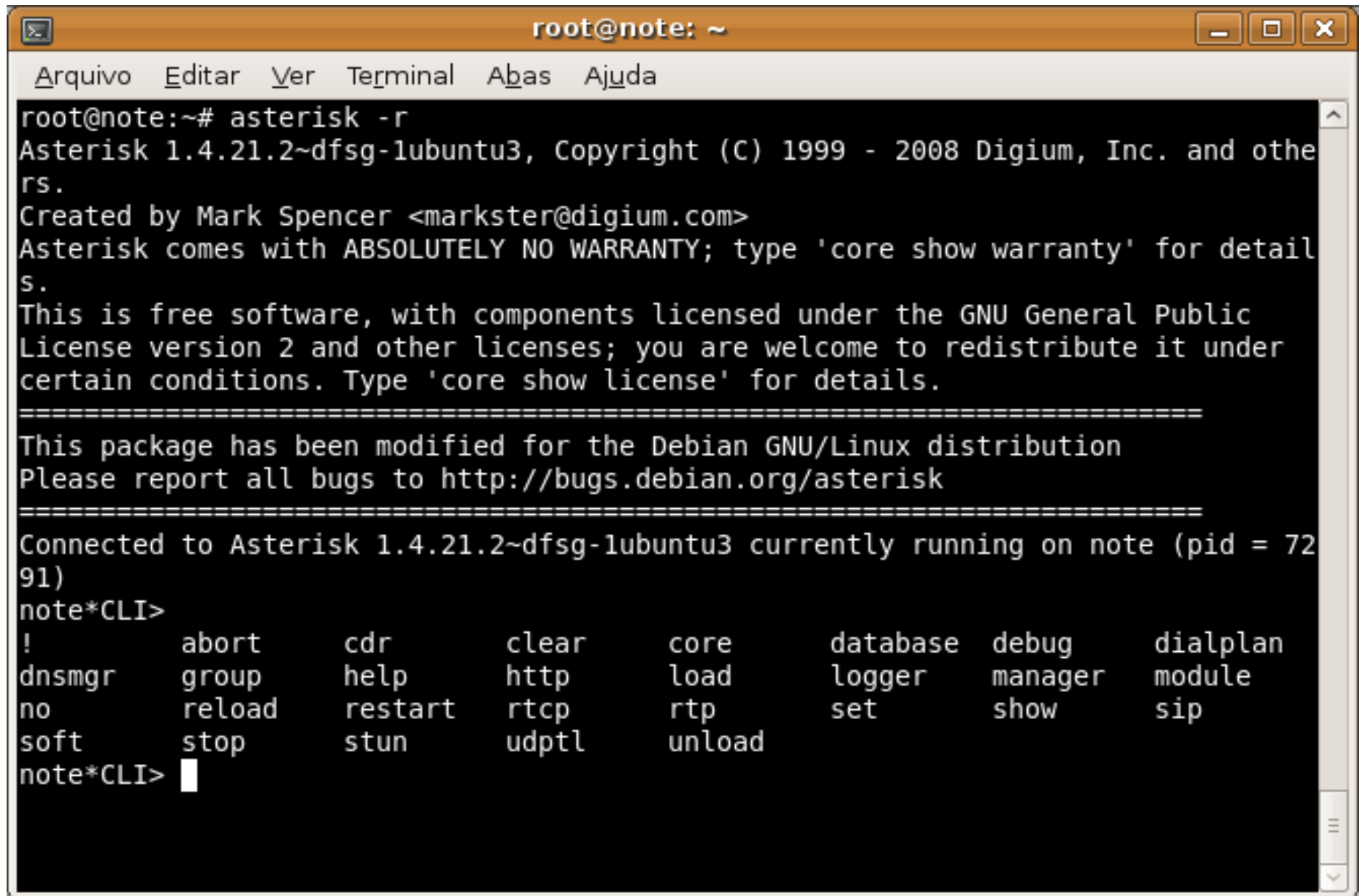
# Criando um plano de discagem

- x Padrões de extensão
- x Os seguintes caracteres são exemplos de padrões:
  - x X – Dígitos de 0 até 9
  - x Z – Dígitos de 1 até 9
  - x N – Dígitos de 2 até 9
  - x [1237-9] – Qualquer dígito entre as chaves e o intervalo 7-9, neste caso 1, 2, 3, 7, 8 e 9;
  - x . (ponto) é um **curinga** que combina com um ou mais dígitos;

# Utilizando Asterisk/CLI

- x CLI – Command Line Interface
- x É uma interface que permite o administrador realizar comandos diretamente ao Asterisk. Os comandos são divididos em grupo. Os principais são:
  - x Comandos de uso geral;
  - x Gerenciamento servidor Asterisk;
  - x Comando manipulação canal IAX, SIP, H323, ZAP e outros, inclusive canais DGV e Khomp;
- x Para acessar Asterisk/CLI use o comando:
  - x *# rasterisk* ou *asterisk -r*

# Utilizando Asterisk/CLI



```
root@note: ~
Arquivo  Editar  Ver  Terminal  Abas  Ajuda
root@note:~# asterisk -r
Asterisk 1.4.21.2~dfsg-1ubuntu3, Copyright (C) 1999 - 2008 Digium, Inc. and others.
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'core show warranty' for details.
This is free software, with components licensed under the GNU General Public License version 2 and other licenses; you are welcome to redistribute it under certain conditions. Type 'core show license' for details.
=====
This package has been modified for the Debian GNU/Linux distribution
Please report all bugs to http://bugs.debian.org/asterisk
=====
Connected to Asterisk 1.4.21.2~dfsg-1ubuntu3 currently running on note (pid = 7291)
note*CLI>
!          abort      cdr        clear      core       database   debug      dialplan
dnsmgr     group      help       http       load       logger     manager    module
no         reload    restart    rtcpl      rtp        set        show       sip
soft       stop      stun       udptl     unload
note*CLI> █
```

# Instalando clientes SIP

The screenshot shows the Synaptic Package Manager window. The search bar contains 'ekiga'. The package list shows 'ekiga' as the selected package. The description pane for 'ekiga' is visible, showing its version and a brief description.

**Gerenciador de Pacotes Synaptic**

Arquivo Editar Pacote Configurações Ajuda

Recarregar Marcar todas atualizações Aplicar Propriedades Recriando índice de pe... Procurar

S	Pacote	Versão instalada	Versão recente	Descrição
<input checked="" type="checkbox"/>	ekiga	2.0.12-0ubuntu5	2.0.12-0ubuntu5	H.323 and SIP compatible VoIP client
<input type="checkbox"/>	gnomemeeting		2.0.12-0ubuntu5	Dummy transition package of GnomeMeeting for Ekiga
<input type="checkbox"/>	ekiga-dbg		2.0.12-0ubuntu5	H.323 and SIP compatible VoIP client - debug symbols
<input type="checkbox"/>	ohphone-basic		1:1.4.5+20060204-	Command line H.323 client with SDL support
<input type="checkbox"/>	ekiga-gtkonly		2.0.12-0ubuntu5	H.323 and SIP compatible VoIP client - GTK-only (GNOME-free)

**H.323 and SIP compatible VoIP client**

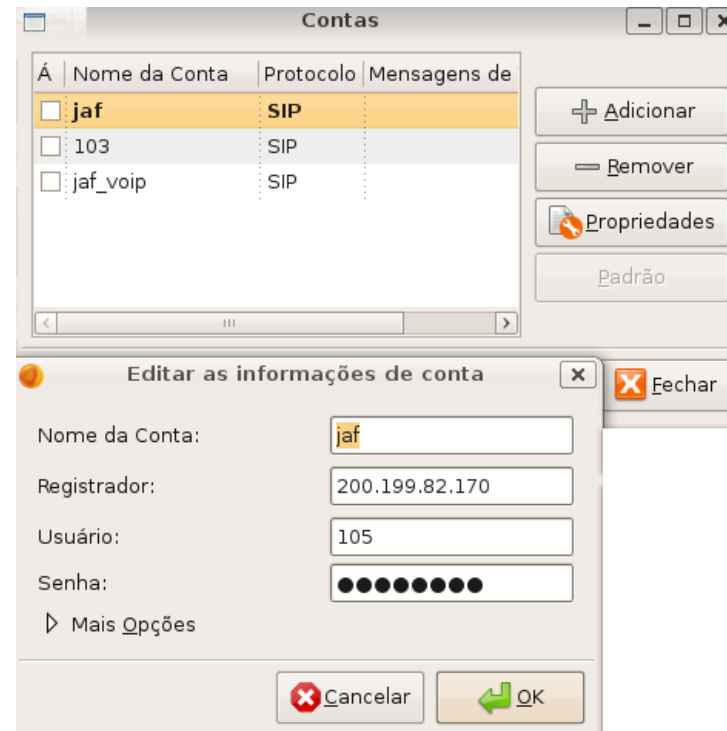
H.323 and SIP compatible videoconferencing and VoIP/IP-Telephony application that allows you to make audio and video calls to remote users with H.323 hardware or software (such as Microsoft Netmeeting) as well as SIP endpoints.

It supports all modern videoconferencing features, such as registering to an LDAP directory, gatekeeper support, making multi-user conference calls using an external MCU, using modern Quicknet telephony cards, and making PC-To-Phone calls.

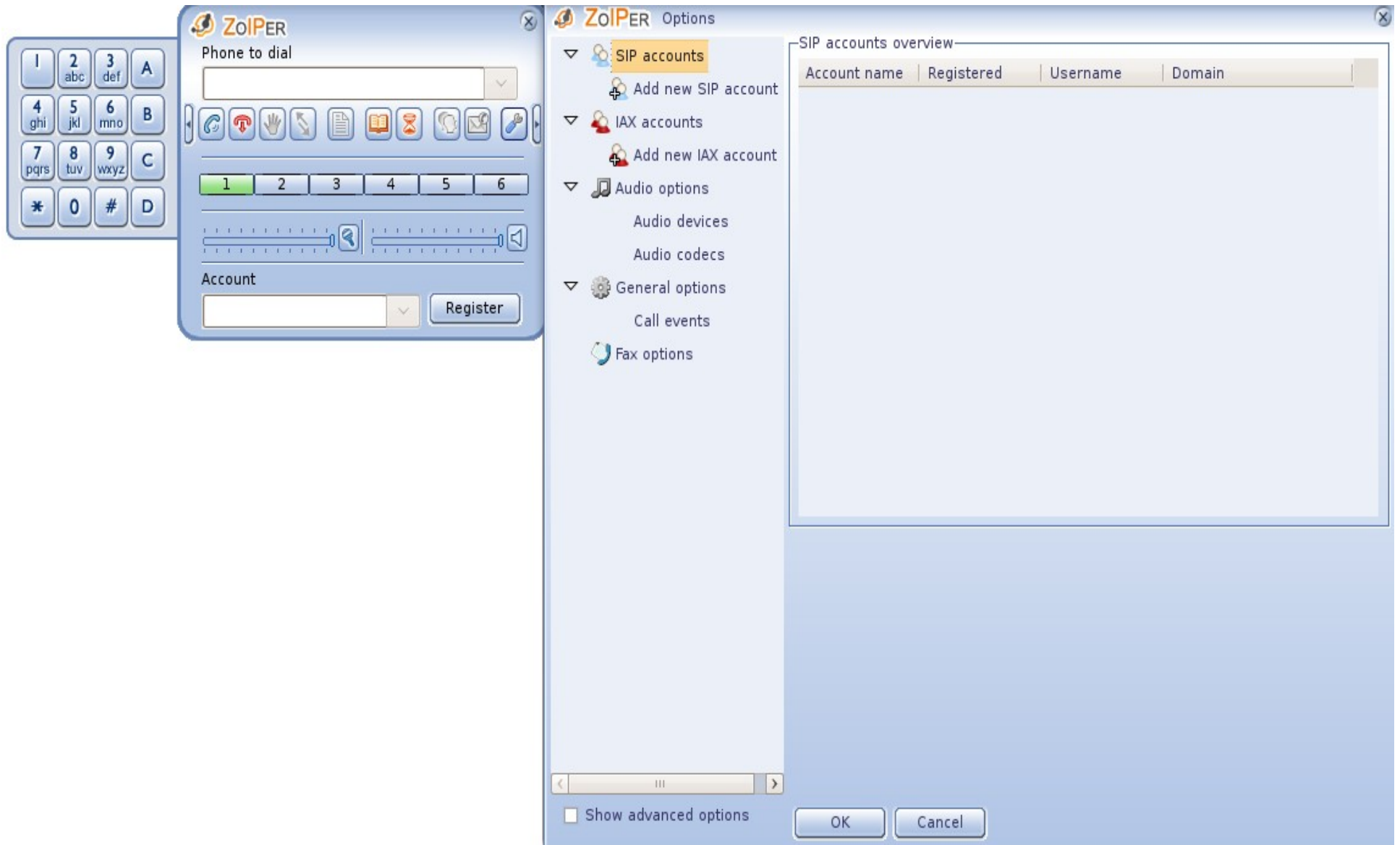
Canonical oferece atualizações críticas para ekiga até maio 2010.

5 pacotes listados, 1688 instalados, 0 quebrados. 0 para instalar/atualizar, 0 para remover

# Instalando clientes SIP



# Instalando clientes SIP



# Roteiro de atividades

- 1) Instalar Asterisk;
- 2) Configurar ramais SIP (pelo menos dois);
- 3) Configurar as extensões permitindo os ramais se comunicarem;
- 4) Utilizar o Asterisk/CLI para mapear as ligações;
- 5) Utilize Wireshark para mapear os pacotes na rede dos procedimentos com o cliente SIP;