Applications between Welltech and Asterisk

This document is describing the configuring manner of registering and communicating with Asterisk only. Please visit the official WEB of Asterisk http://www.asterisk.org directly for searching the related information if there is any question about the detail operating of Asterisk.

Version: 1.2

Introduction

This is the documentation for the applications between Welltech units and Asterisk IP PBX. It will show you some basic configurations in our units to work with Asterisk for some extra functions and also something you have to be note while you need these kinds of functions. We will introduce the application as following:

- **Before the configurations**

  This chapter will show you the software version of Asterisk and firmware of the testing units. Please read this chapter before starting the configurations.

- **Registration with Asterisk**

  This chapter will show you how to configure the units and Asterisk to make the registrations successfully. This is the basic configurations before all the functions. It also shows you the endpoints register on Asterisk behind the NAT.

- **Making the basic calls**

  All the calls making have to work with the dial-plan in Asterisk, this chapter will show you how to configure it.

- **Extra Functions of Asterisk**

  This chapter will describe some extra functions, including:
  Auto attendant
  Music on hold
  Voice Mail
  Transfer
  Forward

- **Q&A**

  How to get it work with one stage dialing
  click-click noise
  Caller ID display
  FXO port locked
  DTMF error

- **Configuration Example**

  Here is the simple configuration and setup procedure of FXS, FXO and Asterisk.

Before the configurations

Please get the detail info about the firmware version of our units and Asterisk:

Model Name	35xx	37xx	38xx	Asterisk
Firmware version	111	105	106	1.2.10
CPU				P4-2.5G
RAM				512M
OS				Red Hat 9.0

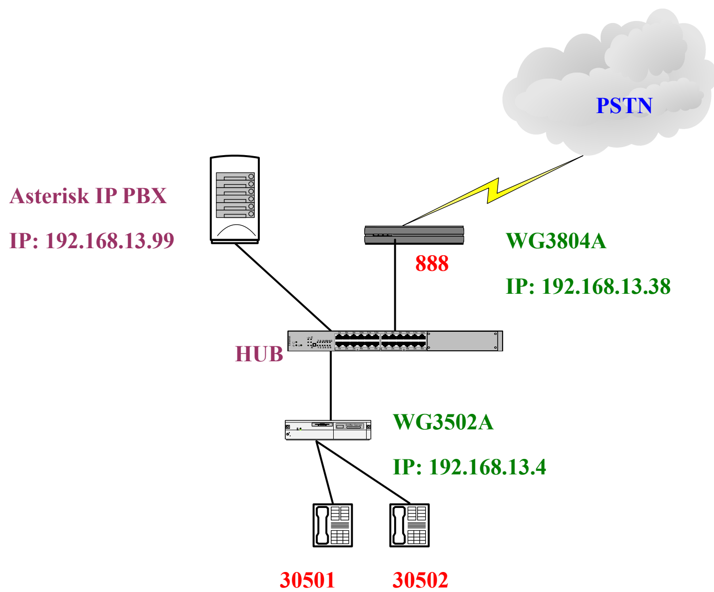
Before you start, please check out the hardware specification of Asterisk from the Asterisk's web site: http://www.asterisk.org

About the firmware version of WellGate units, please check out our technical support web site: http://www.welltech.com.tw/support/support_e.htm

Registration with Asterisk

There is a file named sip.conf could manage all the sip setting for the client or the server. User could define this file for the registration of the client, including the phone number, account, password…etc. Please find out this file with the location /etc/asterisk and get the more detail info from the technical support web site of Asterisk: www.asterisk.org

There is nothing special for the configurations between Welltech CPE and Asterisk. For the Welltech CPE, the settings of registration for Asterisk are as same as the other proxy server.



Registration with Authentication

The following is the example for the configurations in Asterisk with the registration from WellGate:

In sip.conf

```
[30501]
type=friend
username=30501
secret=30501
host=dynamic
canreinvite=no
dtmfmode=rfc2833
context=from-internal
```

- [30501]

The extension name is defined within square brackets ([ ]). In this case, we have defined the extensions 30501 as the extension number.

- type

According to the explanations from the documents of sip.conf file, there are three types for the client registrations. The **friend** is a device that can both receive and send calls through the Asterisk; **peer** is a device which could receive calls from the Asterisk; **user** is a device that could make calls through Asterisk. This makes sense for most desk handsets and other devices. If you are not sure which type to choose, we suggest you set the type to the **friend**.

- username

This option sets the username that the Asterisk attempts to connect when a call is received. Used when for some reasons the value is not the same as the username the client registered.

- secret

This is the password for the registration. It takes an alphanumeric string.

- host

This configures the host to which this peer is to connect, and user can define an IP address, FQDN or dynamic in here. In this case we set it to "dynamic" so the call sent from the caller to the callee [30501] will be routed to the registered IP address.

- canreinvite

This potion is used to tell the server to never issue a re-invite to the client. If the server sends the re-invite to both of the caller and callee, the voice RTP could be sent to both peers directly, won't go through the Asterisk IP PBX.

- dtmfmode

Set default dtmfmode for sending DTMF. Default is rfc2833, you can also set it to "inband" or "info" type.

- context

A context is assigned to a channel definition to direct incoming calls into the matching context in extensions.conf file. Any channel connecting to an Asterisk machine has to have a context defined into which it will arrive. In this case, somebody call 30501, the asterisk will then check the context "from-internal" of extensions.conf to make sure where and how to route this call to the peer.

Based on the above configuration of Asterisk, below is the configuration in Welltech CPE.

In FXS gateway

```
usr/config$ sip –px 192.168.13.99 (the IP address of Asterisk)
usr/config$ sip –line1 30501 (set the line number)
usr/config$ security –line 1 –name 30501 –pwd 30501 (set the ID and password)
```

WellGate couldn't support the configurations in sip.conf while users define the different line number and account as so far. So please set the item of square brackets ([ ]) and the username as the same.

Please commit the data and reboot the CPE after finish the configurations as above.

Registration without Authentication

The configuration is just like the registration with authentication, except the username and secret value. Please check out the example as following:

In sip.conf

```
[30501]
type=friend
;username=30501
;secret=30501
host=dynamic
canreinvite=no
dtmfmode=rfc2833
context=from-internal
```

Users could only assign a symbol just like ";" to disable the configurations you want. The example as above could disable the authorization require for the registration procedure.

According to the SIP standard, the first registered packet from CPE won't contain ID and password, and then the proxy server will reply the 401 or 407 messages to the CPE for authentication. The Welltech CPE will re-send the registered packet within ID and password for authentication.

Because of that the authentication isn't a requirement of the Asterisk, the account and password could be set as any values or "x" for null. Please check out the example as following:

In FXS gateway

```
usr/config$ sip –px 192.168.11.200 (The IP address of Asterisk)
usr/config$ sip –line1 30501 (set the line number)
usr/config$ security –line 1 –name x –pwd 30501 x (set the ID and password to x)
```

After all of the configuration files, such as the sip.conf file had been changed, users have to reload the configurations via the command line interface of Asterisk by "**reload**" command, otherwise the new setting won't be useful.

Registration behind NAT

Users could define the NAT flag for the configurations with the client to resolve the application while Asterisk is outside the NAT and clients inside the NAT connecting with Asterisk. Please check out the example as the following:

In sip.conf

```
[30501]
type=friend
username=30501
secret=30501
host=dynamic
canreinvite=no
dtmfmode=rfc2833
context=from-internal
nat=yes
```

For the NAT flag, it could be set with four types with this flag. Please get more detail info from the Asterisk wi-ki web site as following:

http://www.voip-info.org/tiki-index.php?page=Asterisk+sip+nat

Please pay more attentions on this function. All the NAT function was controlled by the Asterisk not WellGate. All the RTP packets for the voice will be passed through the Asterisk, so please check out the available codec for this application in the configurations of Asterisk.

According to the experiment in our side, we were using the codec G711 for this kind of application only. Please check out this from the Asterisk web site before you need this function.

Making the basic calls

For the configurations in the Registration chapter, the registration should be OK but the client still can not make call until you prepare the dial plan. Due to Asterisk is just like an IP PBX, users have to configure the dial plan in the extensions.conf file which could be found in the location /etc/asterisk. All of the dial plans were controlled by this file, so please configure this file before you start to test the basic call.

The extensions.conf file lays out the dial-plan, bringing channels together with applications and service. This file features extension matching logic and intelligent call routing logic.

This file could support the some kind of protocol for the dialing, such as the Zap, IAX, and SIP…etc. In the setting rule, it could also support more parameters for the call making, such like the answering the calls automatically, hanging up automatically, playing IVR, and dialing the number…etc. It could also define some kind of parameters for this call, such as the color ring back tone, timer for timeout…etc. So please get more detail info from the Asterisk we site if you need more function with this extensions.file.

Making calls to the registered clients

This has to be worked with the sip.conf while you try to configure the extensions.conf file for the dialing plan.

The following example was according to the sip configurations in the first chapter:

In sip.conf

```
[30501] type=friend
username=30501
secret=30501
host=dynamic
canreinvite=no
dtmfmode=rfc2833
context=from-internal
nat=yes
```

The extensions.conf was based on the context named in the sip.conf file. Please check out the example in extensions.conf as following:

In extensions.conf

```
[from-internal]
exten => 30501,1,Dial(SIP/30501,20,Ttm)
```

While the caller make a call to number 30501, the call will be routed to Asterisk first. The Asterisk will find out the matching rule in the extensions.conf and decide the destination here.

The Dial is an application defined in Asterisk.

When Asterisk got the called number 30501, the call signal will be sent to the destination where the CPE 30501 is registered with SIP protocol, and the value of timeout is 20 seconds.

The Ttm is the flags that provide some special feature for the CPE. For more information about the flags, please go to:
http://www.voip-info.org/tiki-index.php?page=Asterisk+standard+extensions

Making calls to the none-registered clients

Asterisk could also send the calls to the clients which were not registered on it. This could be used like set the calls to the Trunk gateway or FXO unit for the outbound calls. Please check out the example as following:

In extensions.conf

```
[from-internal]
exten => _0XXXXXXXXX,1,Dial(SIP/${EXTEN}@192.168.13.38)
```

The call will be sent to the client where was located with the IP address 192.168.13.38 with the prefix number 0 and digits length 10. This syntax was followed the extension document from the Asterisk web site. The parameter ${EXTEN} will capture the real numbers which was be sent from the client side.

For this configuration, please also checkout the dialing plan in the Trunk or FXO gateway. In WellGate FXO unit, because of the route table configuration, this call will be passed through the FXO port to the PSTN directly. This is not necessary for other special configurations in WellGate FXO units.

For more information about the routing table of Welltech FXO gateway, please refer to the user's manual of FXO or go to:
http://www.welltech.com/support/voip/Application%20Note/English/How%20to%20achieve%20one-stage%20dialing.PDF

Extra Functions

This chapter will describe some special function of the Asterisk IP PBX.

Auto attendant

Due to the Asterisk is just like an IP PBX, it could also provide the auto attendant for the incoming calls.

If a call was been sent to a certain number which was define as the auto attendant number in Asterisk, the caller should hear the greeting define in the asterisk. This function was provided by Asterisk and users have to enable this function in Asterisk before using it. This function was also controlled by the file named extensions.conf. Please check out the following configurations in Asterisk about this function:

In extensions.conf

```
[IVR]
exten => 999,1,Answer
exten => 999,2,Background(greeting)
exten => 999,3,WaitExten(3)
exten => hangup()
```

While the client sends the number 999 to Asterisk, the first emotion in Asterisk is answering this call, the second is playing the IVR which was named "greeting" and waiting for the extension number after the IVR. The Asterisk will wait for 3 seconds then hangup the call.

User can record his own greeting file by follow configuration:

In extensions.conf

```
[record]
exten => 001,1,Record(greeting:wav)
```

While the caller call 001, he should hear a "bi" tone to indicate the beginning of recording. The file will be saved as the name "greeting" with "wav" format.

In WellGate FXS model, users could dial the number "999" directly to get the auto attendant works. This is the normal dialing and users didn't have any special configurations in WellGate; In WellGate FXO model, it could pass through the incoming calls from the PSTN side to the auto attendant of Asterisk directly. It has to depend on the hotline function in WellGate FXO.

Below is the command for hotline of FXS and FXO gateway

In FXS gateway

```
usr/config$ sysconf –service 1 (set the FXS's service mode to hotline mode)
usr/config$ bureau -hotline 1 192.168.13.99 999 (hotline to the Asterisk with number 999)
```

In FXO gateway

```
usr/config$ line –config 1 hotline 999 (hotline to the Asterisk with number 999)
```

While someone dials the PSTN number into the FXO port, WellGate FXO will hotline to the number “999” to the Asterisk automatically. Then the caller could get the greeting from Asterisk while Asterisk answers this call.

Music On Hold

The music on hold is a common function of the traditional PBX, and the Asterisk IP PBX also provides such feature. If your Asterisk implements the music on hold, the user will hear music during the period of hold. Because the music is under “MP3” format so you should compile and install the MP3 player called **mpg123** first.

- Installing mpg123

1) Download mpg123
2) tar –zxvf mpg123-<version>.tar.gz
3) cd mpg123-<version>
4) make
5) make install

- Add the Music On Hold feature

Just like we said in the previous chapter, there are some flags exist within the application “Dial”

In extensions.conf

```
[from-internal]
exten => 30501,1,Dial(SIP/30501,20,Ttm)
```

The flag “m” indicates the Asterisk will provide the music to the calling party until the call is answered. And it will also provide the music if the user perform the “hold” or “transfer”.

Voice Mail

One of the most popular features of any modern PBX is voicemail. Asterisk also provides a voicemail system.

- Creating Mailboxes

You can create a Mailbox for each extension in the voicemail.conf which is located in /etc/asterisk.

In voicemail.conf

```
[default]
30502 => 1234,name
```

We will create a context named [default]. In this context we will define a new mailbox. The way of creating a mailbox is the following:

First you write the mailbox number (30502). It is followed by the => character. After this character, the first argument is the desired password (1234) for the mailbox. The password is followed by the name.

- Add the register account with MWI

MWI means “Message Waiting Indicator”. If there is a new voice mail exists in the user’s mailbox, the Asterisk will send a “SIP notify” to trigger for the MWI feature.

In sip.conf

```
[from-internal]
type=friend
username=30502
secret=30502
host=dynamic
canreinvite=no
dtmfmode=rfc2833
mailbox=30502@default
context=from-internal
```

- Adding the voicemail to the Dialplan

In extensions.conf

```
[from-internal]
exten => 30502,1,Dial(SIP/30502,20,Ttm)
exten => 30502,2,VoiceMail(30502@default)
exten => 30502,3,Hangup()
```

In this example when somebody dials 30502, the call will be routed to extension 30502 with SIP channel.

If there is no one answers this call, the Asterisk will perform the next step.

Now, if the user is available and the line is free but the called person does not answer within 20 seconds, the Asterisk will go to next step. This one contains another copy of the VoiceMail application.

As arguments in its brackets we have set the following: 30502@defaultl. The word 30502 is for the mailbox which we have set in the voicemail.conf. The last part is @default. This shows the context in voicemail.conf, in which the mailbox 30502 is configured.

- Accessing Voicemail

In extensions.conf

```
[from-internal]
exten => 500,1,VoiceMail()
```

In this variant, when somebody dials 500, the system will ask the caller to enter a mailbox number and then to enter the password for this number. The caller will hear whether it has new messages or not. The caller could leave the mailbox either by hanging up the line or by pressing the pound key (#).

Transfer function

The SIP standard has defined the standardization for transfer function, but Asterisk also provides its own procedure for transfer.

This chapter we will let you know how to get transfer works between Welltech and Asterisk.

- Transfer by Asterisk

User can define a feature key for Asterisk based transfer in the features.conf which is located in /etc/asterisk.

In features.conf

```
[featuremap]
blindxfer => #              ; Blind transfer
;disconnect => *0           ; Disconnect
;automon => *1              ; One Touch Record
atxfer => *2                ; Attended transfer
```

In the features.conf, you will find out the context called [featuremap], here defines 4 feature keys. Please unmark the feature key for “blindxfer” and “atxfer” for transfer function. The “blindxfer” is for “blind transfer” and “axfer” is for attended transfer.

The default feature key of blind transfer is “#” and the default feature key of blind transfer is “*2”, these two keys are changeable here.

If you press the “#” or “*2” during conversation, there will be some announcement appeared to indicate you the transfer state is ready, so that you can continue to call the 3rd called party. You do not need to do anything in Welltech CPE if you are perform the Asterisk based transfer. You just need to make sure the Asterisk can recognize the DTMF sent by the Welltech CPE.

- Transfer by SIP standard

Both of the Asterisk and Welltech CPE can support the SIP standard transfer. You don’t need to do anything in the Asterisk and Welltech LAN phone. But one thing you should pay attention is the Welltech FXS and FXSO gateway default is transfer disabled. User should enable manually via command line.

In FXS gateway

```
usr/config$ sysconf –transfer 1 (enable the transfer function)
```

Forward function

The SIP standard has defined the standardization for froward function, but Asterisk also provides its own procedure for transfer.

This chapter we will let you know how to get forward function works between Welltech and Asterisk.

- Forward by Asterisk

It is not easy to implement the Asterisk based forward function because you should

implement the AGI function first, so we do not discuss this feature here.
For more information about the AGI and Asterisk based forward function, please visit the website of Asterisk: www.asterisk.org.

- Forward by SIP standard

The Welltech CPE can support the SIP standard forward. It is depended on the SIP 302 message. You don't need to do anything in the Asterisk but you should enable this function in Welltech CPE.

The welltech CPE can support three kinds of forward: unconditional, busy and no answer.

In FXS gateway

```
usr/config$ support –uncon 123 (define the forward number for unconditional forward.)
usr/config$ support –busy 123 (define the forward number for busy forward.)
usr/config$ support –noanswer 123 (define the forward number for no answer forward.)
```

Q and A

How to get it work with one stage dialing

- Q:

Hi, I hope that I can dial the PSTN number from my SIP gateway with Asterisk directly. But I always hear dial tone….why??

- A:

The FXO device has a routing table function. There are two default routes exist in the routing table, one is for IP incoming call another is for FXO incoming call. So the IP call will be routed to FXO and the Call from FXO side will be routed to IP side.

If the 380x got an incoming call from IP side and the line number is not its local SIP number. Then the 380x will forward this number to the FXO which is based on the default route, and the FXO will dial this number through PSTN automatically. User just need to dial the destination PSTN number, this is called one-stage dialing. For more information about one-stage dialing, please go to:

http://www.welltech.com/support/voip/Application%20Note/English/How%20to%20achieve%20one-stage%20dialing.PDF

If the 380x got an incoming call from IP side, and the line number is the FXO's local SIP number, then the FXO will answer this call and user will hear dial tone from PSTN side, then they should re-dial the destination PSTN number. User should dial number twice, so this is called two-stage dialing.

So you should check the dial plan of the extensions.conf now, make sure the Asterisk can send the correct PSTN number to FXO gateway.

Below is an example of dial plan.

In extensions.conf

```
[from-internal]
exten => _0XXXXXXXXX,1,Dial(SIP/${EXTEN}@192.168.13.38)
```

click-click noise

- Q:

We are trying to make Welltech 3802 gateway work with Asterisk proxy server. We can register and make a call successfully. But sometimes we will hear a strange voice, like [click].....Do you know why? Thanks in advance.

- A:

The Asterisk proxy default only supports G711 voice codec, but it does not support CNG feature. This problem (click-click noise) is caused by the non-support CNG feature. The Welltech FXO devices default is CNG enabled with G711u/a codec, so you need to disable the G711u CNG by "**sysconf -silence 0**" command.

Caller ID display

- Q:

Hi, our networking department is testing some VoIP devices for providing PBX solutions for small and medium sized business. One of our choices, for FXO equipment is WellGate 3804. I would like to connect 4 external PSTN lines into an Asterisk proxy over SIP protocol. I have some questions and your help would be appreciated a lot. I'm interested in getting the CallerID from the PSTN line, signaling the call in Asterisk Proxy and pick-up the call from Asterisk SIP commands. (I would like to obtain something similar with Zapata telephony interface) When will be the next version of firmware available?

- A:

The FXO has the ability to detect the PSTN caller ID, and the also show the PSTN caller id if the FXO can send the correct PSTN caller ID to FXS. For more information about FXO caller id detection, please go to:
http://www.welltech.com/support/voip/Application%20Note/English/FXO_Caller_ID_Detection.pdf

FXO port locked

- Q:

Hello. We have a problem with FXO/SIP. If FXO is connected with PSTN and my Asterisk, I call to this FXO port. I will hear a voice greeting from my Asterisk, and I hangup the phone (do not dial any thing).
The GW will not disconnect, and I dial to this FXO port again, it is busy.
Please let me know how to solve my problem.

- A:

I think this problem is caused by the disconnect tone mismatch.

WellGate FXO products has four default settings of disconnect tone. If the disconnect tone is match with one of the four settings, the LINE port from PSTN/PBX will be released after two seconds. Otherwise, it may be released after one minute or lock this LINE permanently.

There are two ways to help you analyze the disconnect tone of your PBX/CO.
One is analyzed by FXO device itself, another is by the other software.

For more information, please go to:
http://www.welltech.com/support/voip/Application_Note/English/How_to_learn_and_record disconnect tone.PDF

DTMF error

- Q:

For some reason that we don't know, the DTMF dialled from the analogue phone reached "twice" the PSTN. For example, if I dial 2532487 the PSTN get 22553322448877, or sometimes any digit may have got successfully… but it's in a random way.

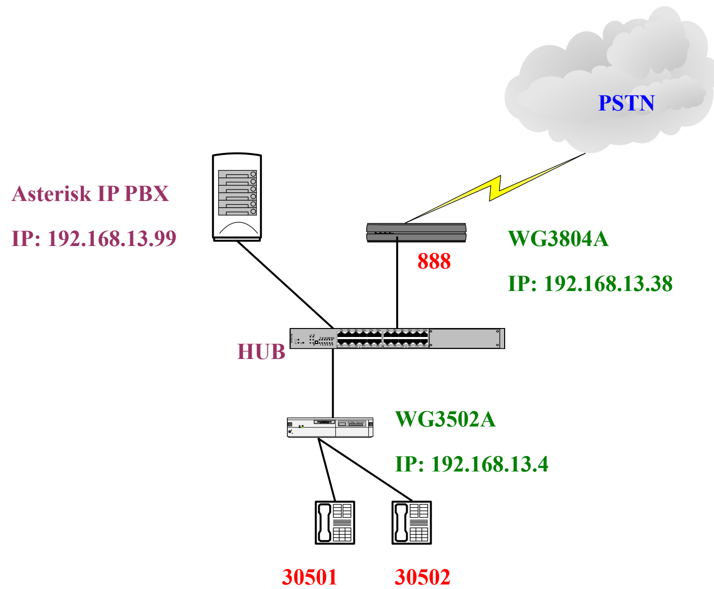
- A:

For our experience, the RFC2833 DTMF of Asterisk seems to have some problems. Asterisk's RFC2833 implementation has no concept of Variable Length DTMF. While technically, this is not in violation of the RFC, it causes issues such as clipped DTMF, duplicate DTMF and occasionally no DTMF on the receiving end. Many gateway vendors do not implement jitter buffers on RTP events, and will handle them in the order they receive, so it is possible for DTMF events to arrive out of sequence and cause all sorts of havoc. Actually, we already to find out some solution to solve this problem in Welltech side, but we think this problem should be caused by the Asterisk side.

For more information about the DTMF error with Asterisk, please refer to:
http://bugs.digium.com/view.php?id=6667

Configure example:

Here is the simple configuration and setup procedure based on the above configuration.



Test Case

- The two extensions of 3502A can register to Asterisk with number 30501 and 30502.
- 30501 and 30502 can call to each other, and they should be able to hear the color ring back tone.
- The trunk (3804A) can also register to Asterisk (registered number 888).
- Both 30501 and 30502 can call out to PSTN with prefix length 0 and digits length 10.
- The call from PSTN side to 3804A will hear an auto attendant and the caller can re-dial the extension number to 30501 or 30502.
- 30501 or 30502 can record the greeting (auto attendant)
- The 30501 and 30502 are connected with Caller ID display phone, and they should display the PSTN caller ID if they get a PSTN incoming call.
- When 30501 got a call, it can transfer to 30502.
- The call will be transfer to the MailBox if the extension 30502 does not answer.
- 30502 can access his mailbox.
- The call will be forward to 30502 if the 30501 is no answer.

Configuration of Asterisk

In sip.conf

```
[30501]
type=friend
username=30501
secret=30501
host=dynamic
canreinvite=no
dtmfmode=rfc2833
context=from-internal

[30502]
type=friend
username=30502
secret=30502
host=dynamic
canreinvite=no
dtmfmode=rfc2833
mailbox=30502@default
context=from-internal

[888]
type=friend
username=888
secret=888
host=dynamic
canreinvite=no
dtmfmode=rfc2833
context=IVR
```

In extensions.conf

```
[record]
exten => 001,1,Record(greeting:wav)

[IVR]
exten => 999,1,Answer
exten => 999,2,Background(greeting)
exten => 999,3,WaitExten
exten => hangup()
include => from-internal
```

```
[from-internal]
exten => 30501,1,Dial(SIP/30501,20,Ttm)
exten => 30502,1,Dial(SIP/30502,20,Ttm)
exten => 30502,2,VoiceMail(30502@default)
exten => 30502,3,Hangup()
exten => _0XXXXXXXXX,1,Dial(SIP/${EXTEN}@192.168.13.38)
exten => 500,1,VoiceMail()
include => record
```

In features.conf

```
[featuremap]
blindxfer => #          ; Blind transfer
;disconnect => *0       ; Disconnect
;automon => *1                 ; One Touch Record
atxfer => *2            ; Attended transfer
```

In voicemail.conf

```
[default]
30502 => 1234,name
```

Configuration of FXO

```
usr/config$ ifaddr –ip 192.168.13.38 –mask 255.255.255.0 –gate 192.168.13.254
usr/config$ sip –px 192.168.13.99 (set IP address for Asterisk)
usr/config$ sip –line1 888 –line2 888 –line3 888 –line4 888 (set line number)
usr/config$ security –line 1 –name 888 –pwd 888
usr/config$ security –line 2 –name 888 –pwd 888
usr/config$ security –line 3 –name 888 –pwd 888
usr/config$ security –line 4 –name 888 –pwd 888 (set ID and Password)
usr/config$ line –config 1 hotline 999
usr/config$ line –config 2 hotline 999
usr/config$ line –config 3 hotline 999
usr/config$ line –config 4 hotline 999 (hotline to auto attendant)
usr/config$ sysconf –callerid 3 ( set the caller ID detection to DTMF type)
usr/config$ sysconf –rba 2 (set ring before answer to 2)
usr/config$ sysconf –keypad 1 (set the keypad type to RFC2833)
usr/config$ sysconf –silence 0 (disable CNG function)
usr/config$ pt –rfc2833 101 –fax 96 (set the RFC2833 and FAX payload type)
```

Configuration of FXS

```
usr/config$ ifaddr –ip 192.168.13.4 –mask 255.255.255.0 –gate 192.168.13.254
usr/config$ sip –px 192.168.13.99 (set IP address for asterisk)
usr/config$ sip –line1 30501 –line2 30502 (set line number)
usr/config$ security –line 1 –name 30501 –pwd 30501
usr/config$ security –line 2 –name 30502 –pwd 30502 (set ID and Password)
usr/config$ sysconf –callerid 2 (enable the DTMF caller ID function)
usr/config$ sysconf –keypad 1 (set the keypad type to RFC2833)
usr/config$ sysconf –transfer 1 (enable transfer function)
usr/config$ sysconf –2833type 101 –faxtype 96 (set the RFC2833 and FAX payload)
usr/config$ support –noanswer 1 30502 (set no answer forward for line 1)
```