

Solution #1) Asterisk GUI (by Digium, Inc.)

Solution #2) Asterisk Management Script (by fivn) ????? DD-WRT ?????????????????????????????????

Asterisk GUI

????? <http://www.asterisk.org/asterisknow/...pers/gui-guide>

?? Asterisk-GUI

? SSH ?????????? root ??

```
#> /opt/bin/ipkg-opt install asterisk-gui
```

?? Asterisk-GUI

```
#> vi /opt/etc/asterisk/manager.conf
```

```
[general]
displayssystemname = yes
enabled = yes
webenabled = yes
port = 5038
httptimeout = 60
...
; for Asterisk-GUI
[admin]
secret = HereisYourPassword
deny = 0.0.0.0/0.0.0.0
permit = 192.168.1.1/255.255.255.0
read = system,call,log,verbose,command,agent,user,config
write = system,call,log,verbose,command,agent,user,config
```

??192.168.1.1 ?????? IP ??

```
#> vi /opt/etc/asterisk/http.conf
```

```
[general]
enabled = yes
enablestatic = yes
prefix = gui
bindaddr = 0.0.0.0
bindport = 8088
```

??bindaddr ?????????????????? IP ???0.0.0.0 ????????

Asterisk????

Notes: autoload ???? yes??? Asterisk-GUI ????????

```
# vi /opt/etc/asterisk/modules.conf
```

```
;
; Asterisk configuration file
;
; Module Loader configuration file By Leif Madsen of www.leifmadsen.com
; Descriptions and some modules added by Bill Weidman
```

```
[modules]
autoload=yes
```

```
;Unload modules
noload => cdr_odbc.so
noload => cdr_radius.so
noload => cdr_sqlite.so
noload => chan_gtalk.so
noload => codec_speex.so
noload => format_ogg_vorbis.so
noload => func_odbc.so
noload => res_config_odbc.so
noload => res_odbc.so
noload => res_snmp.so
noload => res_jabber.so
```

```
; Resources —
;load => res_adi.so ; ADSI Resource
;load => res_agi.so ; Asterisk Gateway Interface (AGI)
;load => res_config_mysql.so ; MySQL Configuration
;load => res_config_odbc.so ; ODBC Configuration
;load => res_crypto.so ; Cryptographic Digital Signatures
load => res_features.so ; Call Parking Resource
;load => res_indications.so ; Indications Configuration
;load => res_monitor.so ; Call Monitoring Resource
load => res_musiconhold.so ; Music On Hold Resource
;load => res_odbc.so ; ODBC Resource
```

```
; PBX —
;load => pbx_ael.so ; Asterisk Extension Language Compiler - Requires ?
load => pbx_config.so ; Text Extension Configuration Requires N/A
;load => pbx_dundi.so ; Do a DUNDi lookup of a phone number. - Requires res_crypto.so
;load => pbx_functions.so ; Builtin dialplan functions - Requires N/A
;load => pbx_loopback.so ; Loopback Dialplan Switch - Requires N/A
;load => pbx_realtime.so ; Realtime Dialplan Switch - Requires N/A
;load => pbx_spool.so ; Outgoing Spool Support Requires - N/A
```

```
; Functions —
load => func_callerid.so ; Gets or sets Caller*ID data on the channel. - Requires ?
;load => func_enum.s ; ENUMLOOKUP and TXTCIDNAME functions - Requires ?
```

```
;load => func_uri.so ; URI encode/decode functions - Requires ?
load => func_logic.so ; Logical dialplan functions
```

```
; Database Call Detail Records —
```

```
;load => cdr_addon_mysql.so ; Mysql CDR Backend - Requires ?
;load => cdr_csv.so ; Comma Separated Values CDR Backend - Requires N/A
;load => cdr_custom.so ; Customizable Comma Separated Values CDR Backend - Requires N/A
;load => cdr_manager.so ; Asterisk Call Manager CDR Backend - Requires N/A
;load => cdr_odbc.so ; ODBC CDR Backend - Requires N/A
;load => cdr_pgsqll.so ; PostgreSQL CDR Backend - Requires N/A
;load => cdr_sqlite.so ; SQLite CDR Backend - Requires N/A
```

```
; Channels —
```

```
;load => chan_agent.so ; Agent Proxy Channel - Requires res_features.so, res_monitor.so,
res_musiconhold.so
;load => chan_features.so ; Provides summary information on feature channels- Requires N/A
load => chan_iax2.so ; Inter Asterisk eXchange (Ver 2) - Requires res_crypto.so, res_features.so
;load => chan_local.so ; Show status of local channels- Requires N/A
;load => chan_mgcp.so ; Media Gateway Control Protocol (MGCP) - Requires res_features.so
;load => chan_modem.so ; Generic Voice Modem Driver - Requires N/A
;load => chan_modem_aopen.so ; A/Open (Rockwell Chipset) ITU-2 VoiceMod- Requires chan_modem.so
;load => chan_modem_bestdata.so ; BestData (Conexant V.90 Chipset) VoiceModem - Requires
chan_modem.so
;load => chan_modem_i4l.so ; ISDN4Linux Emulated Modem Driver - Requires chan_modem.so
;load => chan_oss.so ; OSS Console Channel Driver - Requires N/A
;load => chan_phone.so ; Linux Telephony API Support - Requires N/A
load => chan_sip.so ; Session Initiation Protocol (SIP) - Requires res_features.so
;load => chan_skinny.so ; Skinny Client Control Protocol (Skinny) - Requires res_features.so
;load => chan_zap.so ; Zapata Telephony w/PRI - Requires ?
```

```
; Codecs —
```

```
;load => codec_adpcm.so ; Adaptive Differential PCM Coder/Decoder - Requires N/A
load => codec_alaw.so ; A-law Coder/Decoder - Requires N/A
;load => codec_a_mu.so ; A-law and Mulaw direct Coder/Decoder - Requires N/A
;load => codec_g723.so ; G.723 Codect Translator - Requires N/A
;load => codec_g726.so ; ITU G.726-32kbps G726 Transcoder - Requires N/A
;load => codec_g729.so ; G729/PCM16 (signed linear) Codec Translator - Requires N/A
;load => codec_gsm.so ; GSM/PCM16 (signed linear) Codec Translat - Requires N/A
;load => codec_ilbc.so ; iLBC/PCM16 (signed linear) Codec Translat - Requires N/A
;load => codec_lpc10.so ; LPC10 2.4kbps (signed linear) Voice Codec Translat - Requires N/A
;load => codec_speex.so ; Speex/PCM16 (signed linear) Codec Translat - Requires N/A
;load => codec_ulaw.so ; Mu-law Coder/Decoder - Requires N/A
```

```
; Formats —
```

```
;load => format_au.so ; Sun Microsystems AU format (signed linear) - Requires N/A
;load => format_g723.so ; Raw G.723 data - Requires N/A
;load => format_g726.so ; Raw G.726 (16/24/32/40kbps) data - Requires N/A
;load => format_g729.so ; Raw G729 data - Requires N/A
;load => format_gsm.so ; Raw GSM data - Requires N/A
;load => format_h263.so ; Raw h263 data - Requires N/A
```

```

;load => format_ilbc.so ; Raw iLBC data - Requires N/A
;load => format_jpeg.so ; JPEG (Joint Picture Experts Group) Image - Requires N/A
;load => format_mp3.so ; MP3 - Requires N/A
;load => format_pcm_alaw.so ; Raw aLaw 8khz PCM Audio support - Requires N/A
load => format_pcm.so ; Raw uLaw 8khz Audio support (PCM) - Requires N/A
;load => format_sln.so ; Raw Signed Linear Audio support (SLN) - Requires N/A
;load => format_vox.so ; Dialogic VOX (ADPCM) File Format - Requires N/A
;load => format_wav_gsm.so ; Microsoft WAV format (Proprietary GSM) - Requires N/A
load => format_wav.so ; Microsoft WAV format (8000hz Signed Linear) - Requires N/A

; Applications —
;load => app_addon_sql_mysql.so ; Do several mySQLy things - Requires ?
;load => app_adsiprogram.so ; Asterisk ADSI Programming Application - Requires res_adi.so
;load => app_alarmreceiver.so ; Alarm Receiver for Asterisk - Requires N/A
;load => app_authenticate.so ; Authentication Application - Requires N/A
;load => app_cdr.so ; Tell Asterisk to not maintain a CDR for the current call - Requires N/A
;load => app_chanisavail.so ; Check if channel is available - Requires N/A
;load => app_chanspy.so ; Listen to the audio of an active channel - Requires N/A
;load => app_controlplayback.so ; Play a file with fast forward and rewind - Requires N/A
;load => app_curl.so ; ? - Requires N/A
;load => app_cut.so ; The application Cut is deprecated. - Requires N/A
;load => app_db.so ; Database access functions for Asterisk - Requires N/A
load => app_dial.so ; Dialing Application - Requires res_features.so, res_musiconhold.so
;load => app_dictate.so ; Virtual Dictation Machine - Requires N/A
;load => app_directed_pickup.so ; Directed Call Pickup Application - Requires ?
;load => app_directory.so ; Provide directory of voicemail extensions - Requires N/A
;load => app_disa.so ; Allows someone from outside an "internal" system dialtone - Requires N/A
;load => app_dumpchan.so ; Dump Info About The Calling Channel - Requires N/A
load => app_echo.so ; Echo audio read from channel back to the channel - Requires N/A
;load => app_enumlookup.so ; EnumLookup is deprecated. Use ENUMLOOKUP() function - Requires N/A
;load => app_eval.so ; Reevaluates strings - Requires N/A
;load => app_exec.so ; Allows an arbitrary application to be invoked even when not hardcoded into the
dialplan. - Requires N/A
;load => app_externalivr.so ; External IVR Interface Application - Requires ?
;load => app_festival.so ; Simple Festival Interface - Requires N/A
;load => app_flash.so ; Flashes a Zap Trunk - Requires ?
;load => app_forkcdr.so ; Fork The CDR into 2 separate entities. - Requires N/A
;load => app_getcpeid.so ; Obtains and displays ADSI CPE ID and other info in order to properly setup
zapata.conf for on-hook operations.
;load => app_groupcount.so ; Deprecated, please use the function GroupCount - Requires N/A
;load => app_hasnewvoicemail.so ; Indicator whether a voice mailbox has messages in a given folder. -
Requires N/A
;load => app_ices.so ; Encode and Stream via icecast and ices - Requires N/A
;load => app_image.so ; Sends an image on a channel. - Requires N/A
;load => app_intercom.so ; Obsolete - does not load
;load => app_lookupblacklist.so ; Look up Caller*ID name/number from blacklist database - Requires N/A
;load => app_lookupcidname.so ; Look up CallerID Name from local database - Requires N/A
load => app_macro.so ; Macro Handling Application - Requires N/A
;load => app_math.so ; Basic Math Functions - Requires N/A
;load => app_md5.so ; MD5 checksum applications - Requires N/A

```

```

;load => app_meetme.so ; MeetMe conference bridge - Requires ?
;load => app_milliwatt.so ; Generate a Constant 1000Hz tone at 0dbm (mu-law) - Requires N/A
load => app_mixmonitor.so ; Records the audio on the current channel to the specified file. - Requires ?
;load => app_mp3.so ; Play an MP3 file or stream - Requires N/A
;load => app_nbscat.so ; Play an NBS local stream - Requires N/A
;load => app_page.so ; Places outbound calls and dumps them into a conference bridge, muted - Requires ?
;load => app_parkandannounce.so ; Call Parking and Announce Application - Requires res_features.so
load => app_playback.so ; Sound File Playback Application - Requires N/A
;load => app_privacy.so ; Require phone number to be entered, if no CallerID sent - Requires N/A
;load => app_queue.so ; Queue handling applications - Requires res_features.so, res_monitor.so,
res_musiconhold.so
;load => app_random.so ; Conditionally branches, based upon a probability - Requires N/A
;load => app_read.so ; Reads a #-terminated string of digits - Requires N/A
;load => app_readfile.so ; Stores output of file into a variable - Requires N/A
;load => app_realtime.so ; Use RealTime config handler to read data into channel variables. - Requires N/A
;load => app_record.so ; Record to a file - Requires N/A
;load => app_rxfax.so ; Receive a FAX to a file - Requires ?
;load => app_saycountpl.so ; Polish counting grammar - Requires ?
;load => app_sayunixtime.so ; Says a specified time in a custom format - Requires N/A
;load => app_senddtmf.so ; Sends arbitrary DTMF digits - Requires N/A
;load => app_sendtext.so ; Sends text to current channel (callee). - Requires N/A
load => app_setcallerid.so ; Set Caller*ID on a call to a new value. - Requires N/A
;load => app_setcdruserfield.so ; Append to the CDR user field - Requires N/A
;load => app_setcidname.so ; SetCIDName deprecated in favor of the function CALLERID(name) - Requires
N/A
;load => app_setcidnum.so ; SetCIDNum deprecated in favor of the function CALLERID(number) - Requires
N/A
;load => app_setrdnis.so ; SetRDNIS deprecated in favor of the function CALLERID(rdnis) - Requires N/A
;load => app_settransfercapability.so ; Set ISDN Transfer Capability - Requires N/A
;load => app_sms.so ; SMS/PSTN handler - Requires N/A
;load => app_softhangup.so ; Hangs up the requested channel - Requires N/A
;load => app_stack.so ; Stack routines - Requires ?
;load => app_striplsd.so ; Deprecated - Requires N/A
;load => app_substring.so ; Deprecated - Requires N/A
;load => app_system.so ; Execute a system command - Requires N/A
;load => app_talkdetect.so ; Playback with Talk Detection - Requires N/A
;load => app_test.so ; Interface Test Application - Requires N/A
;load => app_transfer.so ; Transfer caller to remote extension - Requires N/A
;load => app_txfax.so ; Trivial FAX Transmit Application - Requires ?
;load => app_txtcidname.so ; The TXTCIDName deprecated in favor of the TXTCIDNAME dialplan function -
Requires N/A
;load => app_url.so ; Send URL Applications - Requires N/A
;load => app_userevent.so ; Send an arbitrary event to the manager interface - Requires N/A
;load => app_verbose.so ; Send arbitrary text to verbose output - Requires N/A
;load => app_voicemail.so ; Comedian Mail (Voicemail System) - Requires res_adi.so
;load => app_waitforring.so ; Waits until first ring after specified time - Requires N/A
;load => app_waitforsilence.so ; Waits for silence of specified time - Requires N/A
;load => app_while.so ; While Loops and Conditional Execution - Requires N/A
;load => app_zapateller.so ; Block Telemarketers with Special Information Tone - Requires N/A
;load => app_zapbargue.so ; Barges in on a specified zap channel - Requires ?

```

;load => app_zapras.so ; Executes a RAS server using pppd on the given channel - Requires ?

;load => app_zapscan.so ; Scan Zap channels to monitor calls - Requires ?

[global]

chan_modem.so=yes

??????? <http://192.168.1.1:8088/gui/static/config/index.html>????? admin??? HereisYourPassword?

??????

??????

System Status

System Status
Uptime : 23:40:29 up 6:59, load average: 0.03, 0.03, 0.00

Trunks

Status	Trunk	Type	Username	Port/Hostname/IP
Unregistered	iptel	sip	osslabsupport	iptel.org

Conference Rooms
No Conferences setup

Parked Calls
No Parked Calls

Extensions

Free Busy UnAvailable Ringing

Extension	Name/Label	Status	Type
6000	alang	Messages : 0/0	SIP/IAX User
6001	alang-2	Messages : 0/0	SIP/IAX User
-- *No Extension assigned		Check Voicemails	VoiceMailMain
-- *No Extension assigned		Dial by Names	Directory

Dial Plans

DialPlans

Manage DialPlans

A Dial Plan is a collection of Outgoing Call Rules . Dial Plans are assigned to Users to specify the dialing permissions they have. For example, you might have one Dial Plan for local calling that only permits users of that Dial Plan to dial local numbers, via the "local" outgoing calling rule. Another user may be permitted to dial long distance numbers, and so would have a Dial Plan that includes both the "local" and "longdistance" outgoing calling rules.

Default	Dial Plan	Calling Rules	Options
<input type="checkbox"/>	Internal	iptel, ext_6000, ext_6001, default, parkedcalls, conferences, ringgroups, voicemenus, queues, voicemailgroups, directory, pagegroups, page_an_extension	Edit Delete

Asterisk Log Messages

The screenshot shows the Asterisk Log Messages page. The left sidebar contains a navigation menu with items like System Status, Configure Hardware, miSDN Config, Trunks, Outgoing Calling Rules, Dial Plans, Users, Ring Groups, Music On Hold, Call Queues, Voice Menus, Time Intervals, Incoming Calling Rules, Voicemail, Paging/Intercom, Conferencing, Follow Me, Directory, Call Features, VoiceMail Groups, Voice Menu Prompts, System Info, and Backup. The main content area is titled "Asterisk Log messages" and shows a date filter for "23 Oct 2009". The log entries include:

```

[Oct 23 11:31:01] NOTICE[12415] app_playback.c: Reloading say.conf
[Oct 23 11:31:04] NOTICE[12415] pbx_ael.c: Starting AEL load process.
[Oct 23 11:31:04] NOTICE[12415] pbx_ael.c: AEL load process: calculated config file name '/opt/etc/asterisk/extensions.
[Oct 23 11:31:04] NOTICE[12415] pbx_ael.c: AEL load process: parsed config file name '/opt/etc/asterisk/extensions.ael'
[Oct 23 11:31:04] NOTICE[12415] pbx_ael.c: AEL load process: checked config file name '/opt/etc/asterisk/extensions.ael'
[Oct 23 11:31:04] NOTICE[12415] pbx_ael.c: AEL load process: compiled config file name '/opt/etc/asterisk/extensions.ael'
[Oct 23 11:31:04] NOTICE[12415] pbx_ael.c: AEL load process: merged config file name '/opt/etc/asterisk/extensions.ael'
[Oct 23 11:31:04] NOTICE[12415] pbx_ael.c: AEL load process: verified config file name '/opt/etc/asterisk/extensions.ael'
[Oct 23 11:55:16] NOTICE[13099] cdr.c: CDR simple logging enabled.
[Oct 23 11:55:16] NOTICE[13099] loader.c: 136 modules will be loaded.
[Oct 23 11:55:32] NOTICE[13099] pbx_ael.c: Starting AEL load process.
[Oct 23 11:55:32] NOTICE[13099] pbx_ael.c: AEL load process: calculated config file name '/opt/etc/asterisk/extensions.
[Oct 23 11:55:32] NOTICE[13099] pbx_ael.c: AEL load process: parsed config file name '/opt/etc/asterisk/extensions.ael'
[Oct 23 11:55:32] NOTICE[13099] pbx_ael.c: AEL load process: checked config file name '/opt/etc/asterisk/extensions.ael'
[Oct 23 11:55:32] NOTICE[13099] pbx_ael.c: AEL load process: compiled config file name '/opt/etc/asterisk/extensions.ael'
[Oct 23 11:55:32] NOTICE[13099] pbx_ael.c: AEL load process: merged config file name '/opt/etc/asterisk/extensions.ael'
[Oct 23 11:55:32] NOTICE[13099] pbx_ael.c: AEL load process: verified config file name '/opt/etc/asterisk/extensions.ael'
[Oct 23 11:56:11] ERROR[13179] config.c: ***** YOU SHOULD REALLY READ THIS ERROR *****
[Oct 23 11:56:11] ERROR[13179] config.c: Future versions of Asterisk will treat a #include of a file that does not exist
[Oct 23 11:56:11] ERROR[13179] config.c: ***** YOU SHOULD REALLY READ THIS ERROR *****
[Oct 23 11:56:11] ERROR[13179] config.c: ***** YOU SHOULD REALLY READ THIS ERROR *****
[Oct 23 11:56:16] WARNING[13194] app_system.c: Unable to execute 'atscan > /opt/etc/asterisk/atscan.conf'
[Oct 23 11:57:18] ERROR[13256] config.c: ***** YOU SHOULD REALLY READ THIS ERROR *****
[Oct 23 11:57:18] ERROR[13256] config.c: ***** YOU SHOULD REALLY READ THIS ERROR *****
[Oct 23 11:57:18] ERROR[13256] config.c: Future versions of Asterisk will treat a #include of a file that does not exist
[Oct 23 11:57:18] ERROR[13256] config.c: ***** YOU SHOULD REALLY READ THIS ERROR *****
[Oct 23 11:57:18] ERROR[13256] config.c: ***** YOU SHOULD REALLY READ THIS ERROR *****
[Oct 23 11:57:18] ERROR[13256] config.c: ***** YOU SHOULD REALLY READ THIS ERROR *****
[Oct 23 11:57:22] WARNING[13270] app_system.c: Unable to execute 'atscan > /opt/etc/asterisk/atscan.conf'
[Oct 23 12:05:01] WARNING[13624] app_system.c: Unable to execute 'atscan > /opt/etc/asterisk/atscan.conf'
[Oct 23 12:05:02] ERROR[13634] config.c: ***** YOU SHOULD REALLY READ THIS ERROR *****
[Oct 23 12:05:02] ERROR[13634] config.c: ***** YOU SHOULD REALLY READ THIS ERROR *****
[Oct 23 12:05:02] ERROR[13634] config.c: Future versions of Asterisk will treat a #include of a file that does not exist
    
```

Outgoing Calling Rules

The screenshot shows the "Manage Calling Rules" page. It includes a "New Calling Rule" button and a "Restore Default Calling Rules" button. A descriptive text explains that an outgoing calling rule pairs an extension pattern with a trunk used to dial the pattern. Below this is a table of existing rules:

Calling Rule	Pattern	Trunk	Fallover Trunk	
iptel	_013	iptel	None Selected	Edit Delete
ext_6000	6000	Local Destination : Goto User 6000		Edit Delete
ext_6001	6001	Local Destination : Goto User 6001		Edit Delete

System Information

System Information

General | Network | Disk Usage | Memory Usage

OS Version:
Linux DD-WRT 2.4.37 #7598 Sat Oct 10 04:38:28 CEST 2009 mips GNU/Linux

Uptime:
21:10:06 up 8 min, 0 users,
Load Average: 0.18, 0.39, 0.27

Asterisk Build:
Asterisk/1.4.22.1
Asterisk GUI-version : 2.0

Server Date & TimeZone: Fri Oct 23 21:10:06 UTC 2009

Hostname:
DD-WRT

Trunks

Manage SIP & IAX trunks

Analog Trunks | Service Providers | **VOIP Trunks** | T1/E1/BR0 Trunks

+ New SIP/IAX Trunk

Provider Name	Type	Hostname/IP	Username	
iptel	SIP	iptel.org	osslabsupport	Edit Delete

Trunks are outbound lines used to allow the system to make calls to the real world. Trunks can be VoIP lines or traditional telephony lines.

Users

?????./trim.sh <config.conf>

????

- Asterisk GUI
 - www.asterisk.org
 - www.asteriskguru.com
 - wiki.binkey.nl
 - [AstRecipes » Installing the Asterisk GUI](#)
 - astbook.asteriskdocs.org
 - [asterisk-gui Archives](#)
- Asterisk Management Script
 - www.fivn.com