PC Best Networks SIP PBX Reference
Setup and Development Guide
(For V3.80)

support@pcbest.net

Index:

1 Introduction

PC Best Networks provides Windows VoIP development kits to business customers. **PC Best IP-PBX** is a proprietary, Windows-based PBX system developed as a response to the growing needs of businesses who want to deploy voice-over-the-internet through a simple, easy to manage platform. There is no difference in the use of **PC Best IP-PBX** whether you are a one-person business or a company with tens or hundreds of staff. Powerful, flexible, light and user-friendly, **PC Best IP-PBX** can be set up and run within 30 minutes on any of your working computer, with great features like, Auto Attendant, ACD(Automatic Call Distribution), MOH(Message On Hold), Ring Group, Call Parking, Pickup Group, Conference, Auto-Dialer, Database Reports, and Plug-in.

Traditional analogue PBX (private branch exchange) solutions have always been out of reach of most small and medium size businesses. Within the last 5 years, the arrival of VoIP phone systems as well as open-source solutions, such as Asterisk, which run on Linux, have become increasingly popular. Today, powerful IP-PBX system can be deployed at a much lower cost than what available 3 or 5 years ago.

Unlike Linux-based programs which may intimidate those who do not have the required expertise or resource to manage, **PC Best IP-PBX** is a user-friendly, Windows-based system and is based on SIP standard that can be set up with little effort by anyone who can configure simple mail programs like Outlook.

PC Best IP-PBX system lets even the smallest businesses quickly employ its rich features and revolutionize day-to-day business's communications. Here are the fundamental business objectives from which **PC Best IP-PBX** was built:

Increase Productivity

By removing the needs for an operator to accept incoming calls, you and/or your front office staff would be able to continue with other workloads. **PC Best IP-PBX's** digital receptionist and extension management features can be set up to answer and transfer the call as how you want.

Save time

PC Best IP-PBX's auto attendant and MOH (Message On Hold) features allow you to provide information about your business that may be relevant to callers' reason for calling you while they are on hold, thus save your time and save your customer's time. Reduce a considerable amount on time spent on the phone with these great features.

Save Cost

PC Best IP-PBX has been built to simply provide just what you want in a PBX system. We keep the development cost low and pass these savings on to you in the form of low initial investment, rather than building a complex system at higher cost with features that you may not need.

Enhance business image

Gone are the days when PBX systems were only suitable for big companies. No matter how small your company may be, your business deserves an image which big companies expose them. By using **PC Best IP-PBX** system, you give your customers a feeling that they are dealing with a well-established organization, thus enhance their confidence.

Improved Customer Services

You and/or your staff will never miss a call, no matter where you are in the world. Whether you're interstate or overseas, **PC Best IP-PBX** can be set up to connect the call to you on fixed line or mobile phone at a cost that is 5 to 10 times lower than call diversion provided by regular telephone networks. Imagine how frustrating your customer might be for not being able to get hold of you. You may be using telephone answering service but other than taking messages for you, these services are limited in what they can do for your business and your customers.

PC Best IP-PBX FEATURES

- Call Logging
- Call Reporting
- Blind Call Transfer
- Attended Call Transfer
- Call Forward on Busy
- Call Forward on No Answer
- Call Routing (DID)
- Conference Calling
- ACD (Hunt Group)
- Auto Attendant / Digital Receptionist
- Voice Mail
- Music On Hold
- Call Parking
- Call Pick Up
- Call Queue
- Call Recording
- Support Plug-in (Customized IVR Menu)

Unified Communications and Mobility

Receive Voice Mail via Email
Public SIP ID for Extensions
Advanced forwarding rules

Supported Codec (Voice Compression)

G711 (a law and u law)
G726-32
GSM
Speex
iLBC
G729

System configuration and call management can be changed instantly and inexpensively via software, not by plugging in circuit cards and pulling cables.

REQUIREMENT:

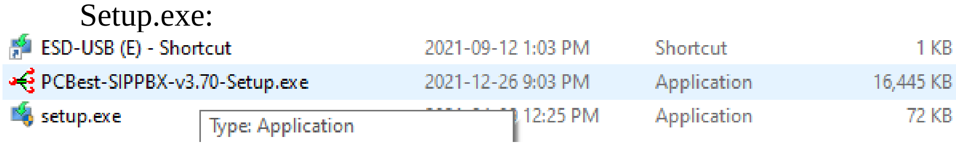
- Broadband connection
- VoIP service account
- FXO Adapter (optional)
- Minimum Pentium III with 512MB RAM, Windows XP or Vista

Our contact information for support:
Email: support@pcbest.net
Toll Free(USA & Canada): 1-888-733-6620
Local and International: 1-613-800-2202

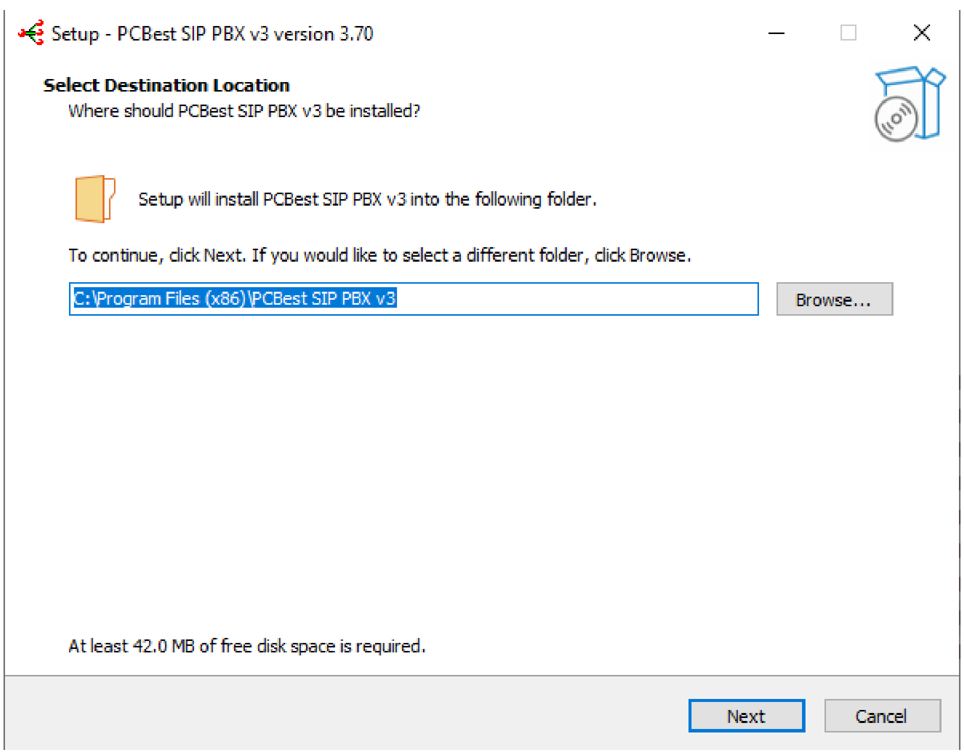
2 Installing PBX

Please install x86 version of **Visual C++ Redistributable for Visual Studio 2015** first on your machine, in order to run the PBX.

1. **Download** PC Best PBX v3 from this page: http://www.pcbest.net/sip_pbx.php
2. **Unzip** the zip file into a folder. You will see one file PCBest-SIPPBX-v3.70-Setup.exe:

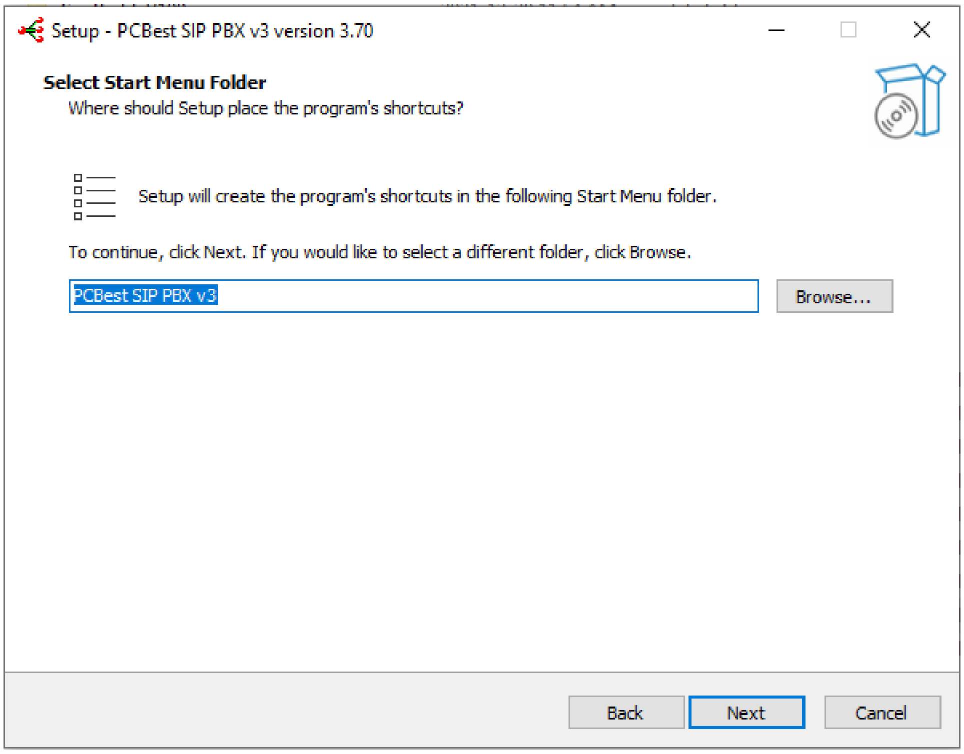


3. Run it.

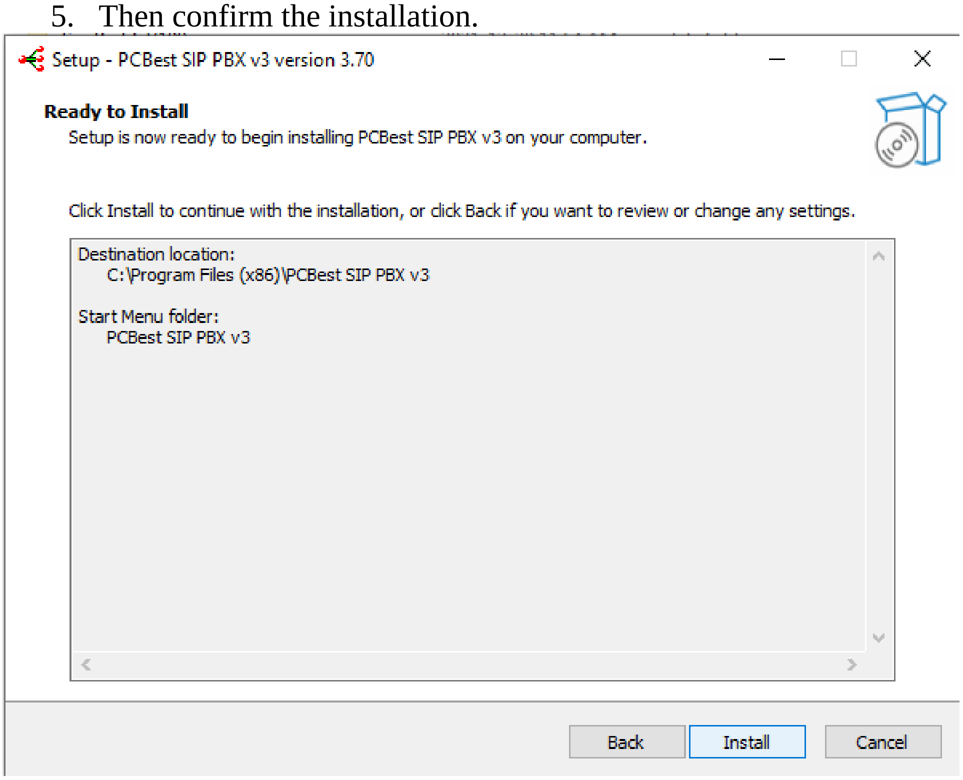


Choose where you want to install the program, and who can access it.

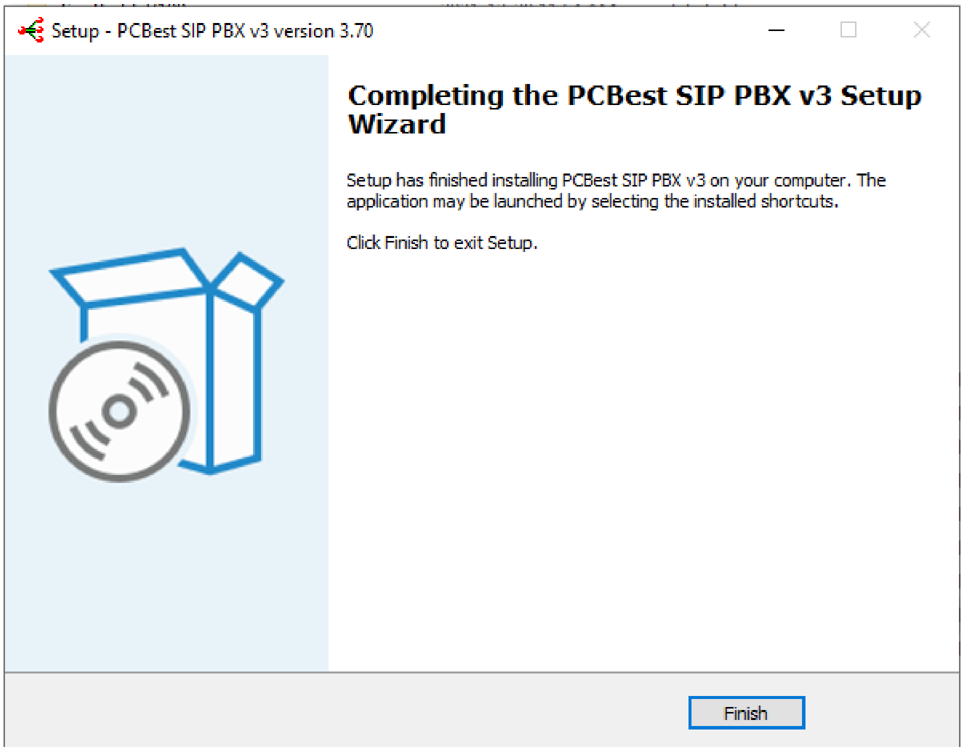
4. Click next.



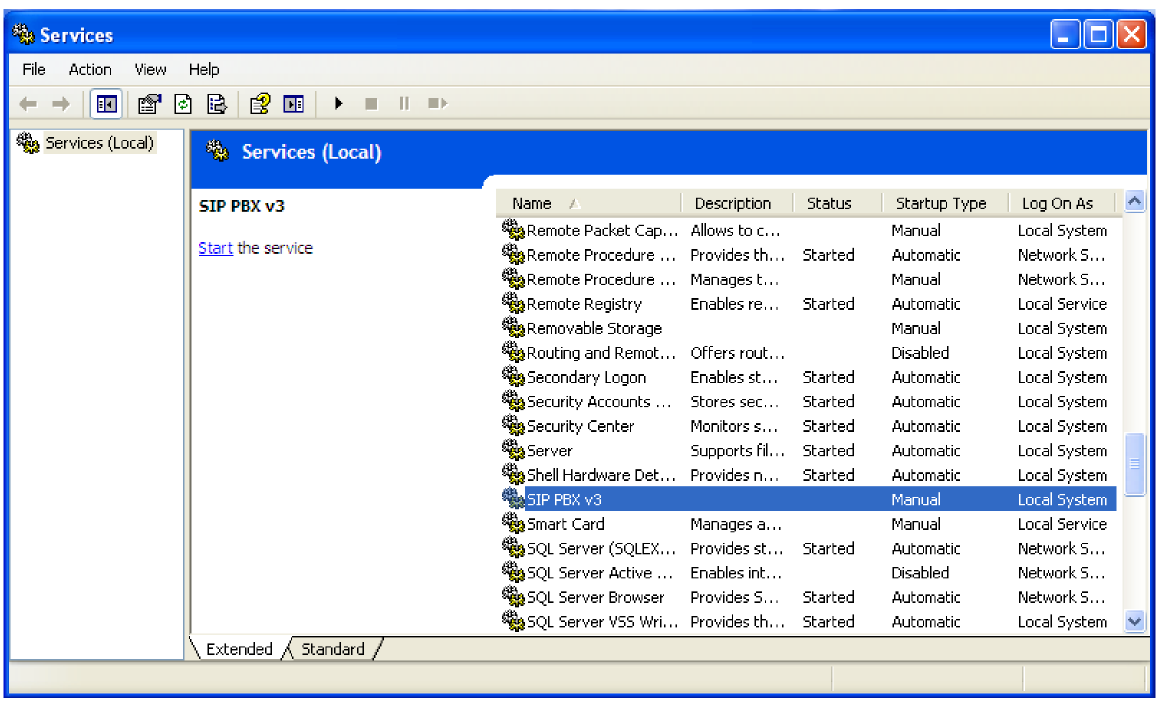
5. Then confirm the installation.



It is done.



Open Windows services:



The SIP PBX v3 service should be in the Windows service list.

6. Setup **Database**.

   Since version 3.7, PCBest SIP PBX can run with no MS SQL Server. It will setup a local embedded database to run with. For some customers they have thin clients, it reduced the load and work more efficiently. Skip this part if you don't want to set up MS SQL Server, and goto 8 directly.

Microsoft SQL Server 2005 Express Edition Service Pack 4:
http://www.microsoft.com/en-ca/download/details.aspx?id=184

Please download SQLEXPR_TOOLKIT.EXE(224.6MB) or MBSQLEXPR_ADV.EXE(254.6 MB).

Microsoft® SQL Server® 2008 Express with Tools:
http://www.microsoft.com/en-ca/download/details.aspx?id=22973

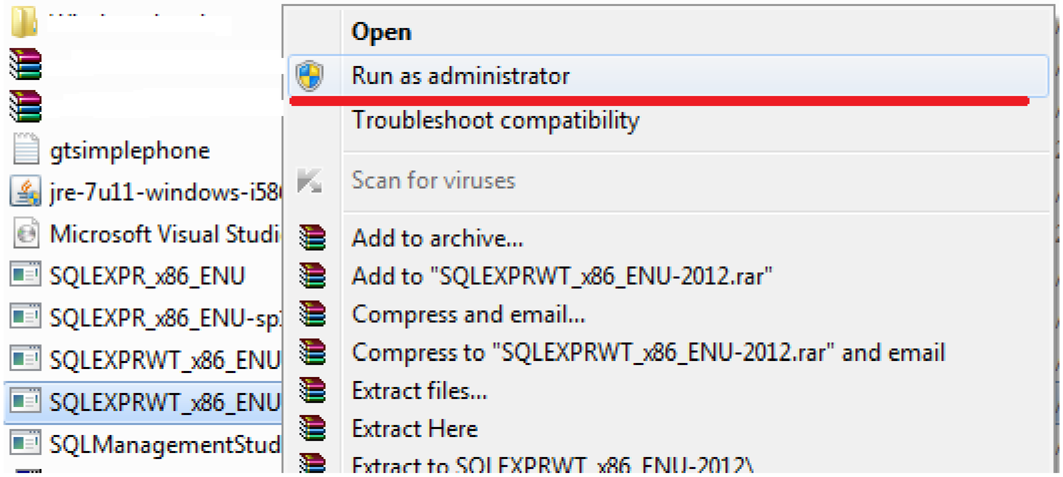
Microsoft SQL Server 2008 R2 RTM - Express with Management Tools:
http://www.microsoft.com/en-ca/download/details.aspx?id=23650

Microsoft® SQL Server® 2012 Express:
http://www.microsoft.com/en-ca/download/details.aspx?id=29062

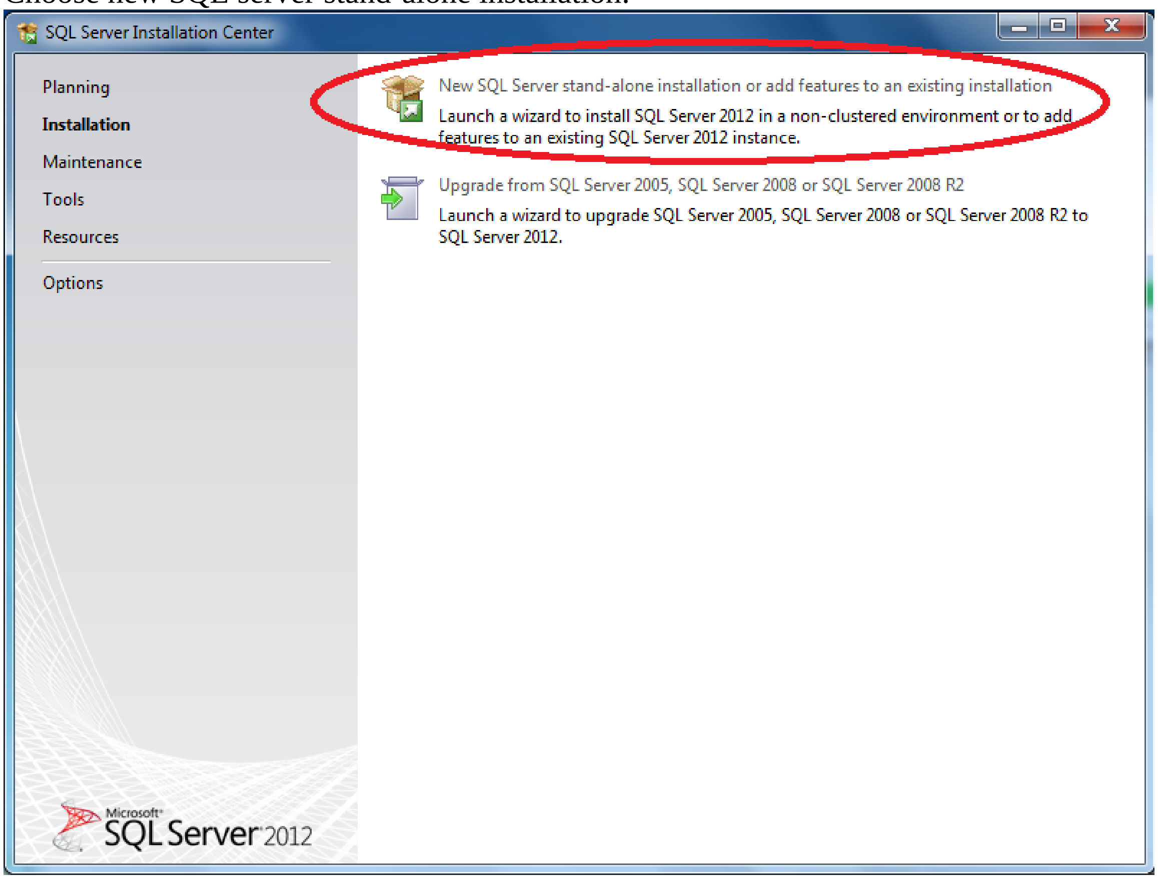
32bit OS download one of the following:
ENU\x86\SQLEXPRADV_x86_ENU.exe 1.3 GB Download
ENU\x86\SQLEXPRWT_x86_ENU.exe 706.1 MB Download

64bit OS download one of the following:
ENU\x64\SQLEXPRADV_x64_ENU.exe 1.3 GB Download
ENU\x64\SQLEXPRWT_x64_ENU.exe 669.9 MB

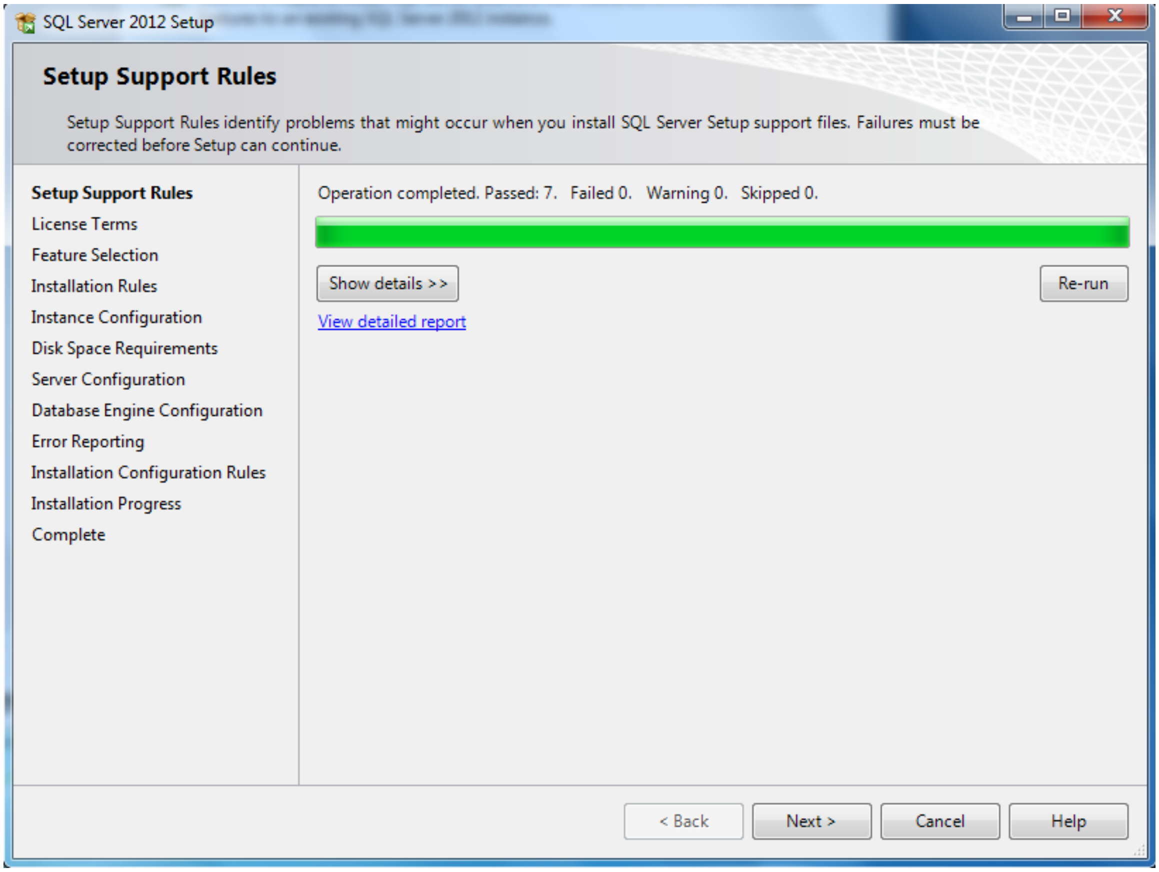
Assume we use SQL Server 2012 Express here. It is free to download from website. We download SQL Server 2012 with tool, which has management studio.
Right click on SQLEXPRWT_x86_ENU.exe for 32bit Windows or SQLEXPRWT_x64_ENU.exe for 64bit Windows, and "Run as administrator":



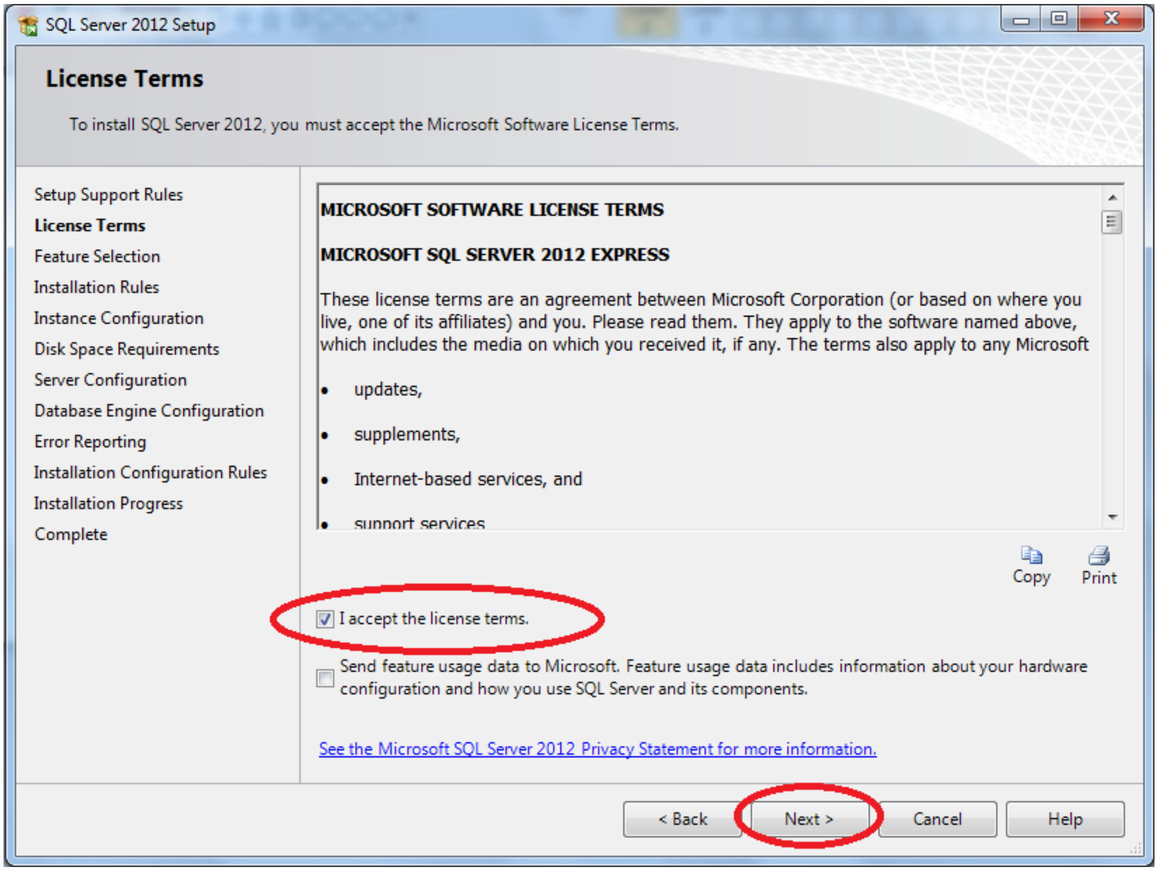
Choose new SQL server stand-alone installation:

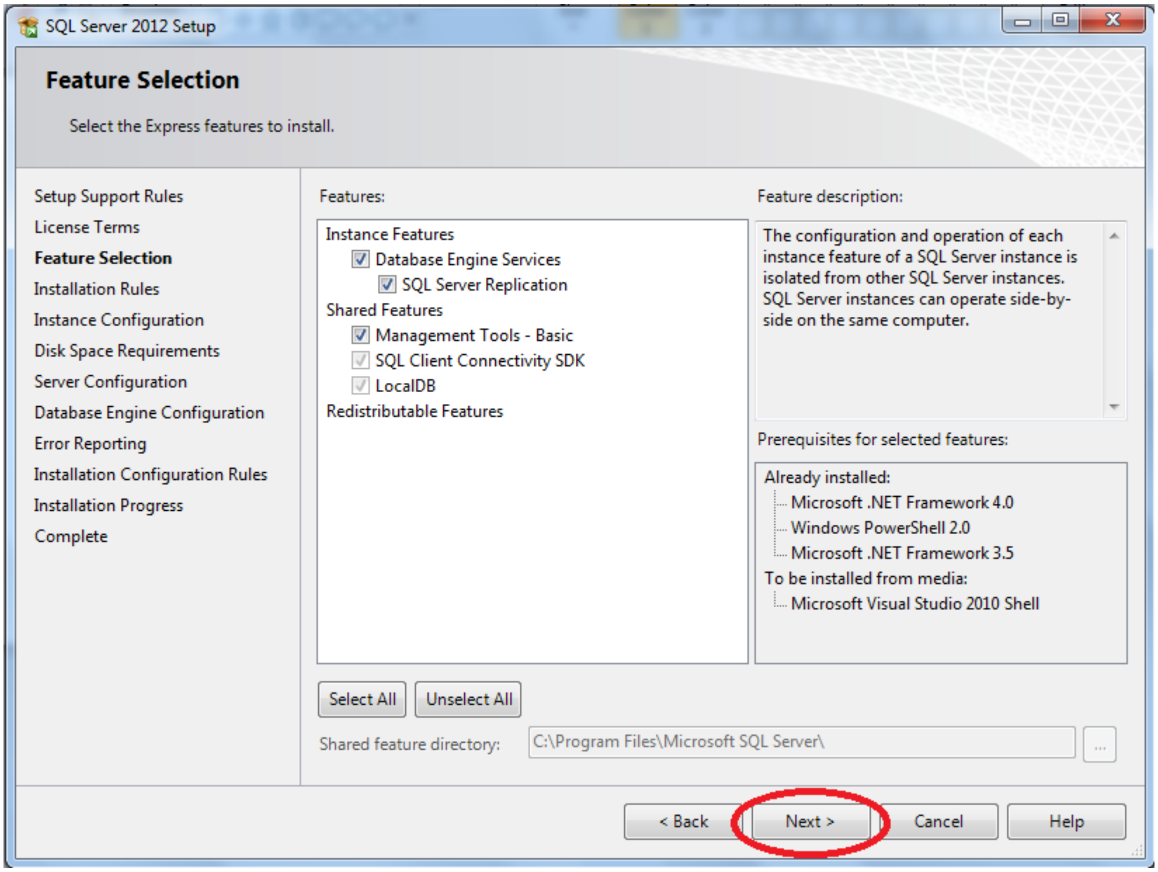


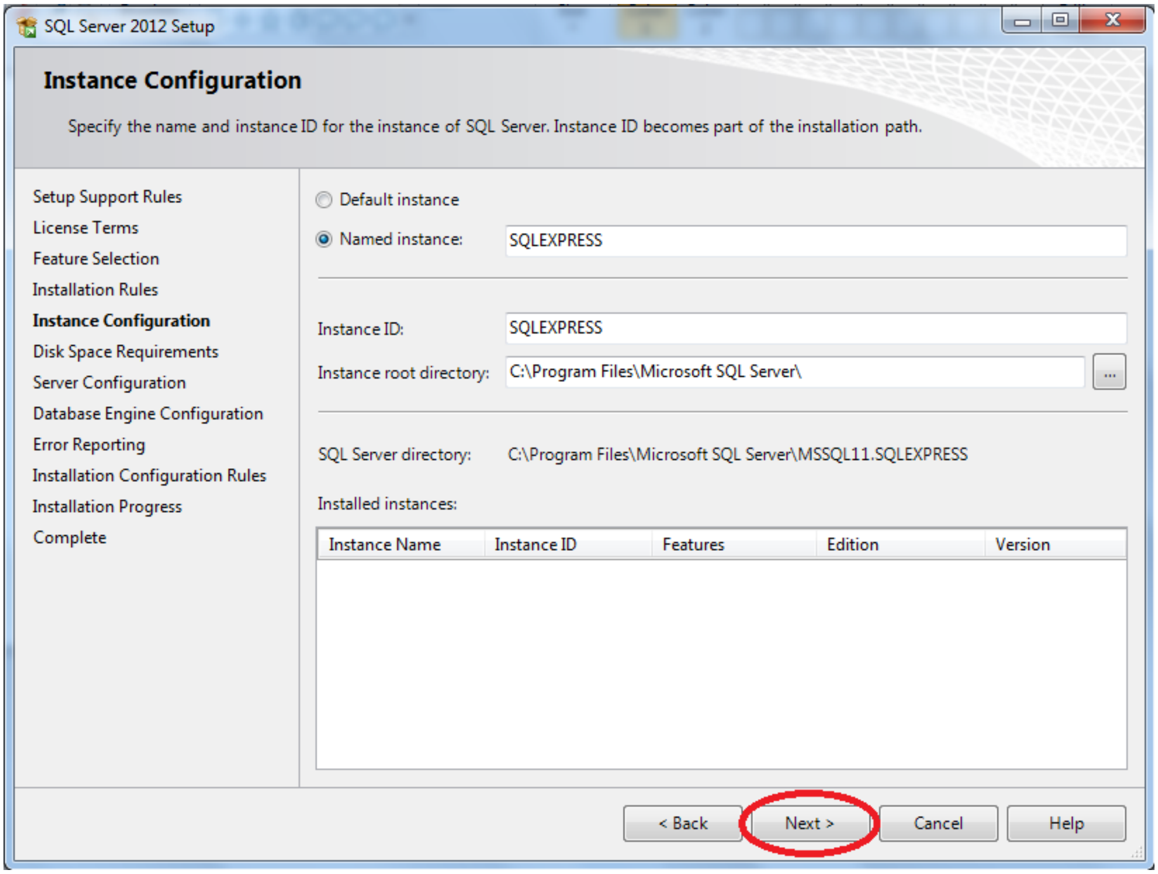
Of course, if you already have 2005, 2008, or 2008R2, you can upgrade it to 2012.
Click next:

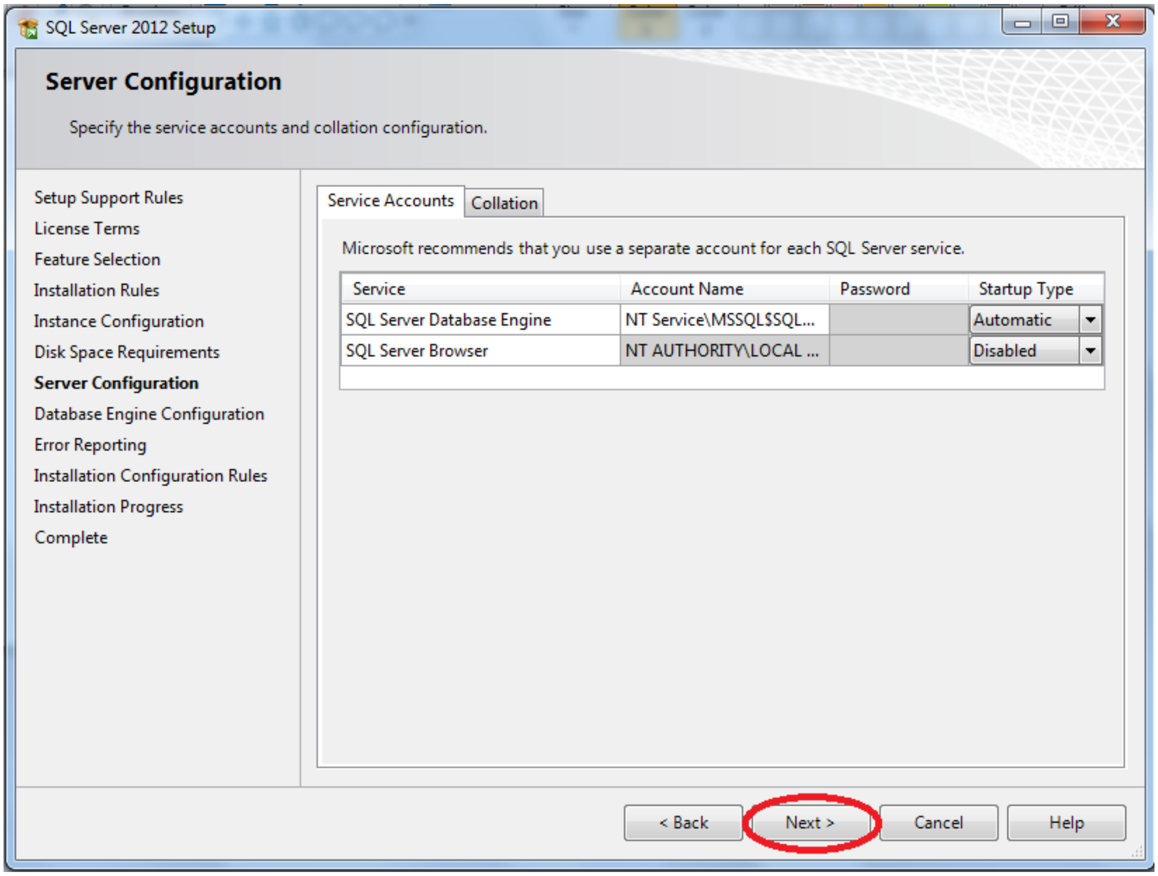


Accept license terms, and clieck Next:

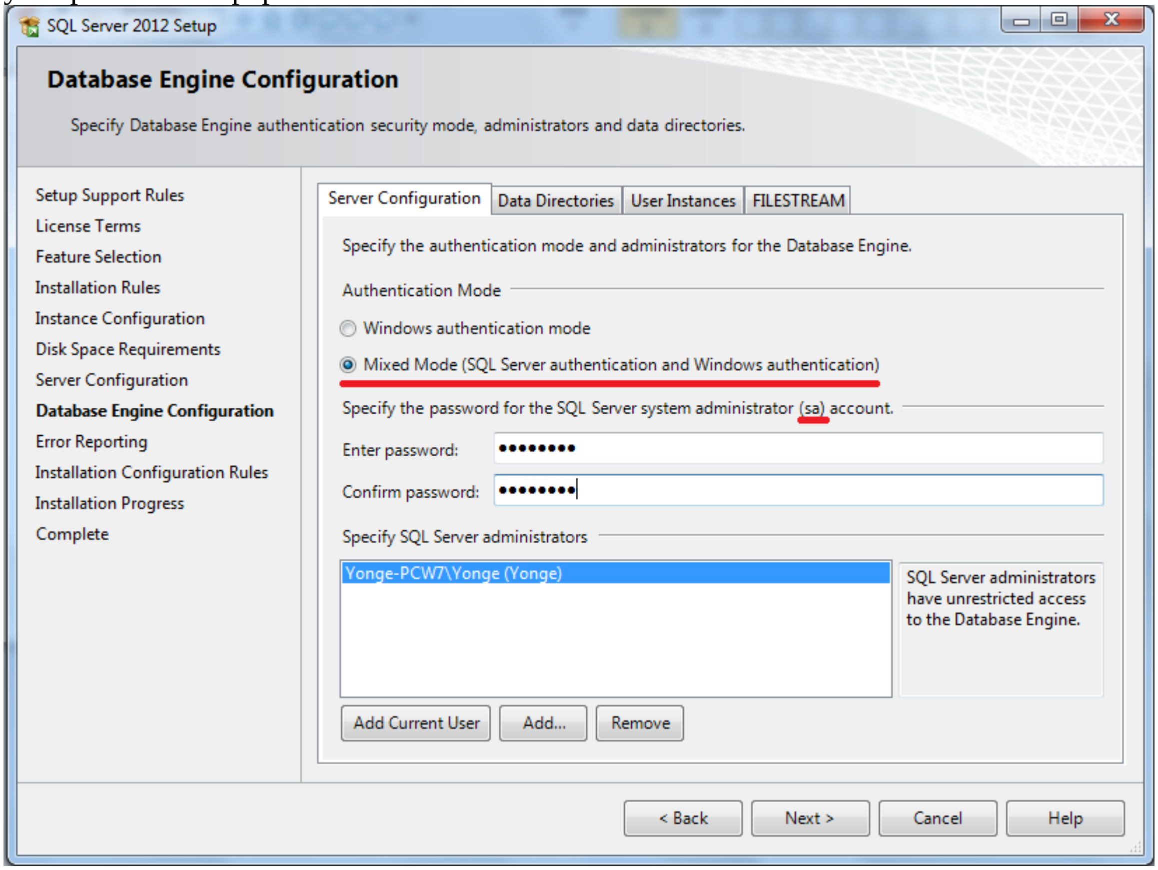

SQL Server 2012 Setup
License Terms
To install SQL Server 2012, you must accept the Microsoft Software License Terms.
Setup Support Rules
License Terms
Feature Selection
Installation Rules
Instance Configuration
Disk Space Requirements
Server Configuration
Database Engine Configuration
Error Reporting
Installation Configuration Rules
Installation Progress
Complete
MICROSOFT SOFTWARE LICENSE TERMS
MICROSOFT SQL SERVER 2012 EXPRESS
These license terms are an agreement between Microsoft Corporation (or based on where you live, one of its affiliates) and you. Please read them. They apply to the software named above, which includes the media on which you received it, if any. The terms also apply to any Microsoft
updates,
supplements,
Internet-based services, and
support services
Copy
Print
I accept the license terms.
Send feature usage data to Microsoft. Feature usage data includes information about your hardware configuration and how you use SQL Server and its components.
See the Microsoft SQL Server 2012 Privacy Statement for more information.
< Back
Next >
Cancel
Help

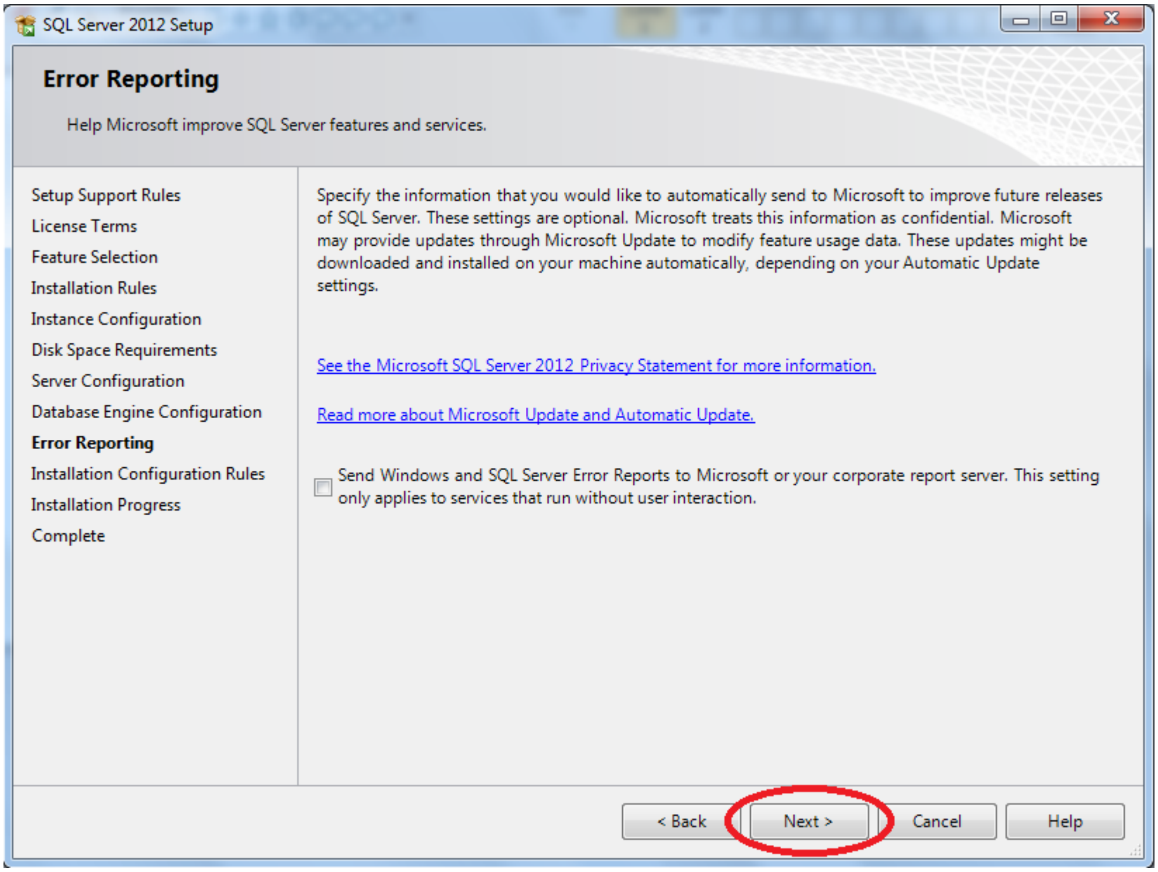

SQL Server 2012 Setup
Feature Selection
Select the Express features to install.
Setup Support Rules
License Terms
Feature Selection
Installation Rules
Instance Configuration
Disk Space Requirements
Server Configuration
Database Engine Configuration
Error Reporting
Installation Configuration Rules
Installation Progress
Complete
Features:
Instance Features
Database Engine Services
SQL Server Replication
Shared Features
Management Tools - Basic
SQL Client Connectivity SDK
LocalDB
Redistributable Features
Feature description:
The configuration and operation of each instance feature of a SQL Server instance is isolated from other SQL Server instances. SQL Server instances can operate side-by-side on the same computer.
Prerequisites for selected features:
Already installed:
Microsoft .NET Framework 4.0
Windows PowerShell 2.0
Microsoft .NET Framework 3.5
To be installed from media:
Microsoft Visual Studio 2010 Shell
Select All
Unselect All
Shared feature directory:
C:\Program Files\Microsoft SQL Server\
< Back
Next >
Cancel
Help

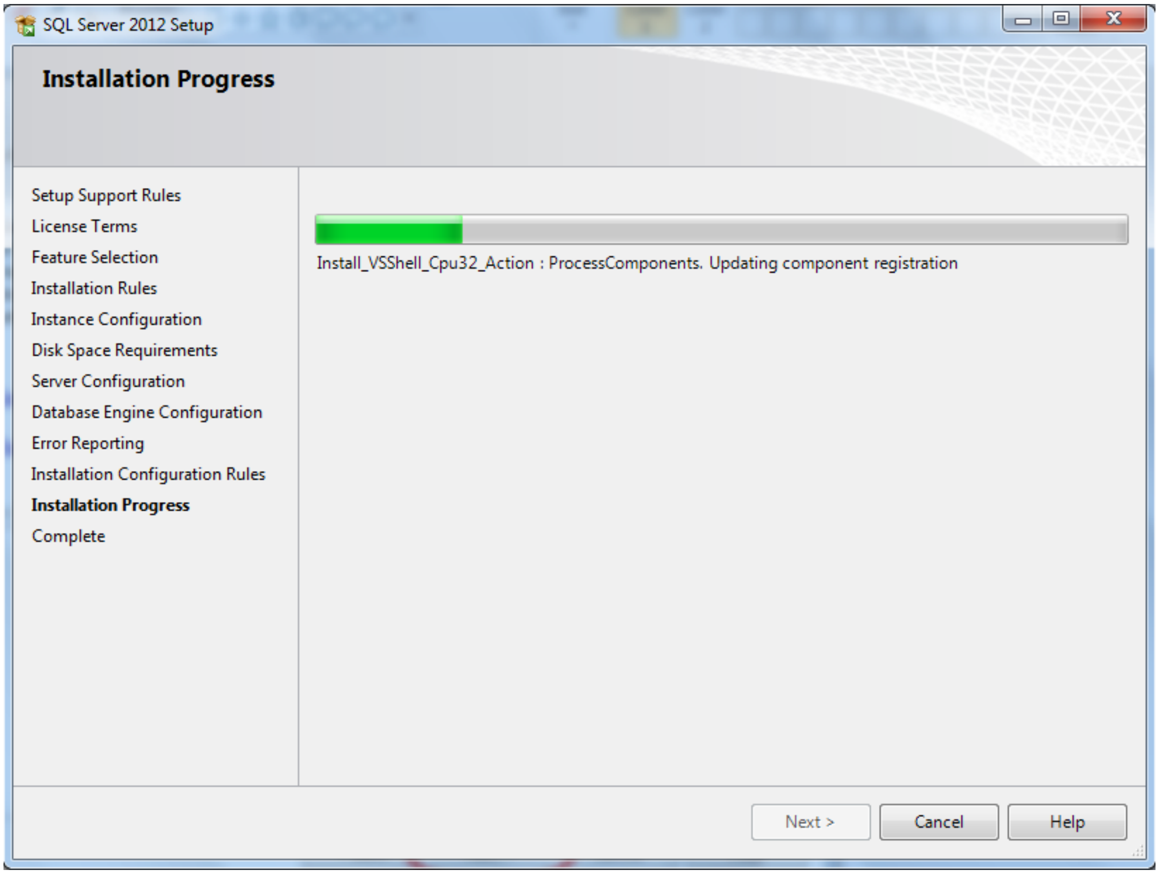

SQL Server 2012 Setup
Instance Configuration
Specify the name and instance ID for the instance of SQL Server. Instance ID becomes part of the installation path.
Setup Support Rules
License Terms
Feature Selection
Installation Rules
Instance Configuration
Disk Space Requirements
Server Configuration
Database Engine Configuration
Error Reporting
Installation Configuration Rules
Installation Progress
Complete
Default instance
Named instance:
SQLEXPRESS
Instance ID:
SQLEXPRESS
Instance root directory:
C:\Program Files\Microsoft SQL Server\
SQL Server directory:
C:\Program Files\Microsoft SQL Server\MSSQL11.SQLEXPRESS
Installed instances:
Instance Name
Instance ID
Features
Edition
Version
< Back
Next >
Cancel
Help


SQL Server 2012 Setup
Server Configuration
Specify the service accounts and collation configuration.
Setup Support Rules
License Terms
Feature Selection
Installation Rules
Instance Configuration
Disk Space Requirements
Server Configuration
Database Engine Configuration
Error Reporting
Installation Configuration Rules
Installation Progress
Complete
Service Accounts
Collation
Microsoft recommends that you use a separate account for each SQL Server service.
Service
Account Name
Password
Startup Type
SQL Server Database Engine
NT Service\MSSQL$SQL...
Automatic
SQL Server Browser
NT AUTHORITY\LOCAL ...
Disabled
< Back
Next >
Cancel
Help

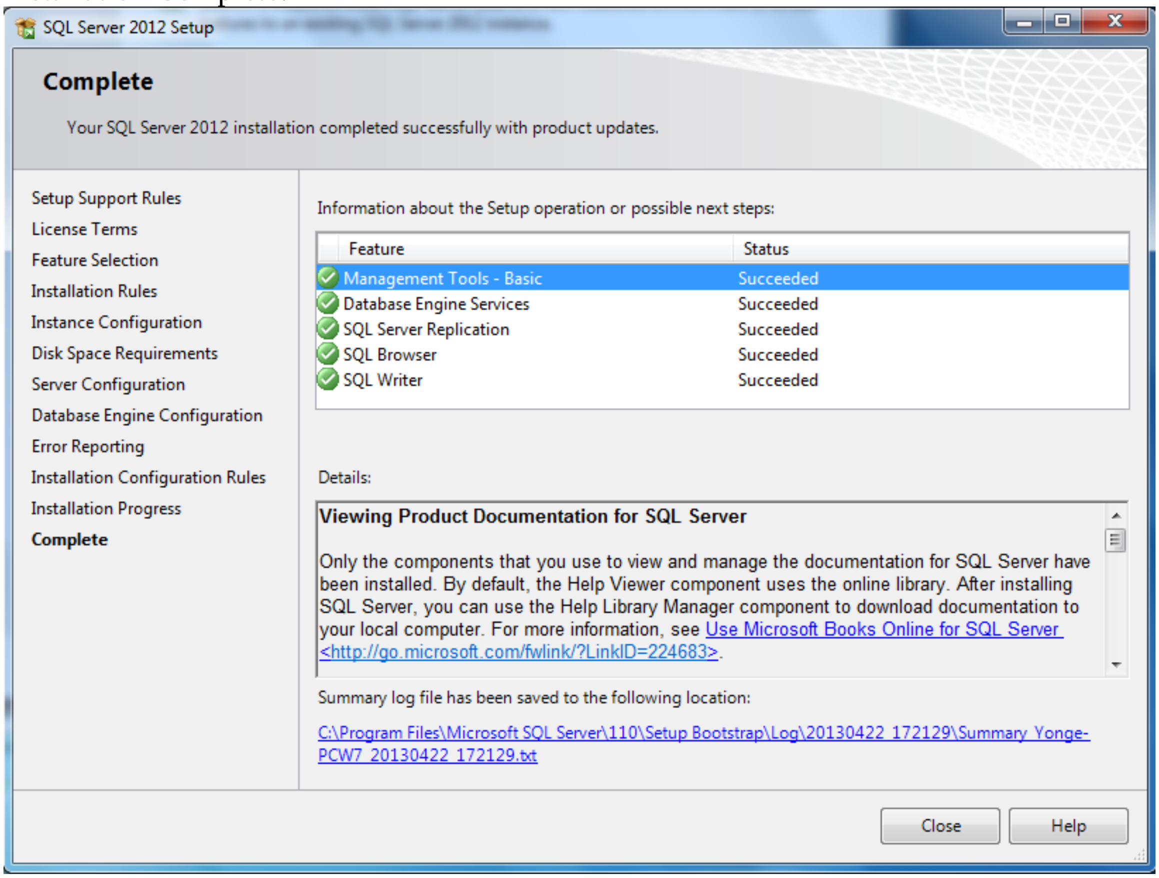
Choose Mixed Mode, and set password for account sa. NOTE: please write down your password in paper for later use.



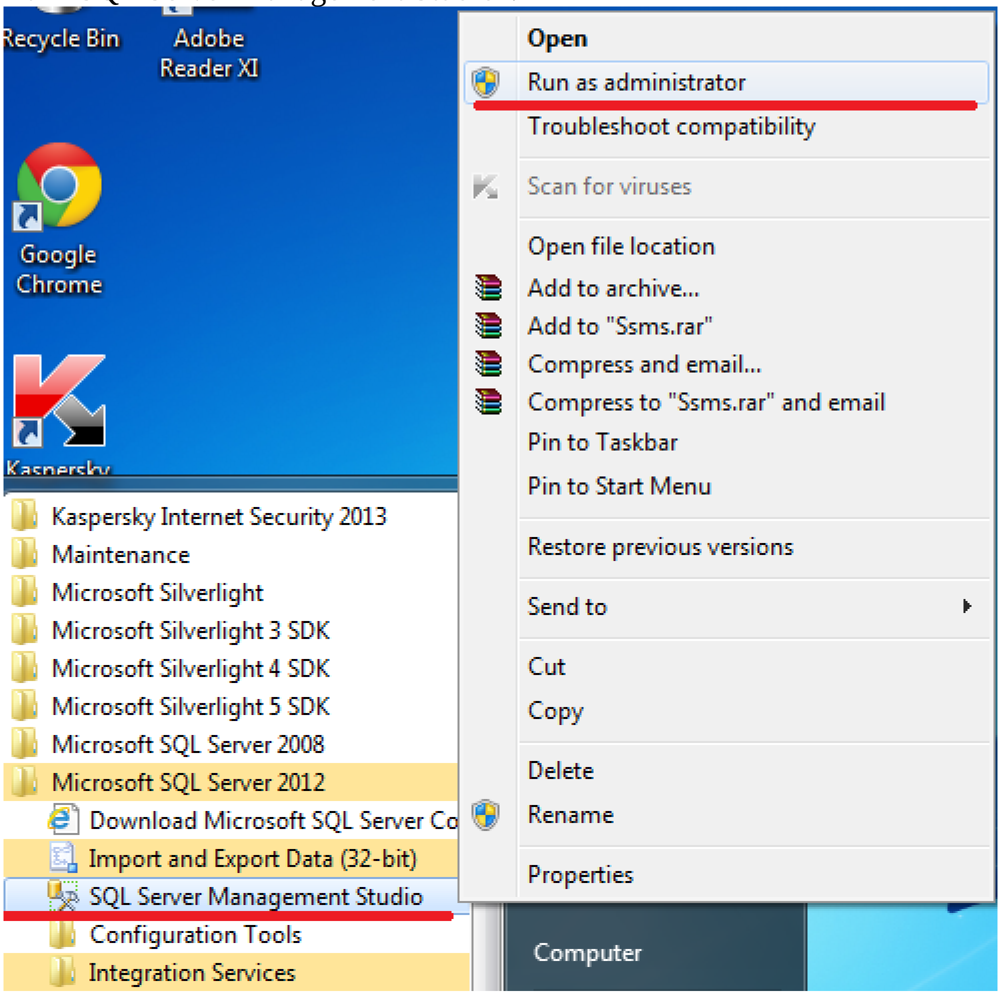

SQL Server 2012 Setup
Error Reporting
Help Microsoft improve SQL Server features and services.
Setup Support Rules
License Terms
Feature Selection
Installation Rules
Instance Configuration
Disk Space Requirements
Server Configuration
Database Engine Configuration
Error Reporting
Installation Configuration Rules
Installation Progress
Complete
Specify the information that you would like to automatically send to Microsoft to improve future releases of SQL Server. These settings are optional. Microsoft treats this information as confidential. Microsoft may provide updates through Microsoft Update to modify feature usage data. These updates might be downloaded and installed on your machine automatically, depending on your Automatic Update settings.
See the Microsoft SQL Server 2012 Privacy Statement for more information.
Read more about Microsoft Update and Automatic Update.
Send Windows and SQL Server Error Reports to Microsoft or your corporate report server. This setting only applies to services that run without user interaction.
< Back
Next >
Cancel
Help


SQL Server 2012 Setup
Installation Progress
Setup Support Rules
License Terms
Feature Selection
Installation Rules
Instance Configuration
Disk Space Requirements
Server Configuration
Database Engine Configuration
Error Reporting
Installation Configuration Rules
Installation Progress
Complete
Install_VSShell_Cpu32_Action : ProcessComponents. Updating component registration
Next >
Cancel
Help

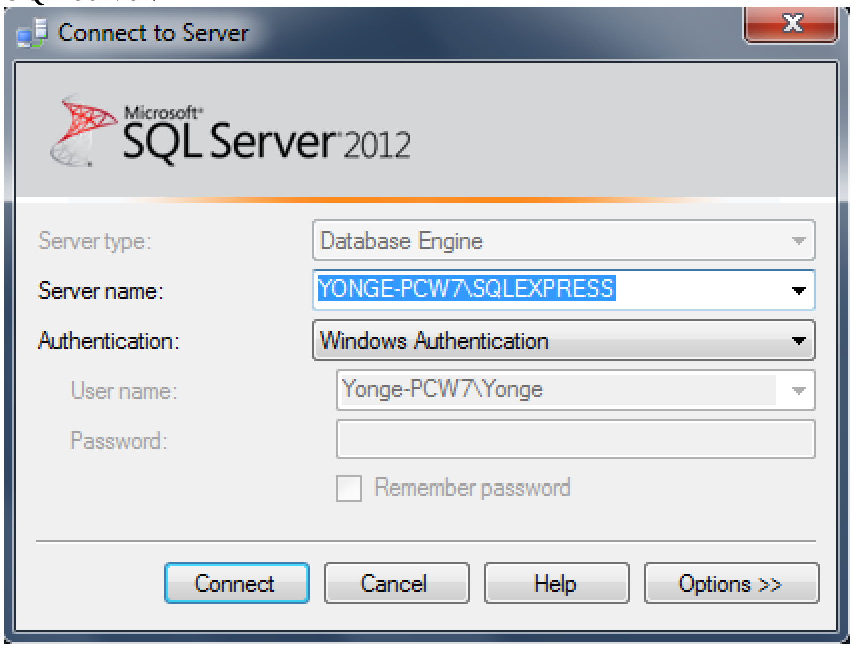
Installation Complete:



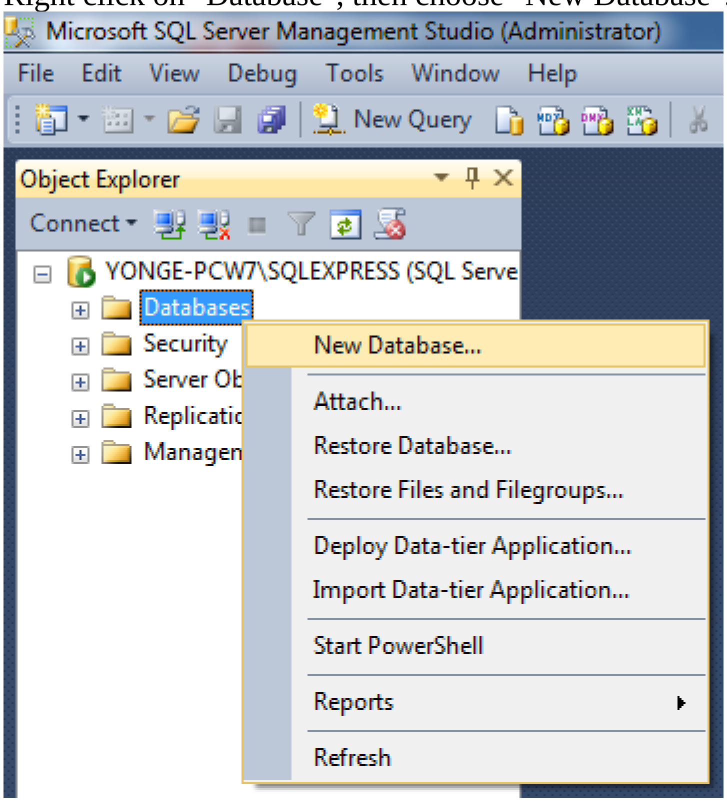
Run "SQL Server Management Studio":



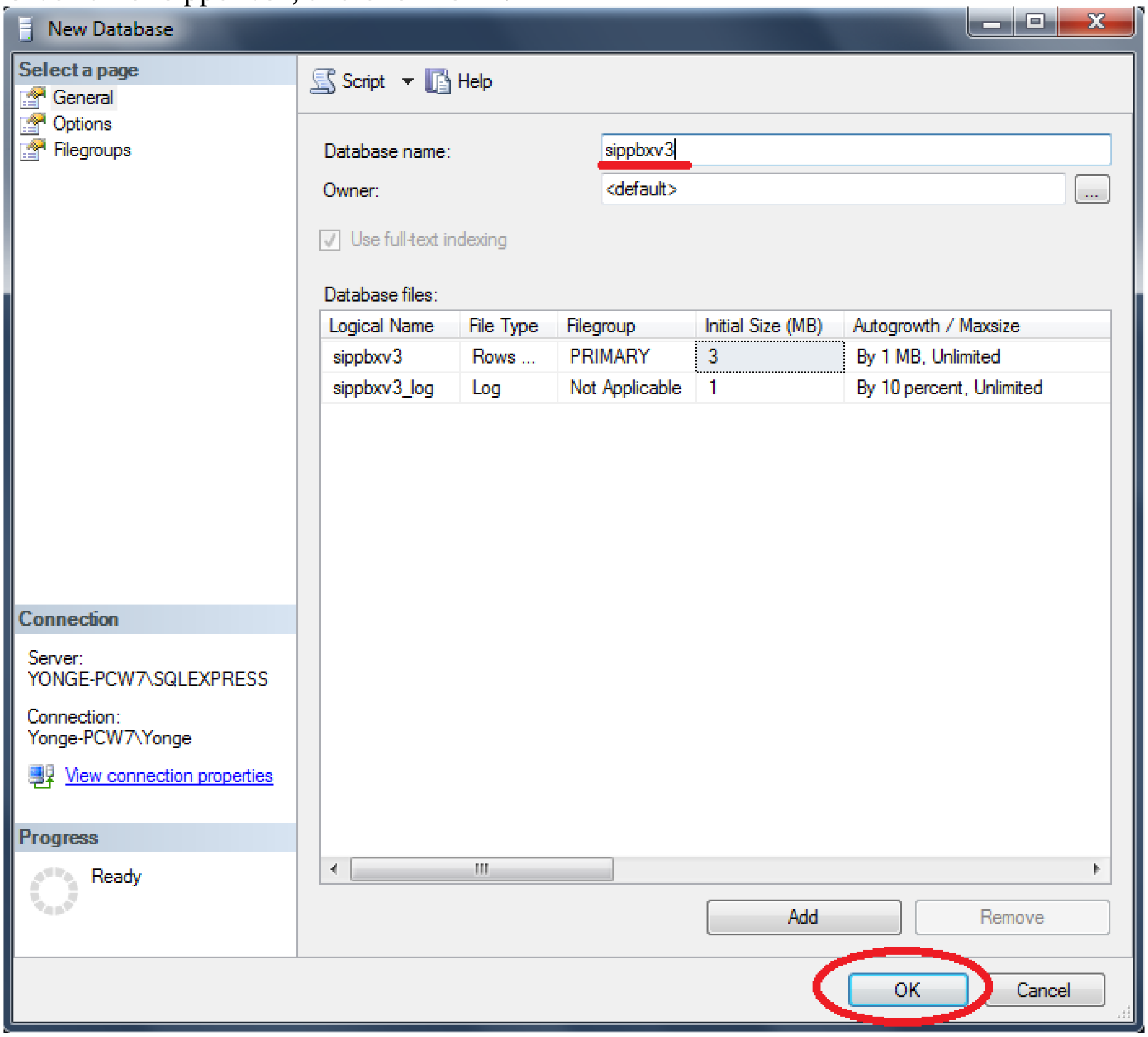
You can use "Windows Authentication" here, and click Next, or use SQL Authentication, then give username sa, password whatever you set when installing SQL server.



Right click on "Database", then choose "New Database":



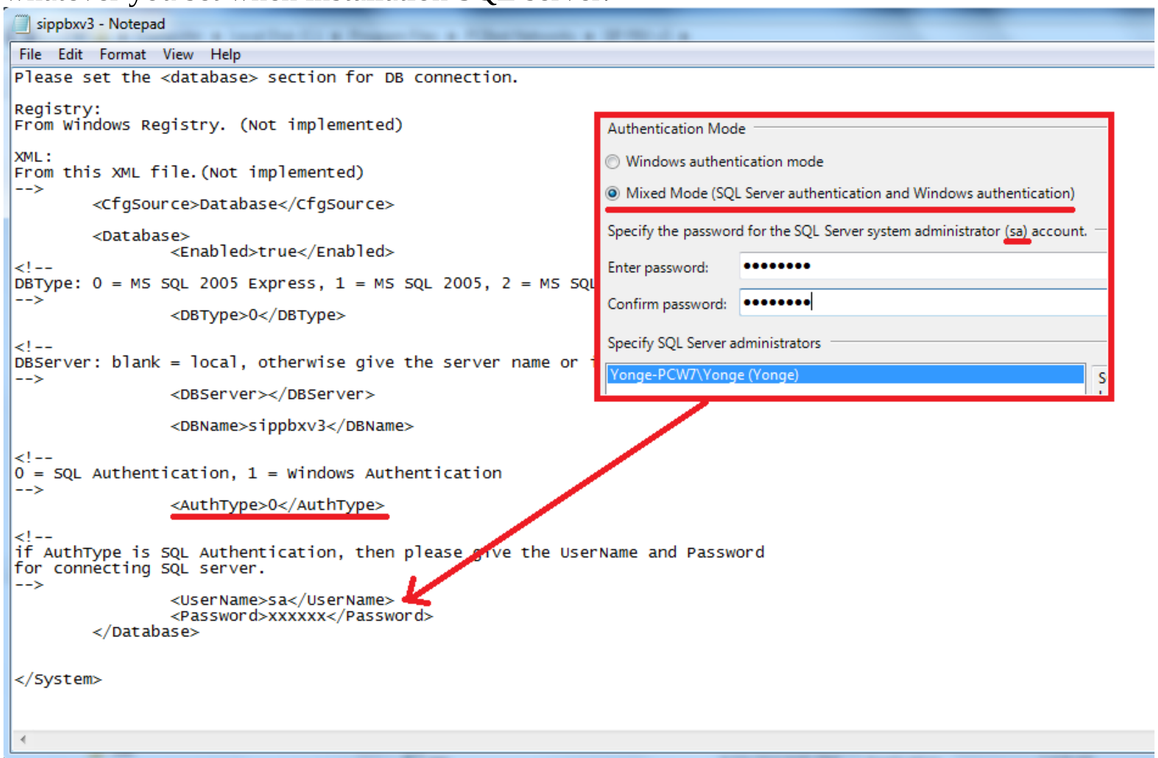
Give name "sippbxv3", and click "OK":



Then database is created, and you can close SQL Server Management Studio.

7. After the database is created, change the configuration file **sippbxv3.xml** for DB connection. The file can be found in PBX installation folder. Use Notepad or any text editor to open it. Under Windows7 or 2008, in order to change this file, you may need to run Notepad as Administrator first, then open **sippbxv3.xml** in order to save.

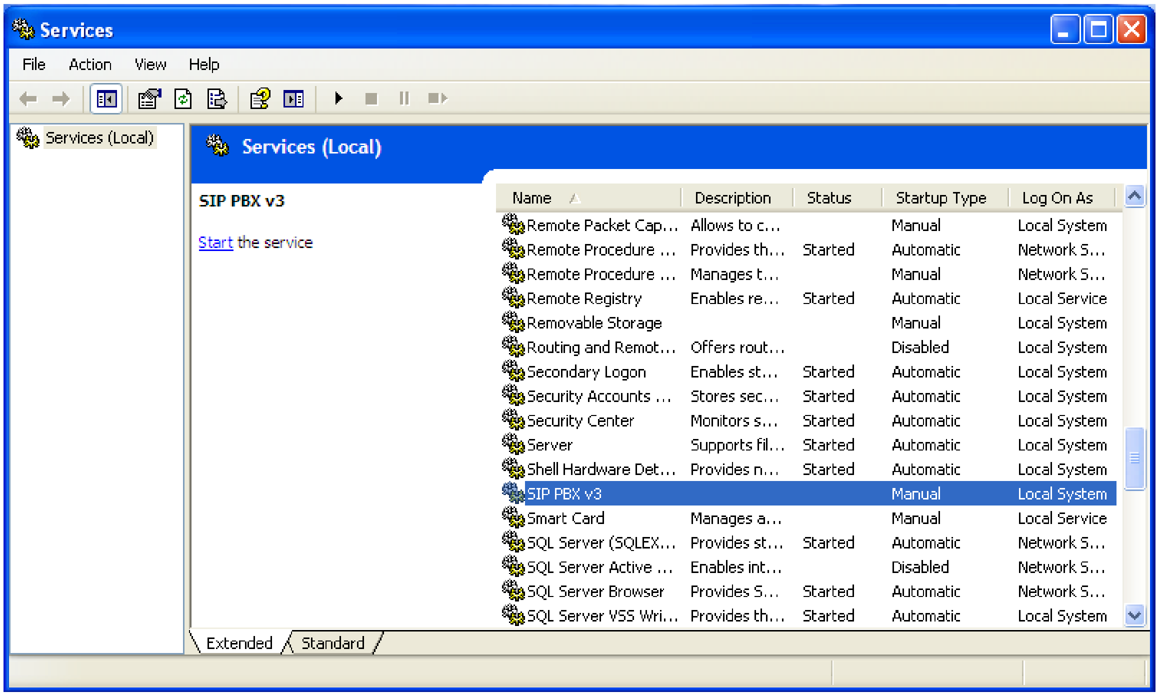
In the file, please **set AuthType to 0**, give UserName sa, and password. The password is whatever you set when installation SQL server.



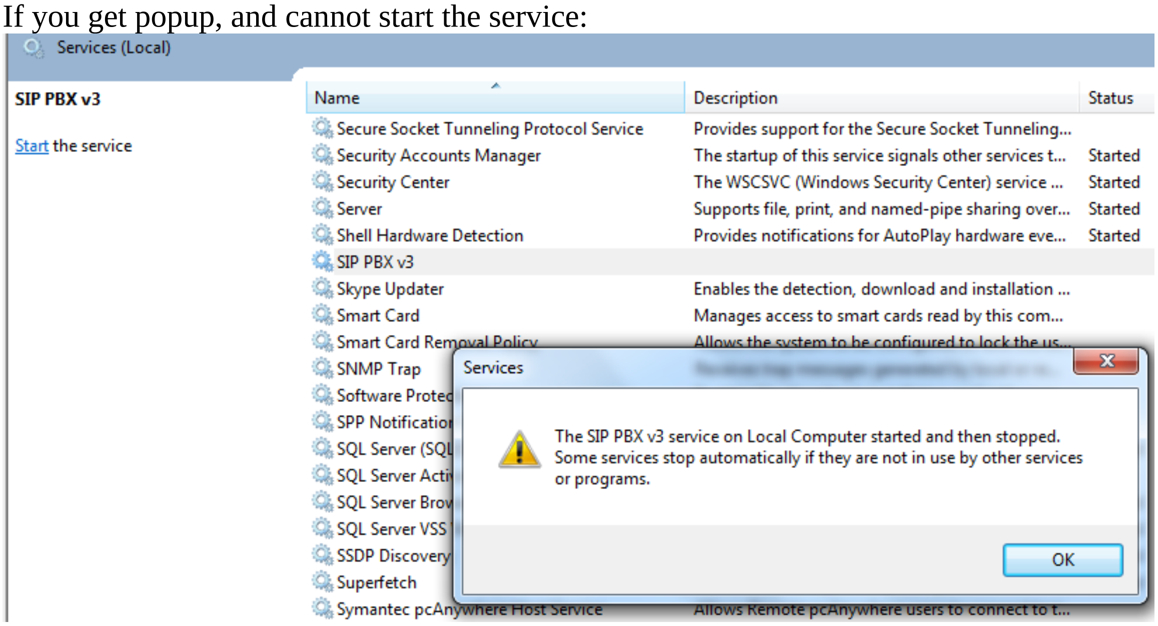
You can set **DBType** to -1 to indicate PBX use a local embedded SQLite DB. Ignore other parameters if you set DBType to -1, but please leave **Enabled** to true.

8. Start SIP PBX v3 service

From Control Panel -> Administrator Tool -> Open Windows Services, then find SIP PBX v3 service, then click start(the triangle button):

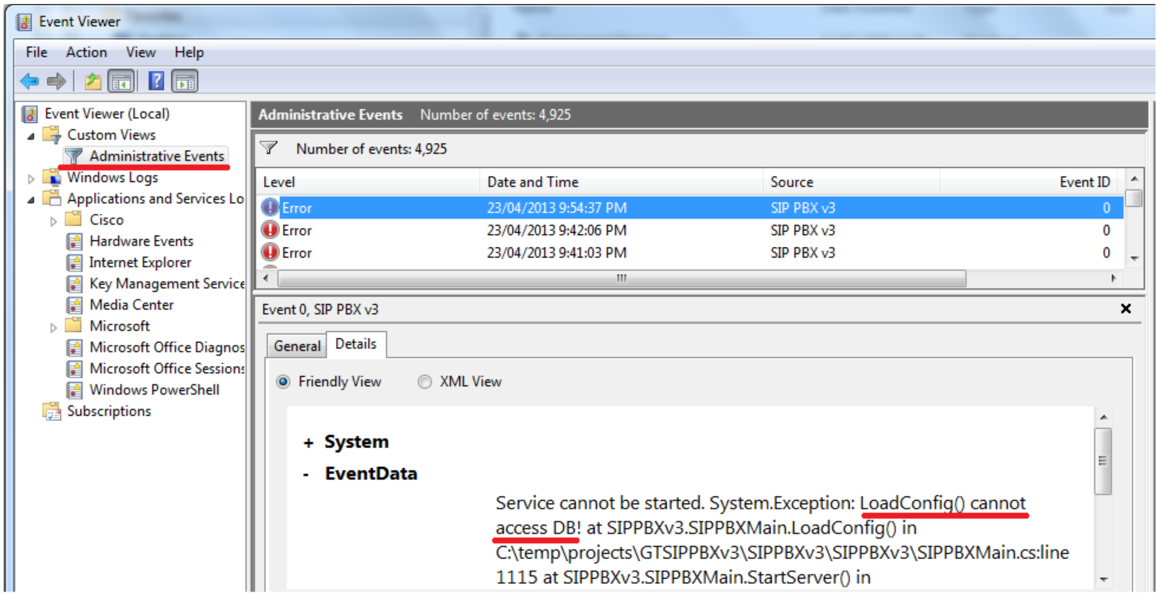


If you get popup, and cannot start the service:

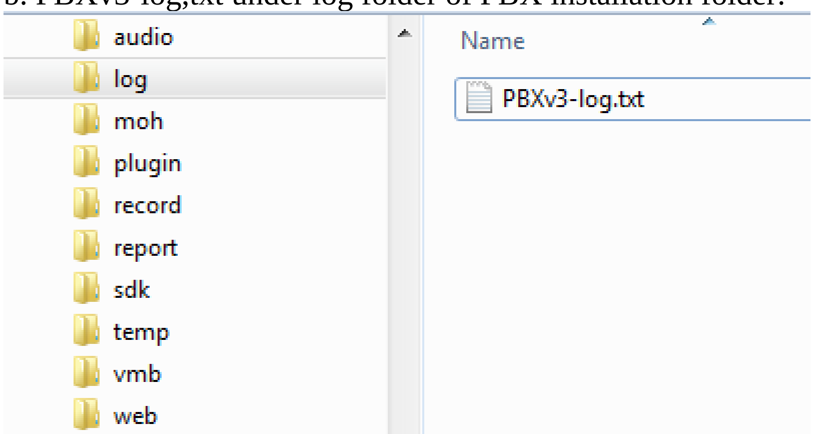


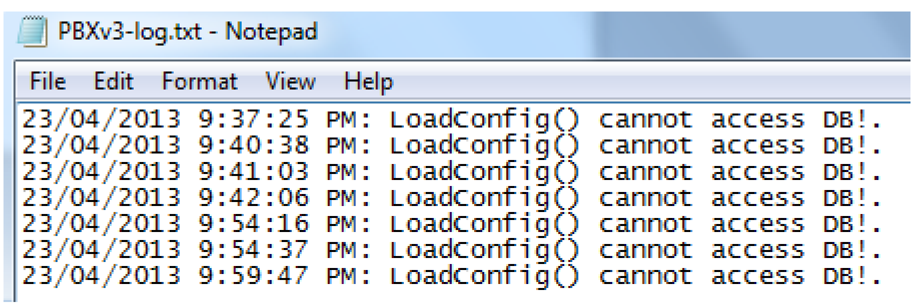
Please check:

a. Event Viewer:



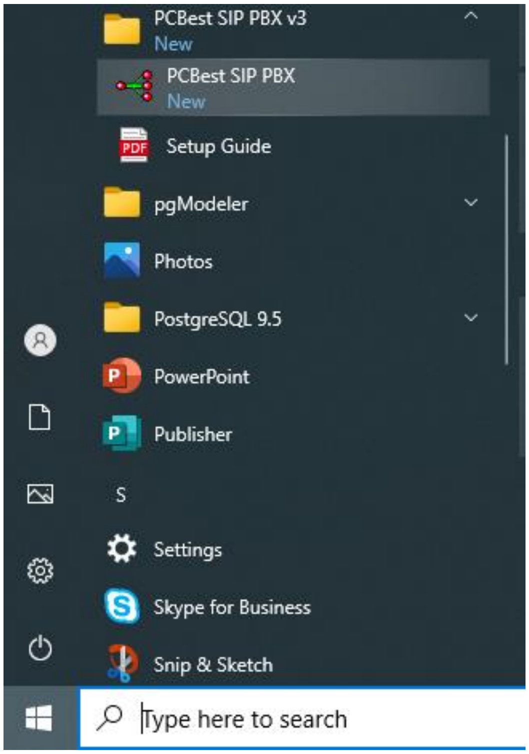
b. PBXv3-log,txt under log folder of PBX installation folder:



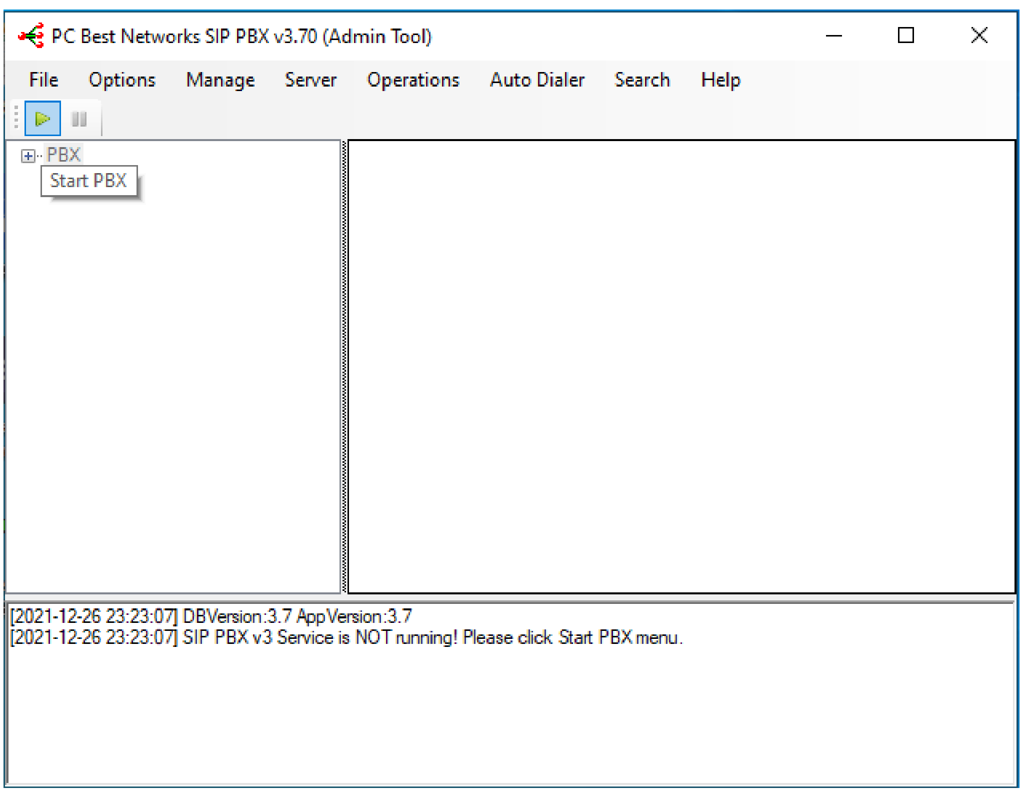


23/04/2013 9:37:25 PM: LoadConfig() cannot access DB!.
23/04/2013 9:40:38 PM: LoadConfig() cannot access DB!.
23/04/2013 9:41:03 PM: LoadConfig() cannot access DB!.
23/04/2013 9:42:06 PM: LoadConfig() cannot access DB!.
23/04/2013 9:54:16 PM: LoadConfig() cannot access DB!.
23/04/2013 9:54:37 PM: LoadConfig() cannot access DB!.
23/04/2013 9:59:47 PM: LoadConfig() cannot access DB!.

9. Run PBX v3 admin tool. **NOTE: if you are using Vista or Windows 7, you need to "Run as administrator" because admin tool needs administrator right to start or stop PBX v3 service.**

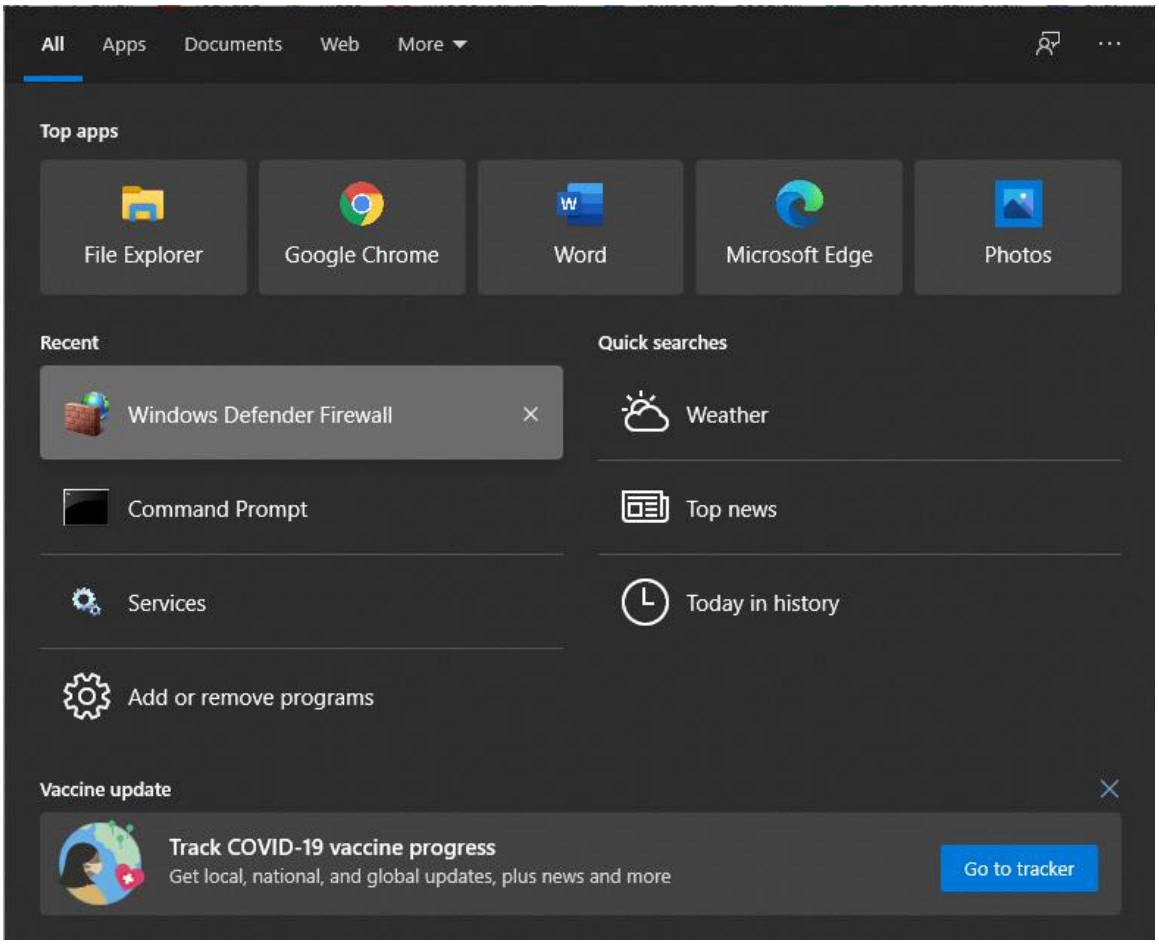


10. If you see this screen, it means it is working. Click the start button to start the service if the service was not started.

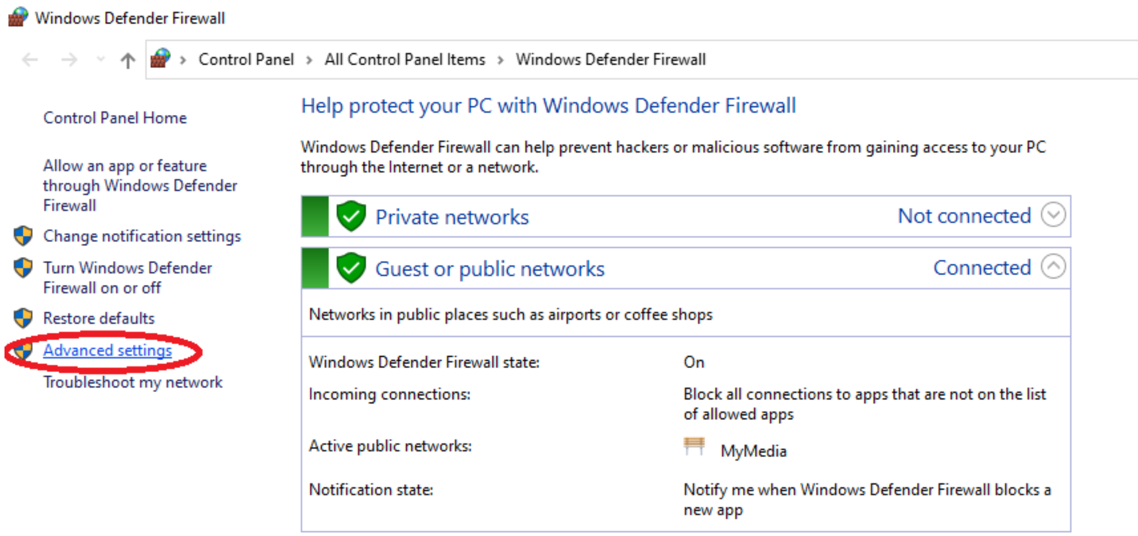


11. Windows Firewall

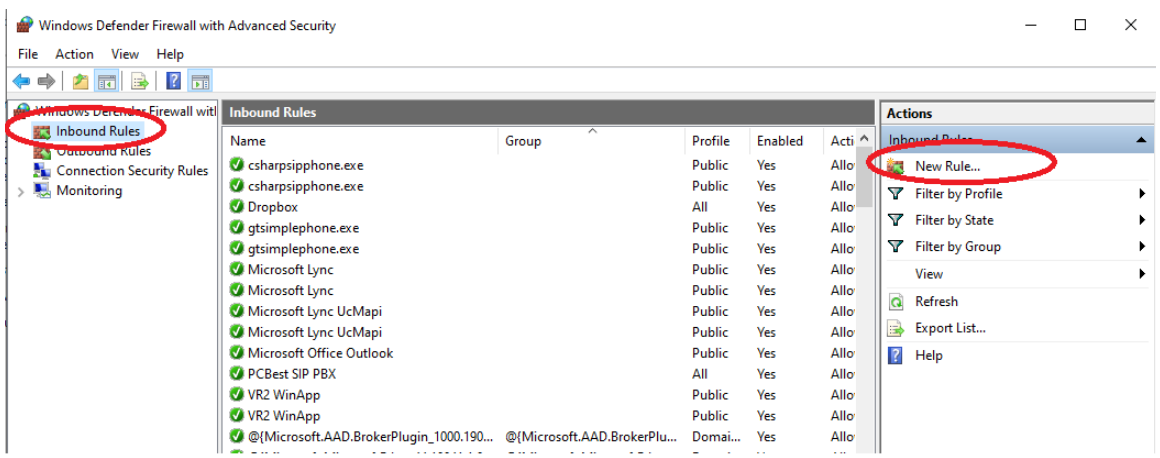
You will need to configure the Windows firewall properly in order to let SIP PBX v3 handle network traffic.



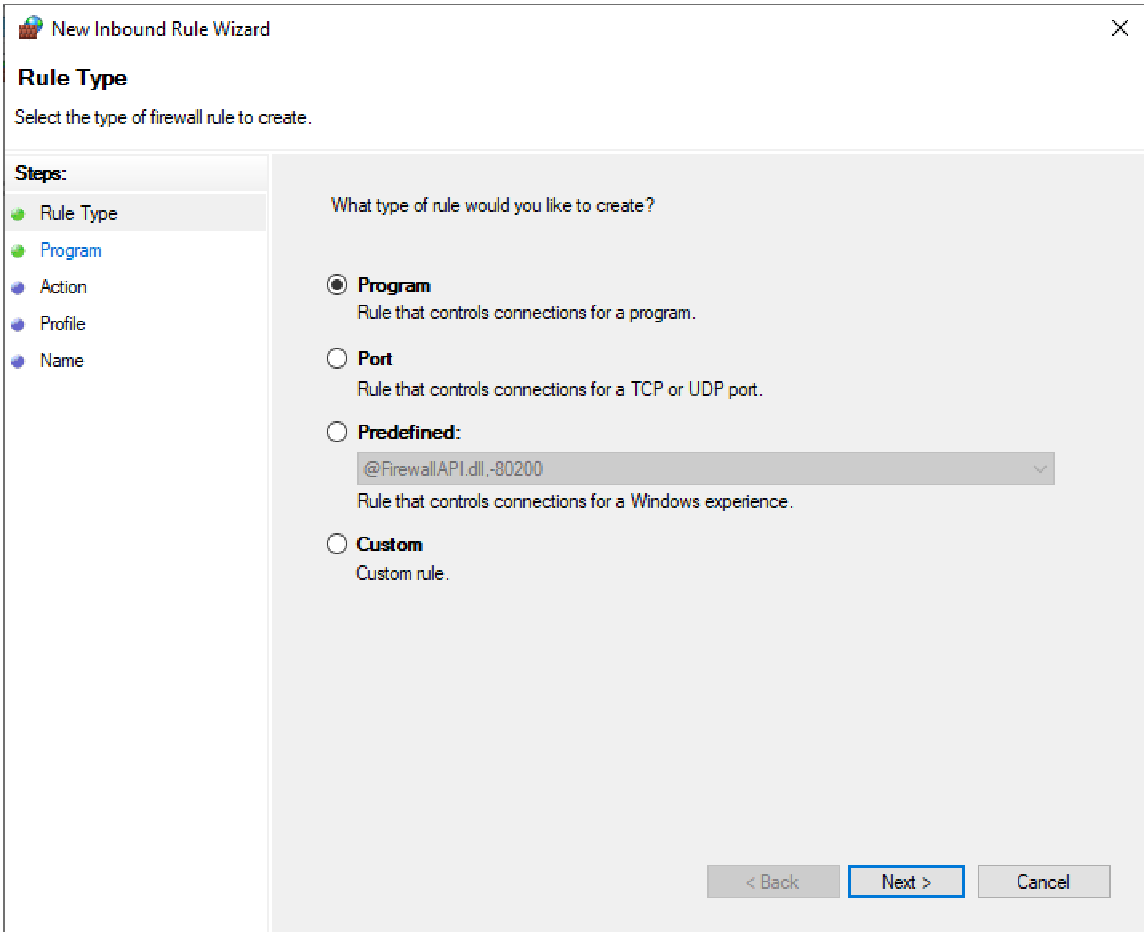
Click “Advanced settings”

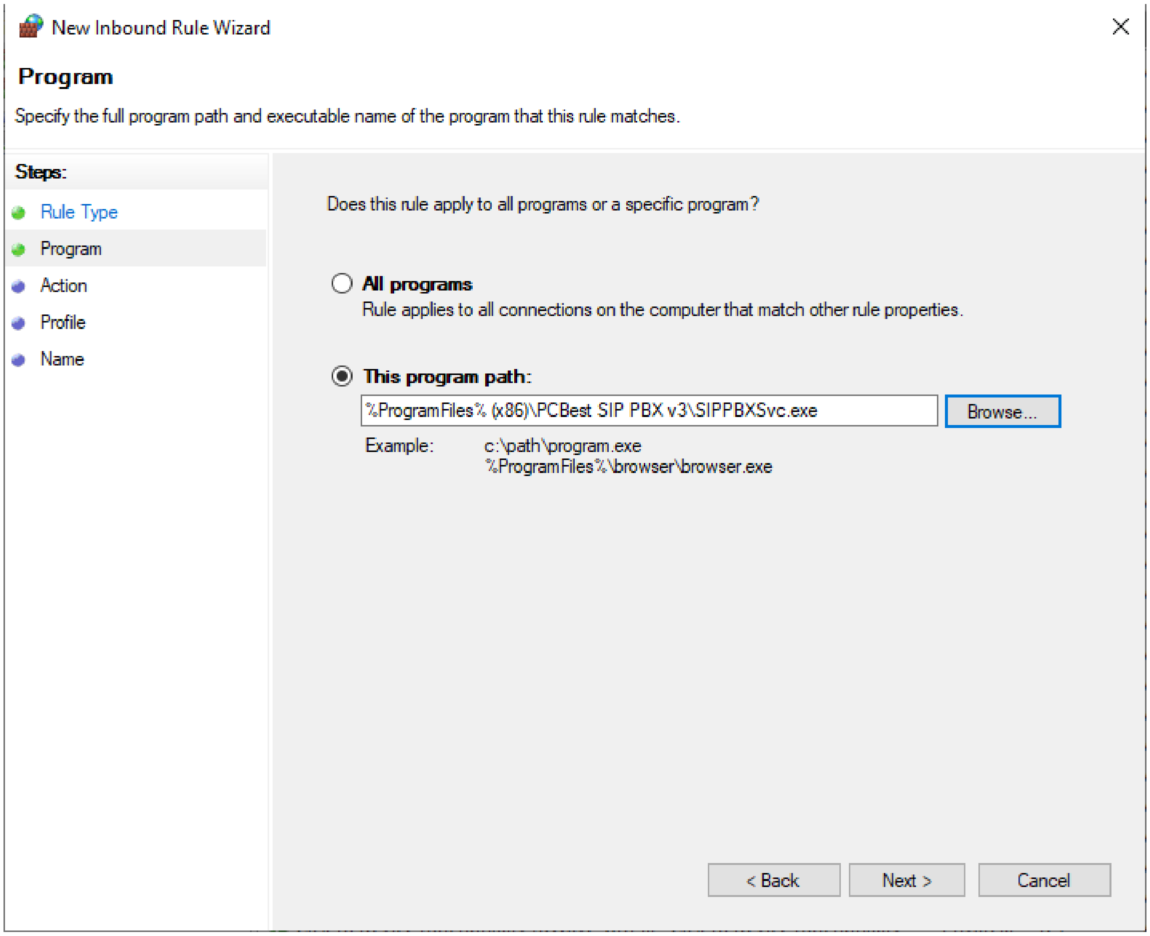


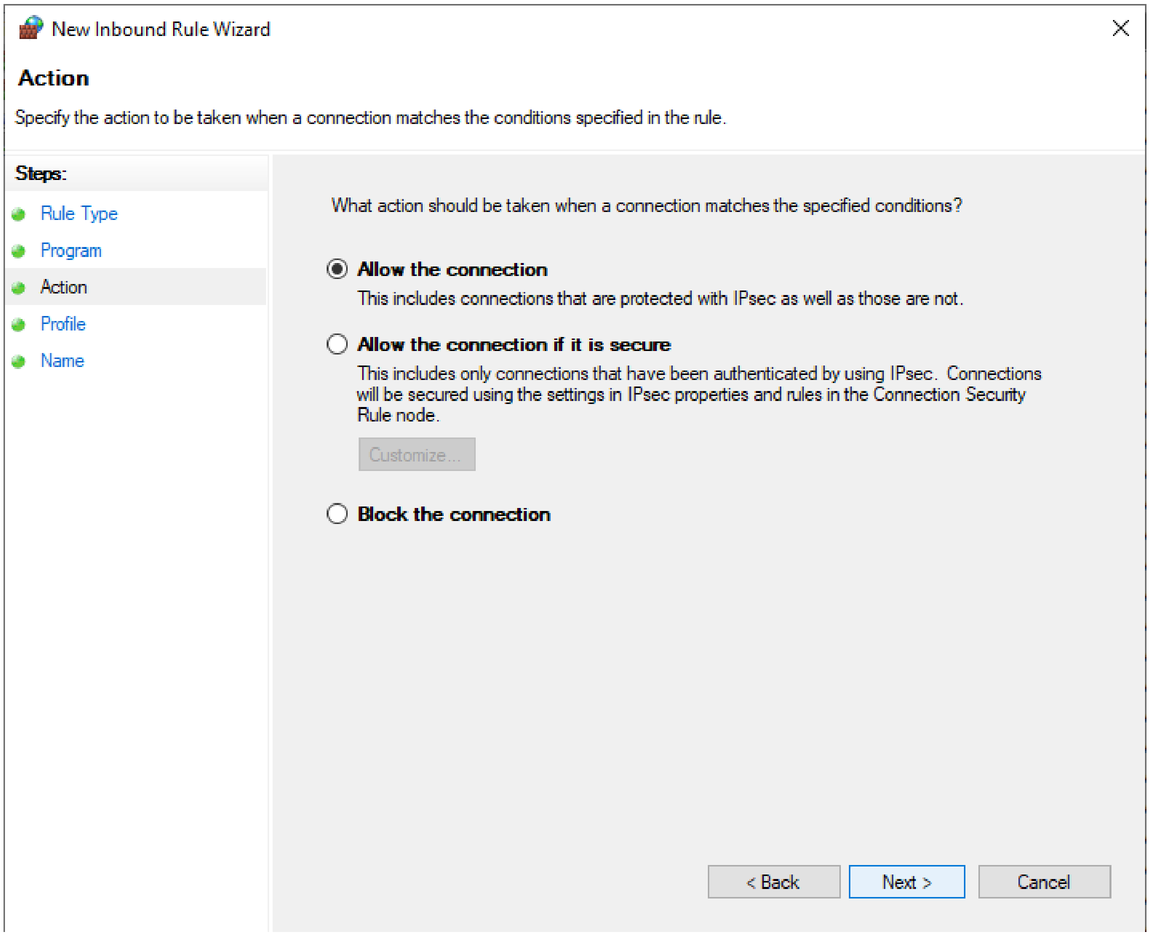
Click “Inbound Rules”, then “New Rule…”.



Choose “program”, then “next”.


New Inbound Rule Wizard
Rule Type
Select the type of firewall rule to create.
Steps:
Rule Type
Program
Action
Profile
Name
What type of rule would you like to create?
Program
Rule that controls connections for a program.
Port
Rule that controls connections for a TCP or UDP port.
Predefined:
@FirewallAPI.dll,-80200
Rule that controls connections for a Windows experience.
Custom
Custom rule.
< Back
Next >
Cancel


New Inbound Rule Wizard
Program
Specify the full program path and executable name of the program that this rule matches.
Steps:
Rule Type
Program
Action
Profile
Name
Does this rule apply to all programs or a specific program?
All programs
Rule applies to all connections on the computer that match other rule properties.
This program path:
%ProgramFiles% (x86)\PCBest SIP PBX v3\SIPPBXSvc.exe
Browse...
Example:
c:\path\program.exe
%ProgramFiles%\browser\browser.exe
< Back
Next >
Cancel



3 PBX Quick Setup Guide

In order to save your time and guide you through the most common scenarios you need to use PCBest SIP PBX for your office environment, this is a quick reference to setup your PBX for Auto Attendant, ACD(Automatical Call Distribution), Outbound Calls, Dial Extension, Virtual Extension, Ring group or Call Parking and etc.

3.1 Common Settings

Before you start, you need to setup the following common settings for all tests.

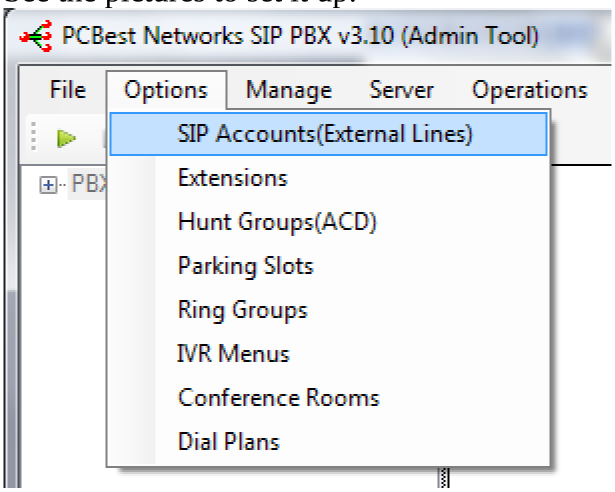
SIP Accounts(External Lines)

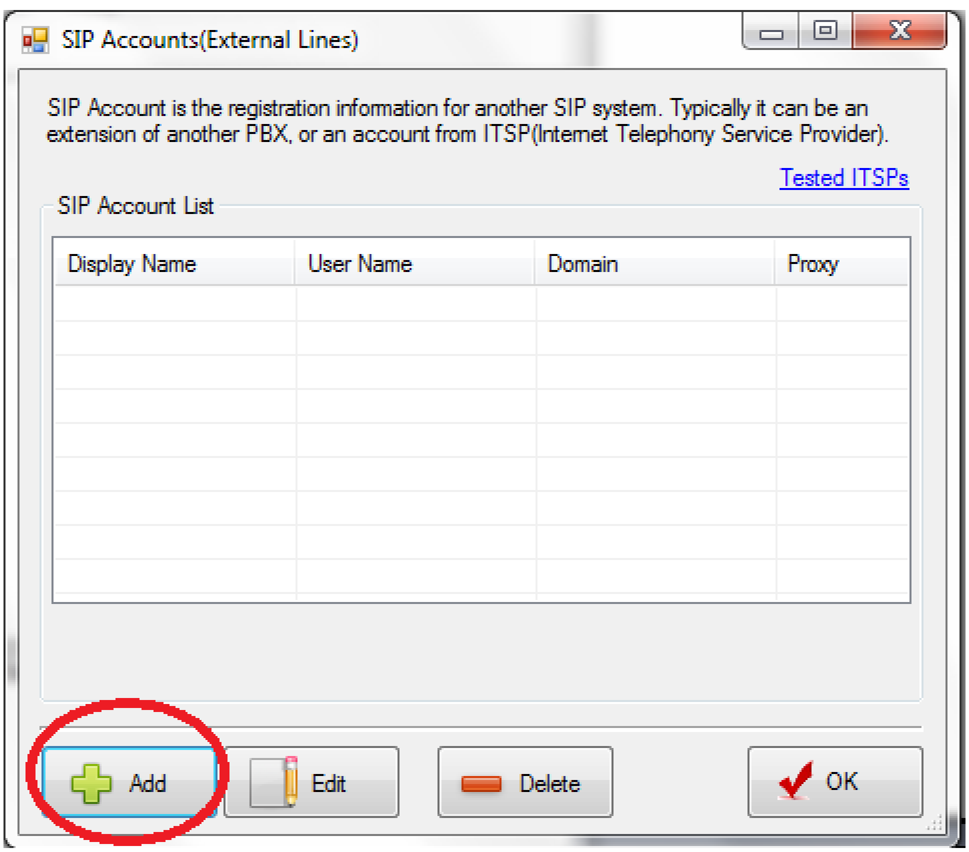
SIP Accounts are the credit info that you can use it to dial out external lines, or receive calls from out lines. For example, you can get a SIP account from ITSP(Internet

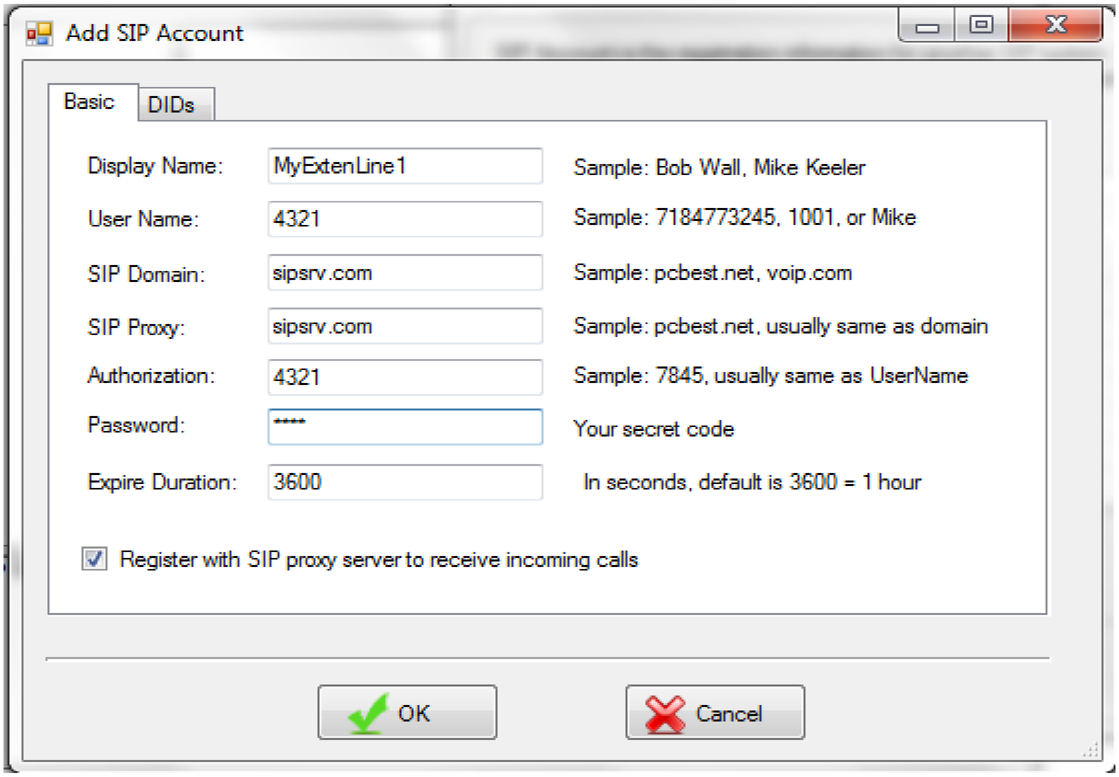
Telephony Service Provider), then you can make calls to regular phone numbers, or receives calls to your DID.

Assume you have a SIP account:
User Name: 4321
Domain: sipsrv.com

See the pictures to set it up:

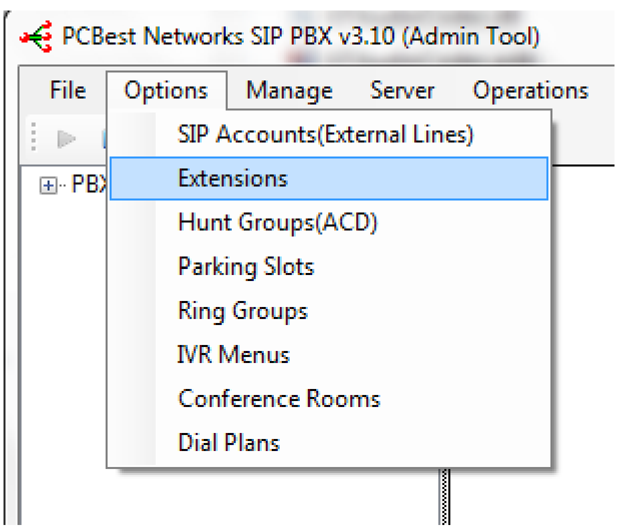


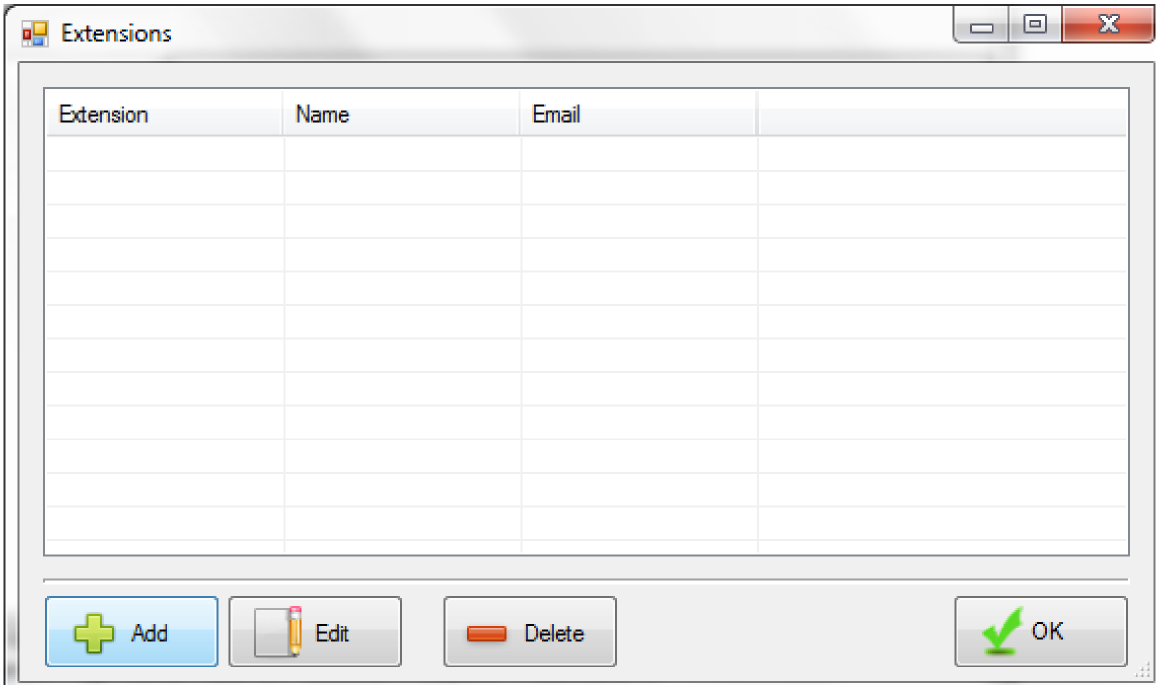


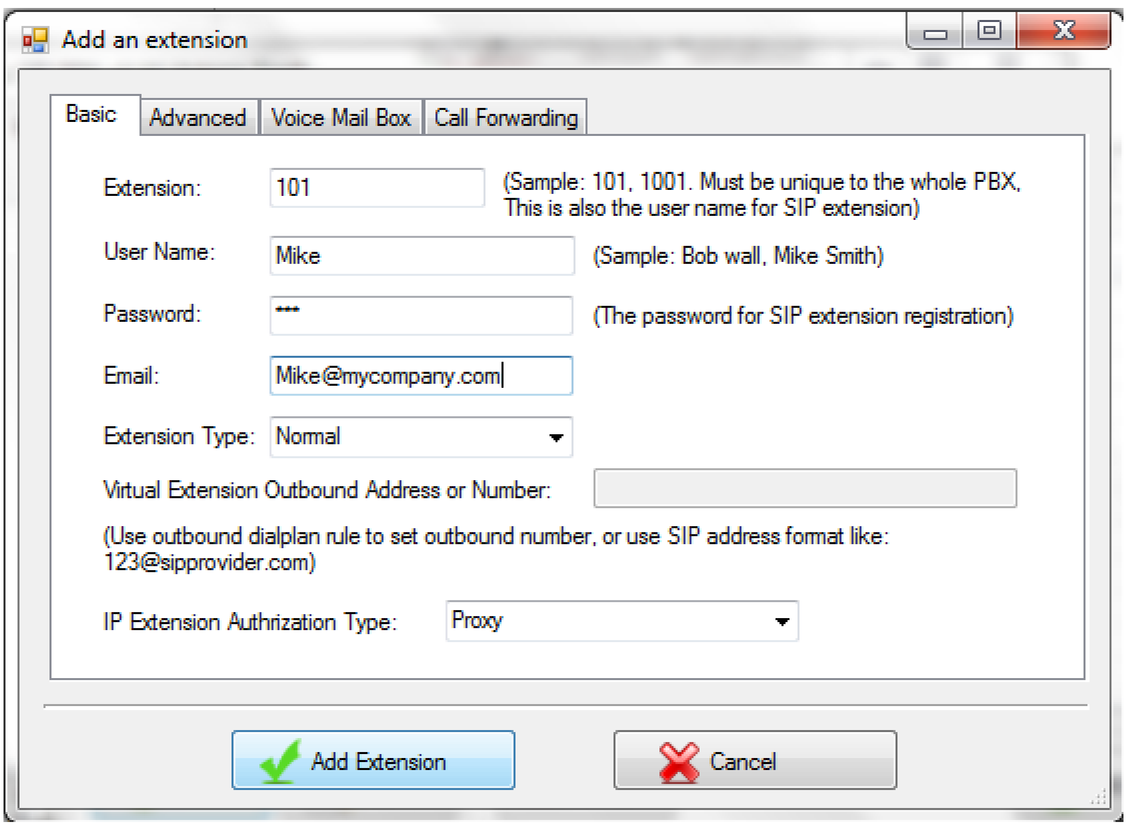


Extensions

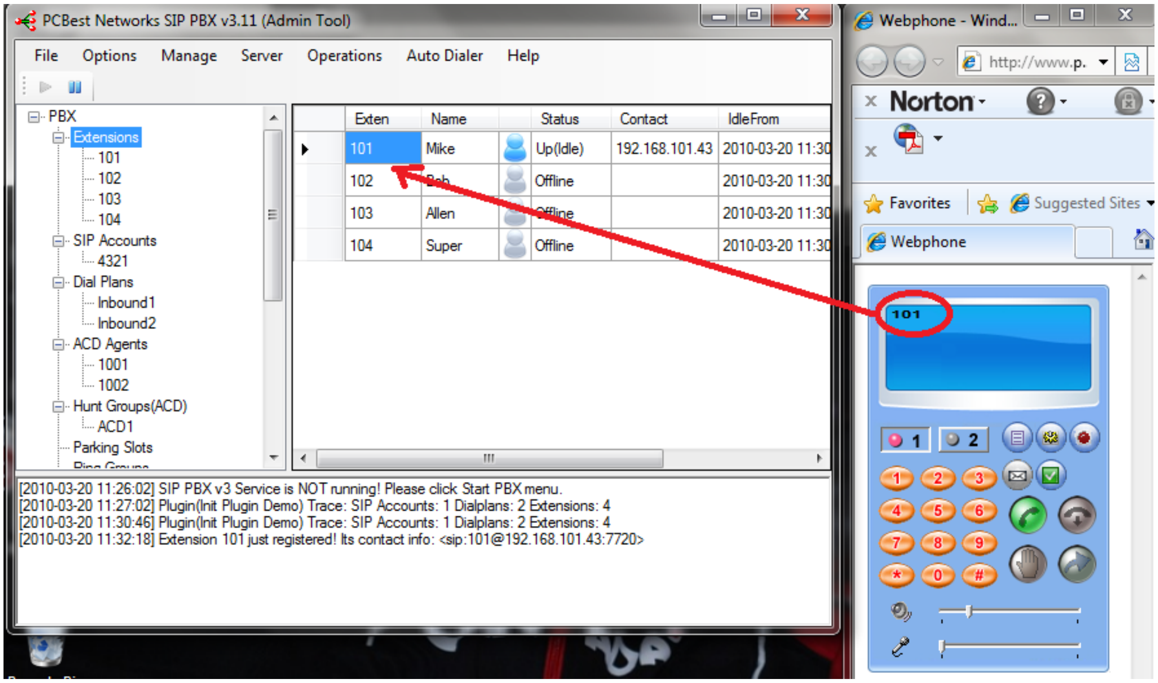
Extensions are internal phones to handle the calls. Usually extension name are three or four digits length, Like 101, 2010. One extension can also be considered as one SIP account for IP phone, or an outline for another PBX. Assume we setup three extensions here.




Extensions
Extension
Name
Email
Add
Edit
Delete
OK

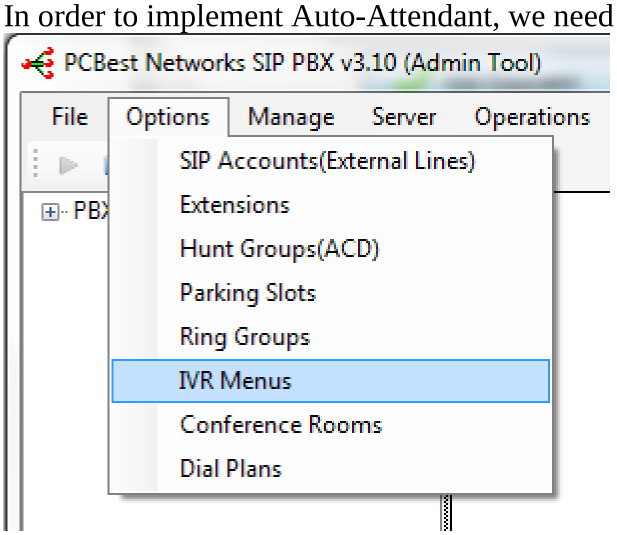

Add an extension
Basic
Advanced
Voice Mail Box
Call Forwarding
Extension:
101
(Sample: 101, 1001. Must be unique to the whole PBX, This is also the user name for SIP extension)
User Name:
Mike
(Sample: Bob wall, Mike Smith)
Password:
(The password for SIP extension registration)
Email:
Mike@mycompany.com
Extension Type:
Normal
Virtual Extension Outbound Address or Number:
(Use outbound dialplan rule to set outbound number, or use SIP address format like: 123@sipprovider.com)
IP Extension Authrization Type:
Proxy
Add Extension
Cancel

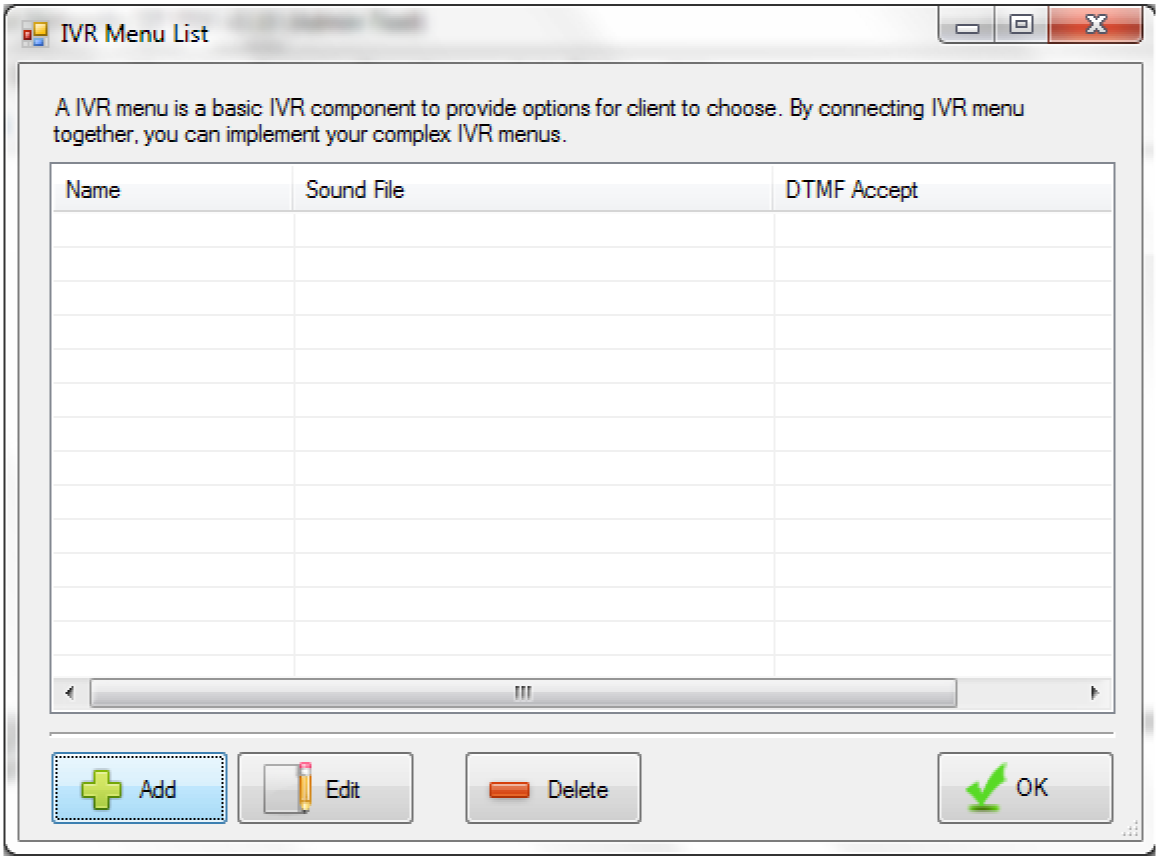
After you have setup three extensions 101, 102, and 103, you need to have 3 ipphones or computers to register on PBX to work as extensions. You can use any SIP hardware phones or softphones, like PCBest SIP ActiveX phone here: http://www.pcbest.net/activex.php

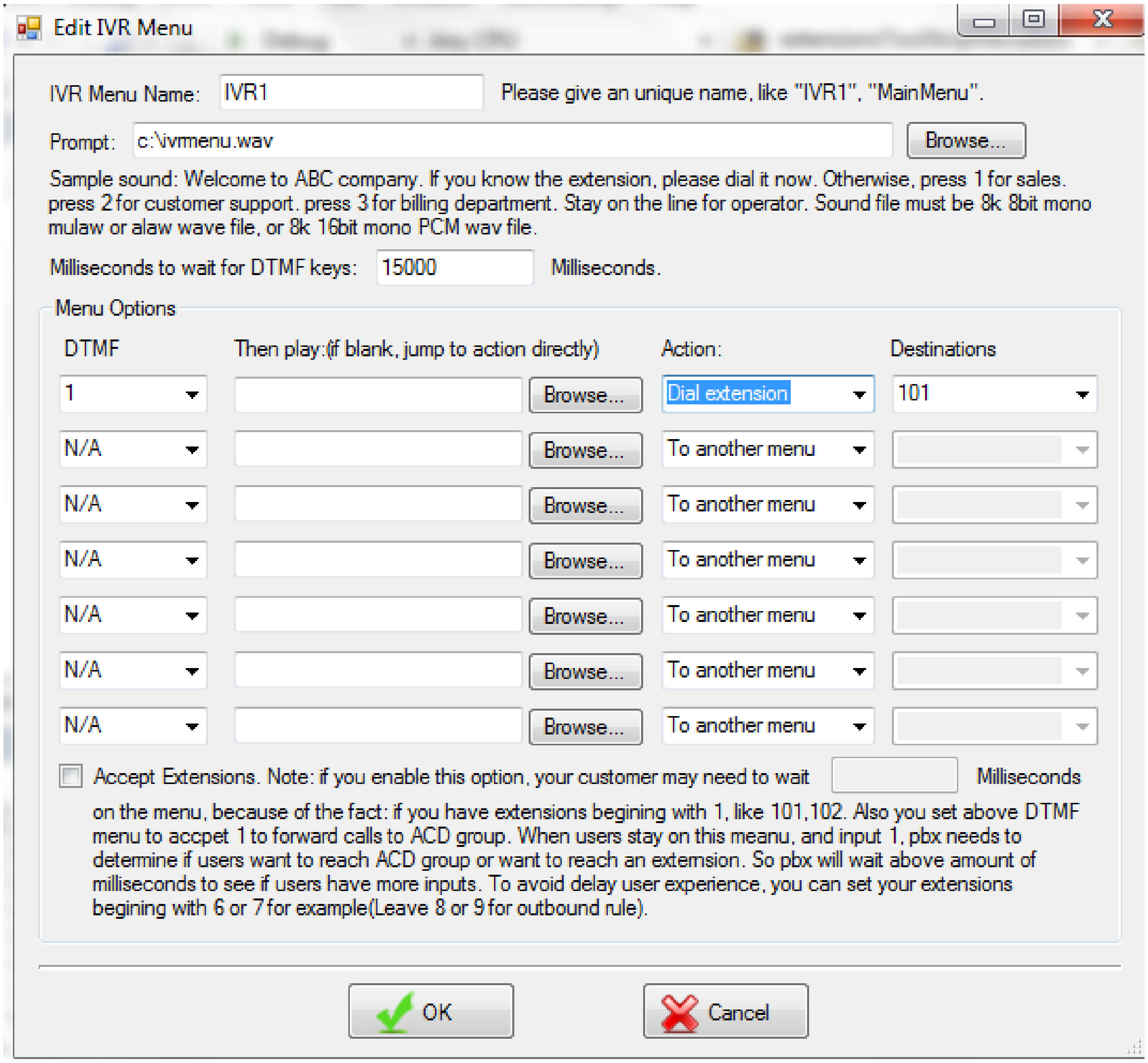


3.2 Auto Attendant

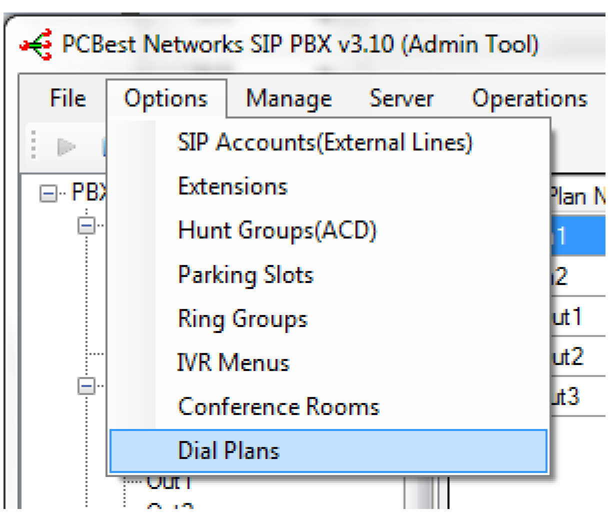
In order to implement Auto-Attendant, we need to set an IVR Menu first to play prompts.

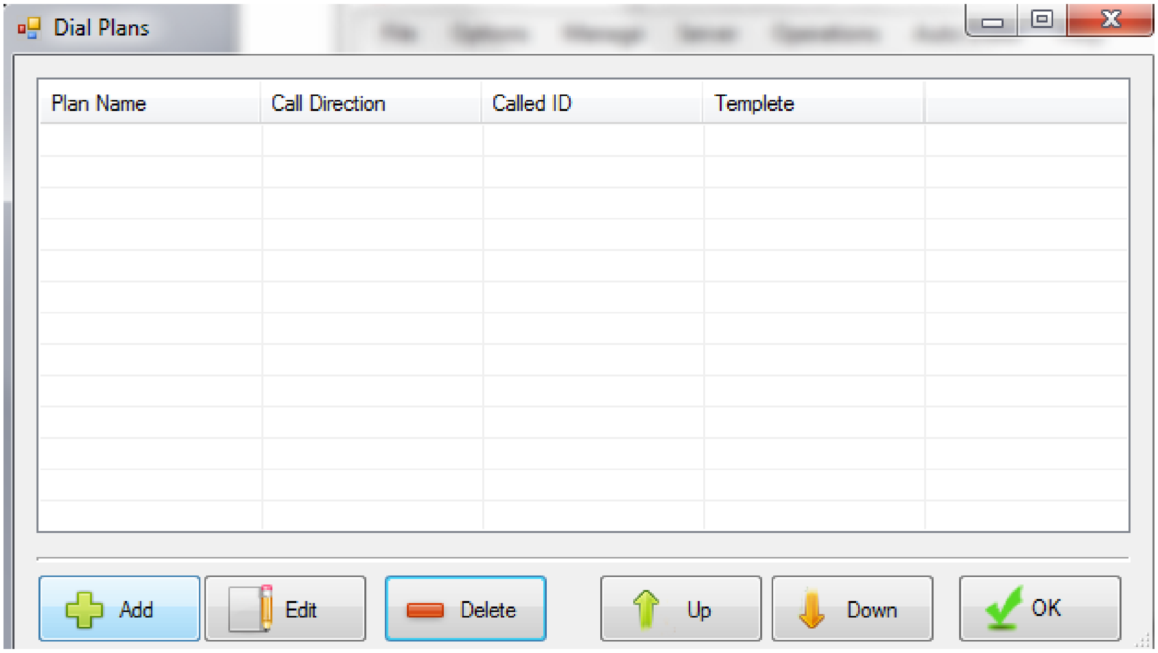



IVR Menu List
A IVR menu is a basic IVR component to provide options for client to choose. By connecting IVR menu together, you can implement your complex IVR menus.
Name
Sound File
DTMF Accept
Add
Edit
Delete
OK

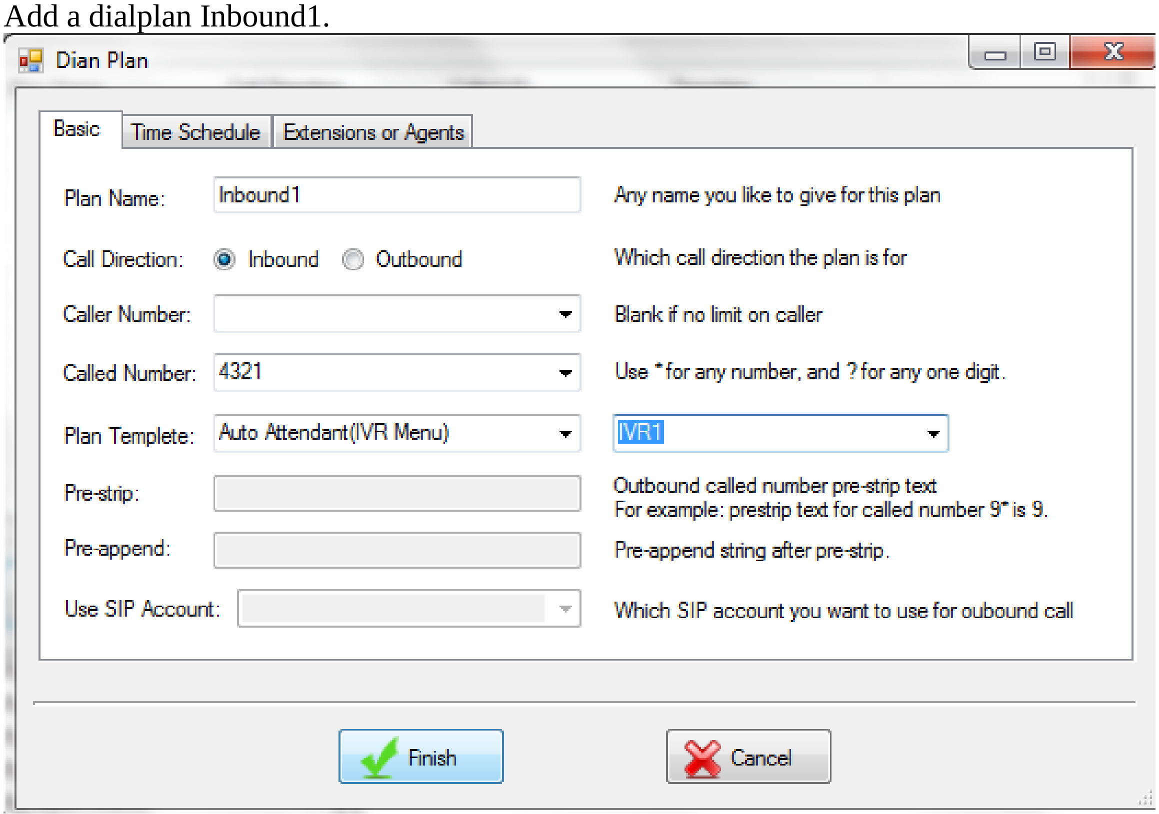


Then we need to setup an inbound dialplan to connect incoming calls into this IVR menu.





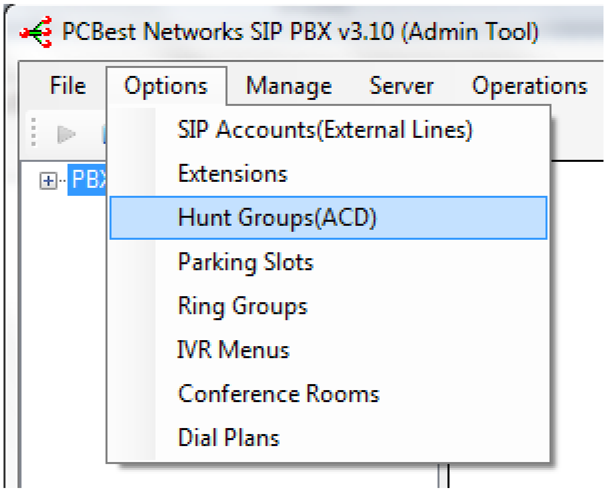
Add a dialplan Inbound1.

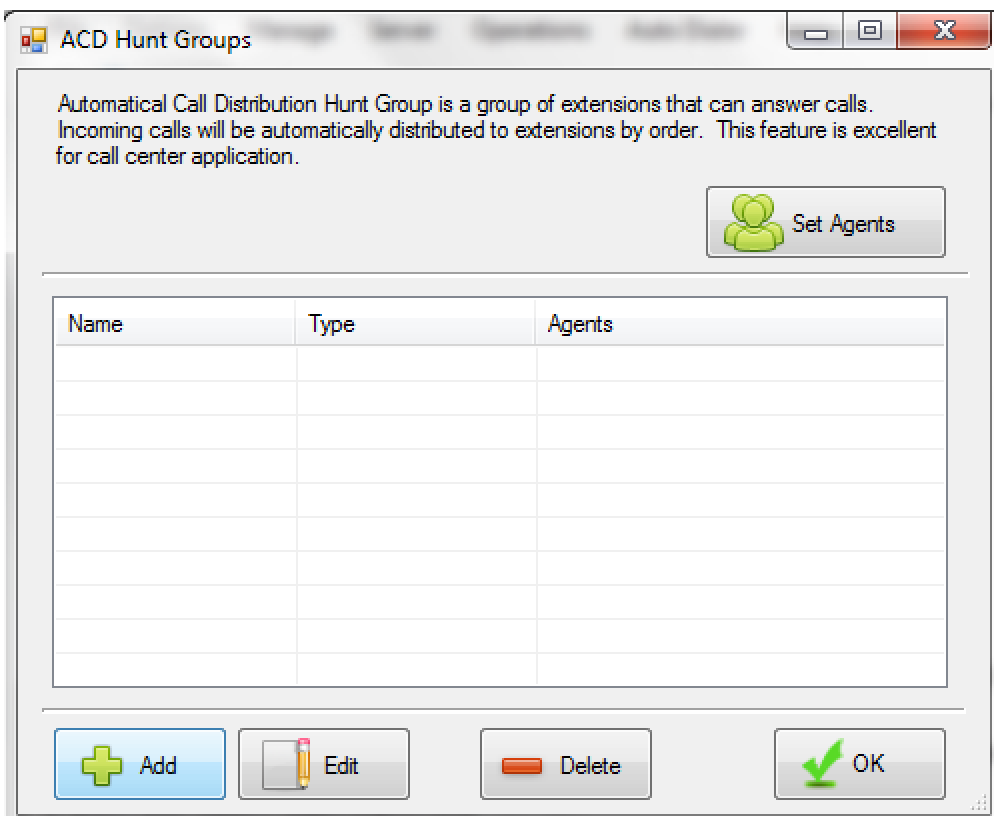


Then when you dial the DID that SIP account 4321 is linked, it will use Dialplan "Inbound1" to handle the call, and call goes to IVR menu "IVR1".

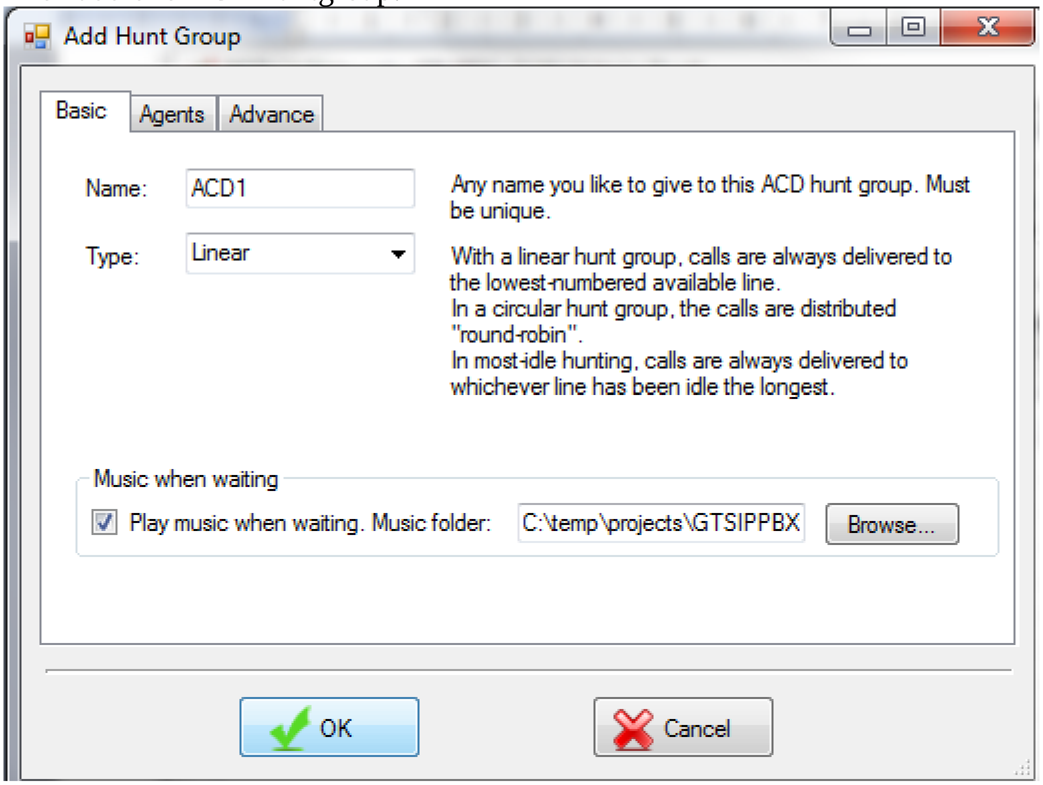
3.3 ACD(Automatical Call Distribution)

ACD is widely used for call centers. Calls will be automatically queued in ACD group(also called huntgroup), and PBX will try to reach an extension or an agent to answer the call on first in first out order. In order to implement ACD, we need to create an ACD group first.

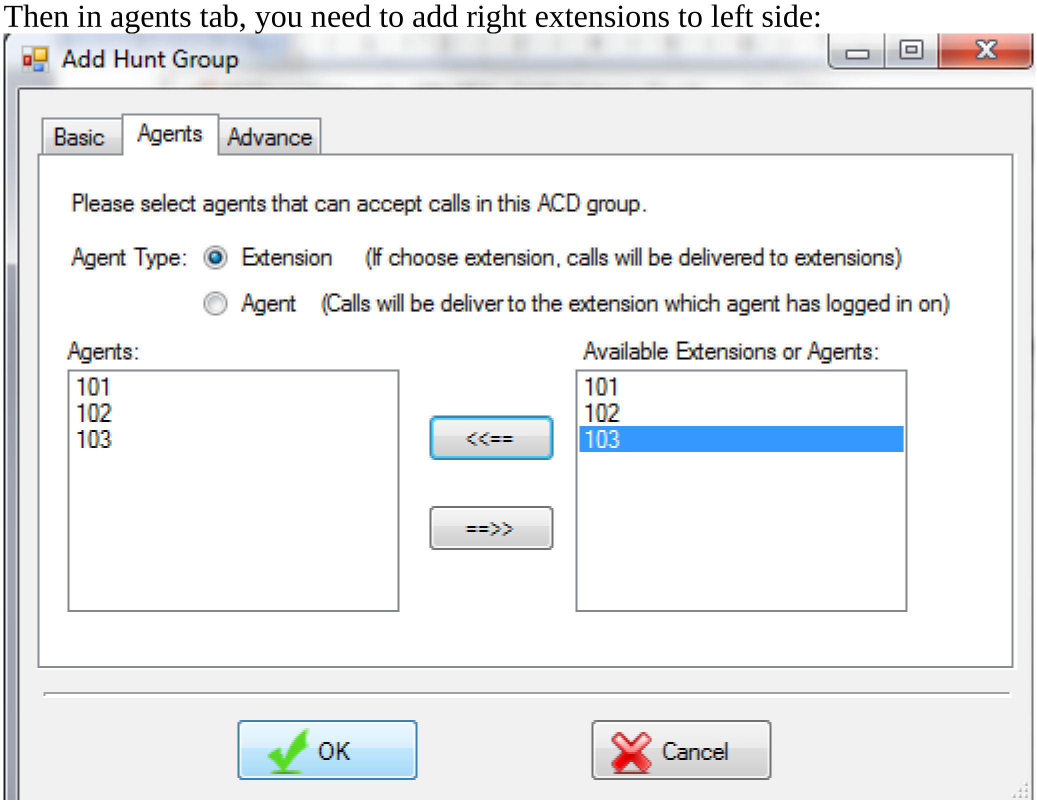




Then add one ACD huntgroup:

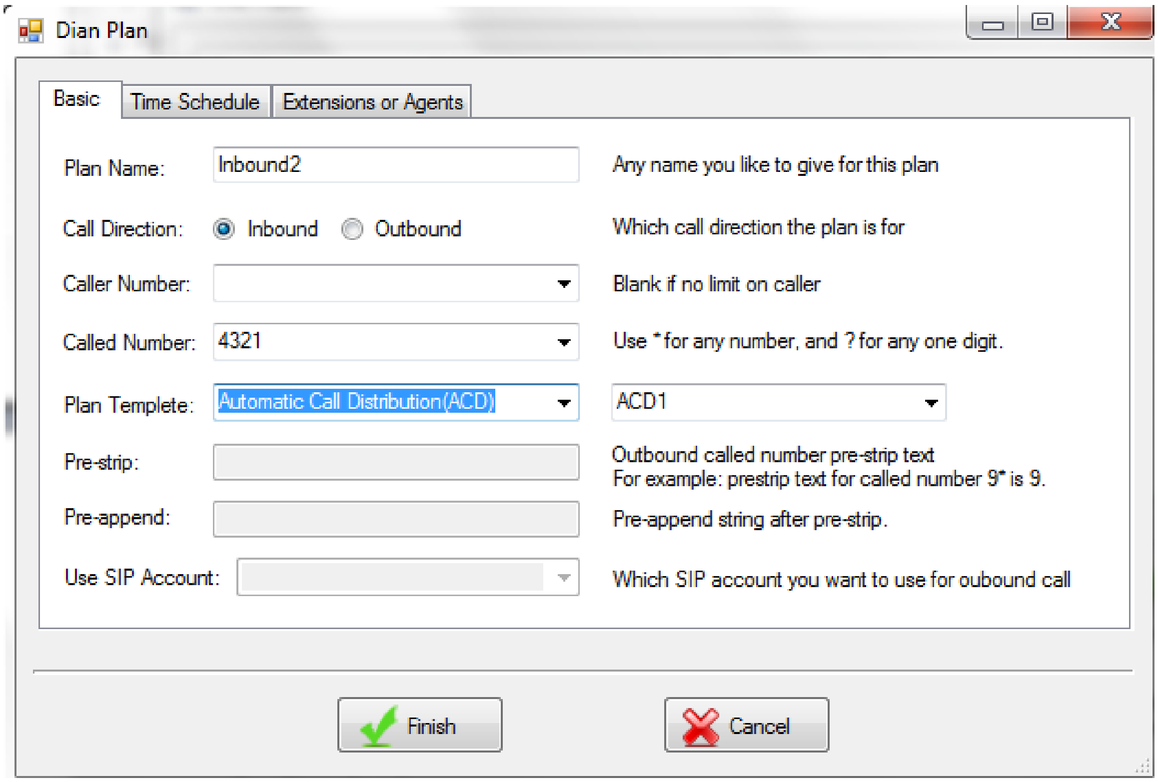


Then in agents tab, you need to add right extensions to left side:



Then click OK.

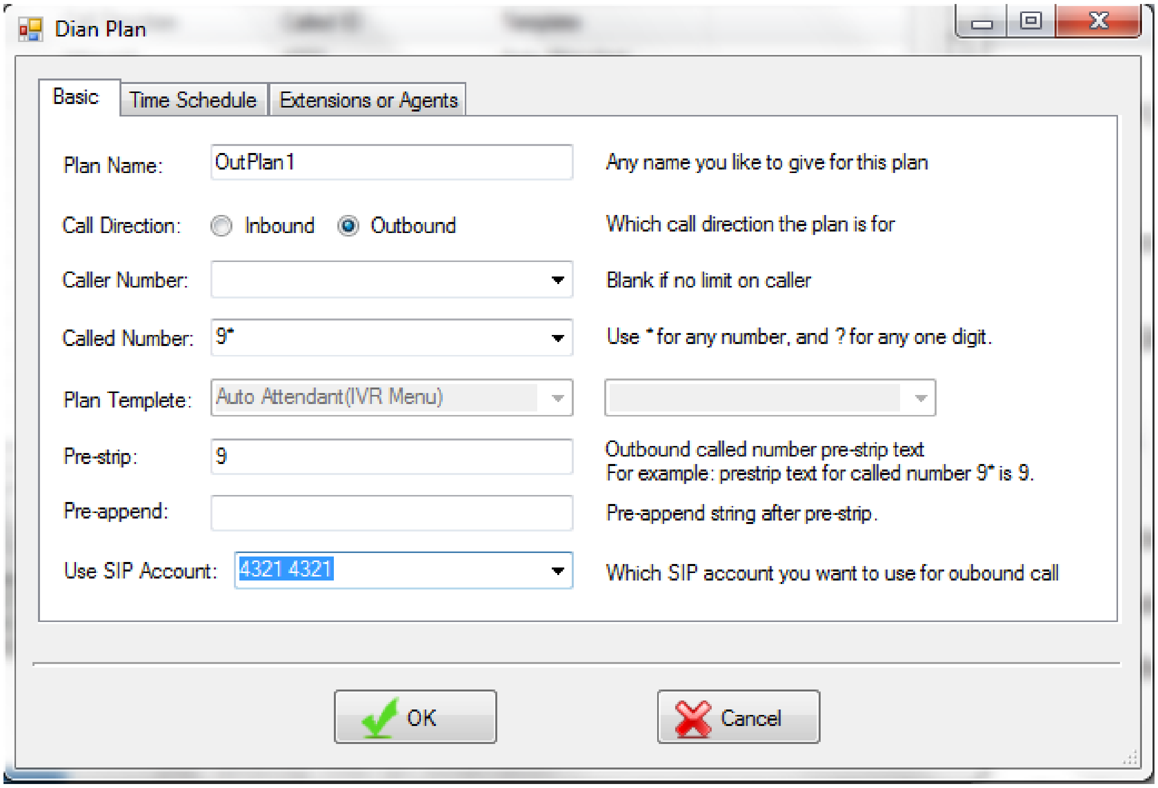
Again, we need to setup an inbound dialplan to connect inbound calls to this ACD huntgroup. Assume we add an inbound dialplan Inbound2 to handle this situation.



Then any calls goes to 4321 SIP account will be forwarded to ACD1.

3.4 Outbound Calls

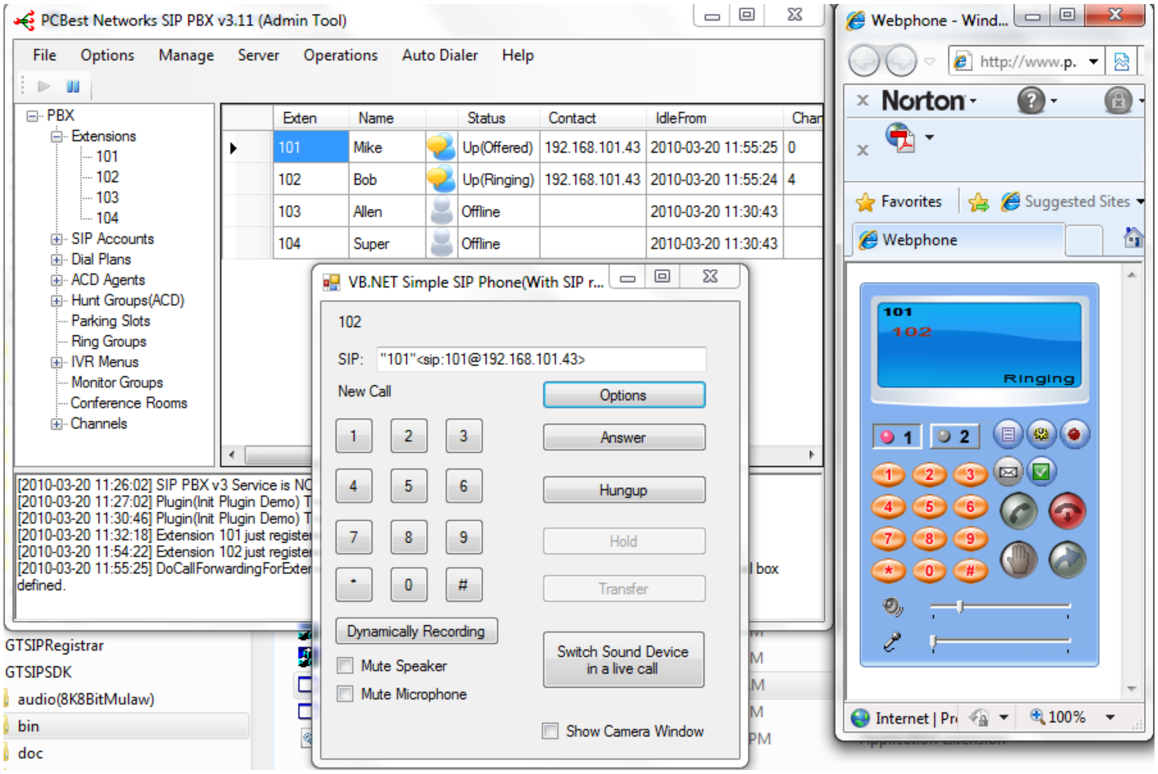
Add a dialplan. Give an plan name like OutPlan1. Set it to outbound type.



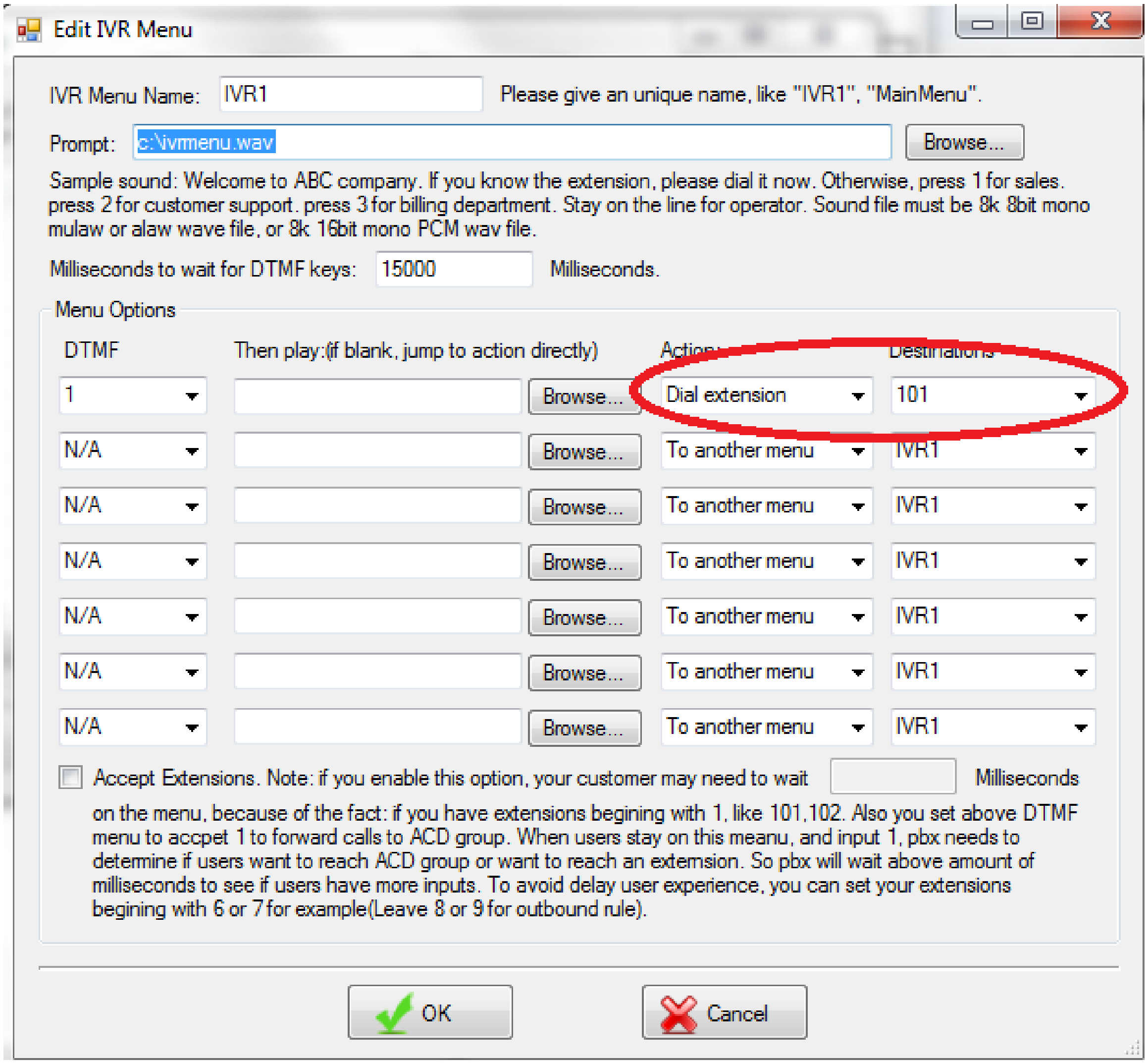
Set called number as 9*, and set the SIP account you want to use for dialing out.
Set pre-strip as 9.
It means that any calls go into PBX, which called id starts with 9, the PBX will regard it as an outbound call. PBX will take 9 in the front of called number, and use SIP account 4321 we created to dial out.
On the sip phone client 101, please dial 9x(x is real phone number you want to reach outside), then PBX should be able to route the call to outside.

3.5 Dial Extension

Extension to extension calls:
You don't need to create any dialplan for extension to extension calls. Assume you have 101 and 102 softphone setup and registered on the PBX. On the softphone 101, you dial 102, then

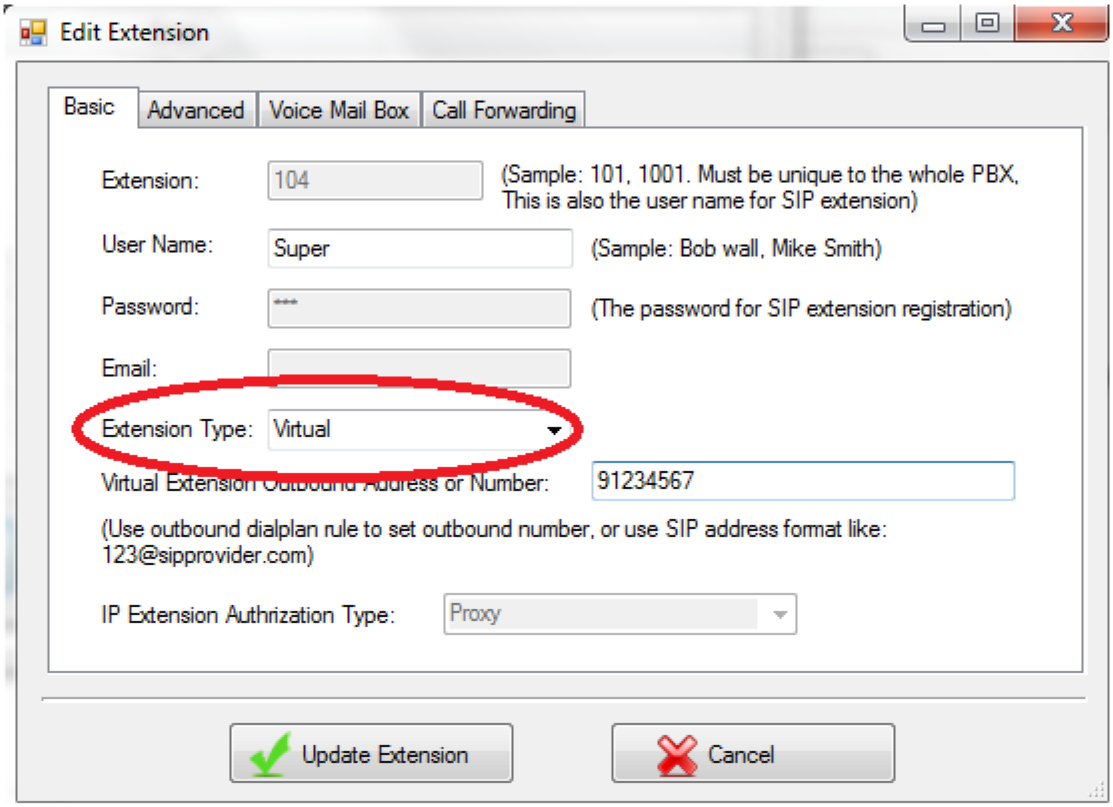


Dial to extension from other options(ACD, IVR menu, …)



3.6 Virtual Extension

Virtual extension is a kind of extension which pointed to an outside phone number. Let us create an extension which has virtual extension type.

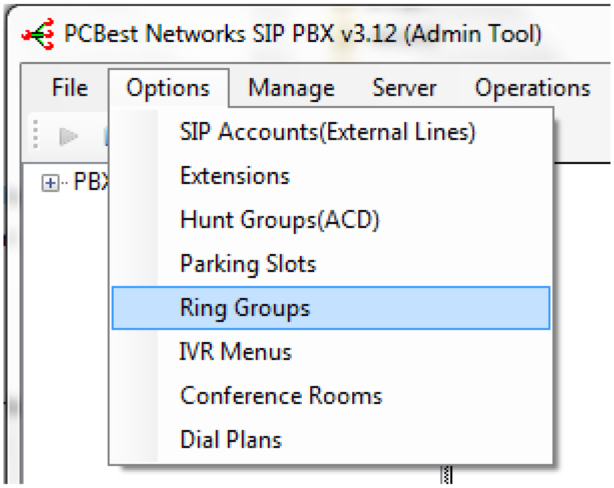


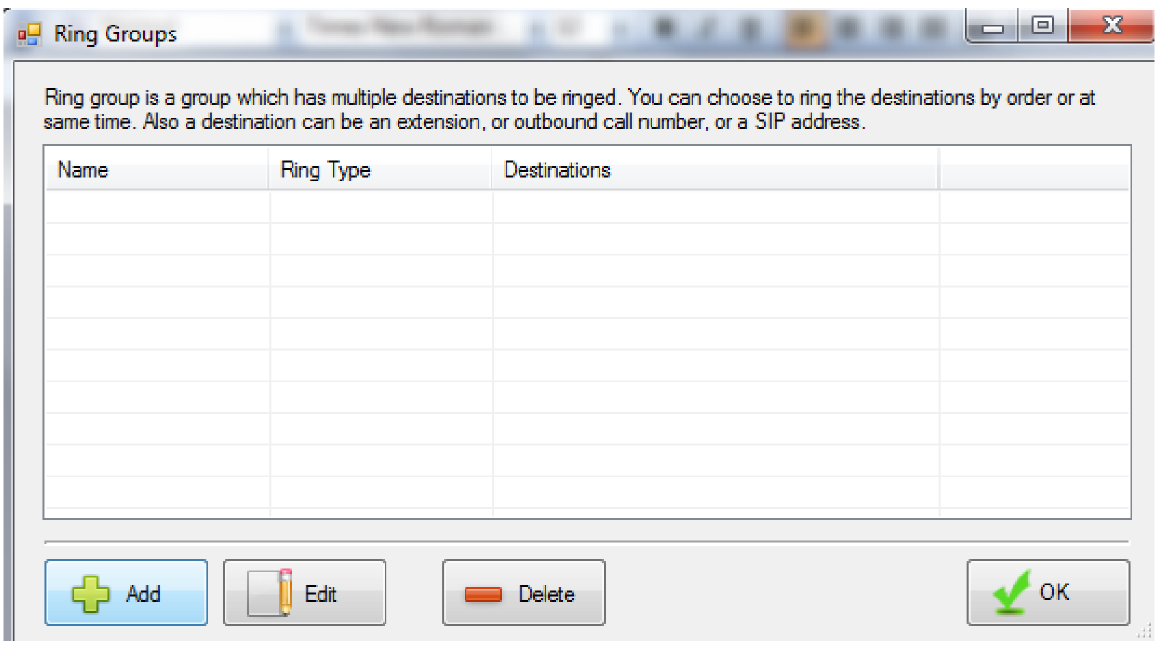
We set 91234567 here, which means using outbound plan 9*. When calls go to this extension, PBX will try to reach outside number 1234567.

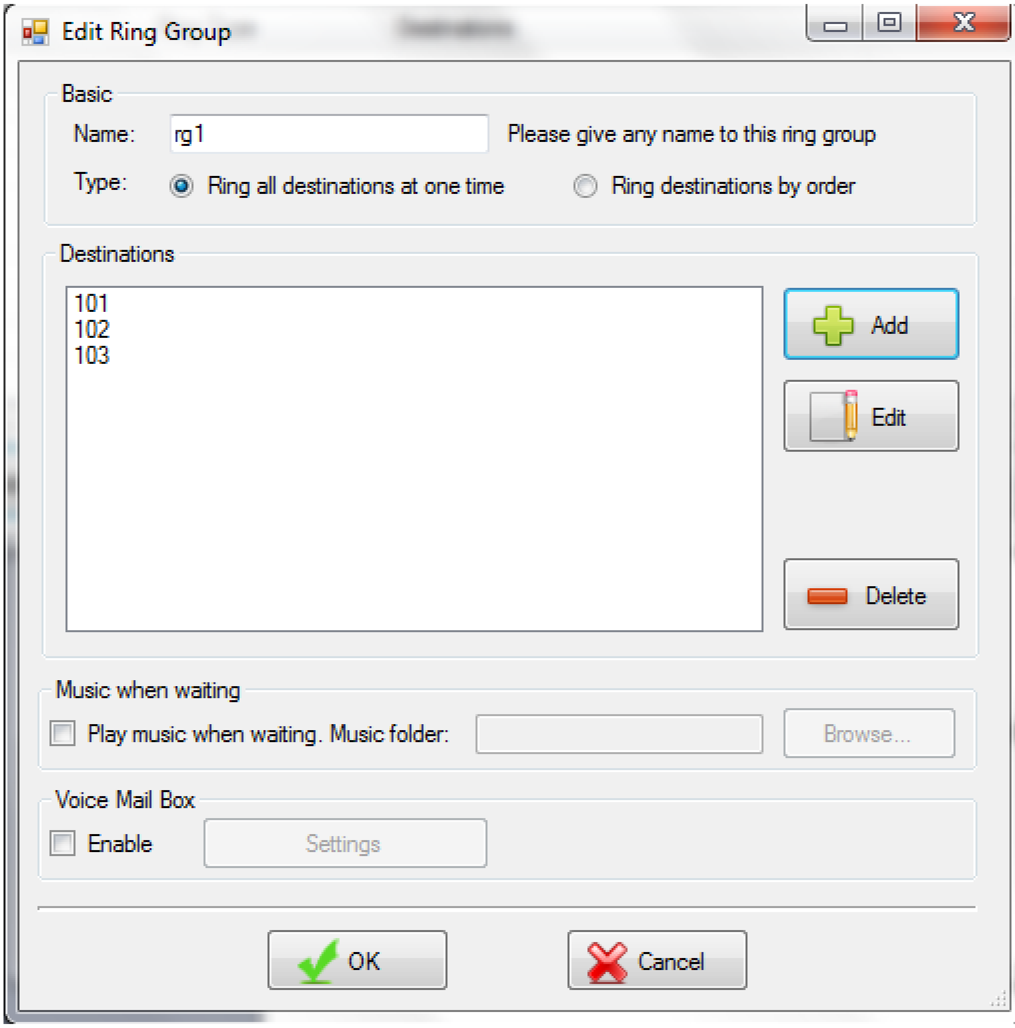
3.7 Ring group

Ring group is a group of extensions or agents that can be ringed(called) by order or same time. Ring group doesn't work like ACD. ACD holds calls until extensions or agents are available to answer the call. Ring group doesn't really hold the calls for long time. It will try to ring the destinations, and the first destination which answered call will be connected to the caller.

Set up a ring group first. Assume its name is rg1.


PCBest Networks SIP PBX v3.12 (Admin Tool)
File
Options
Manage
Server
Operations
SIP Accounts(External Lines)
Extensions
Hunt Groups(ACD)
Parking Slots
Ring Groups
IVR Menus
Conference Rooms
Dial Plans

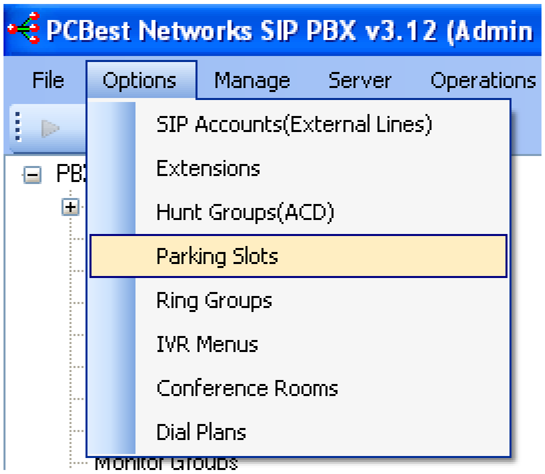

Ring Groups
Ring group is a group which has multiple destinations to be ringed. You can choose to ring the destinations by order or at same time. Also a destination can be an extension, or outbound call number, or a SIP address.
Name
Ring Type
Destinations
Add
Edit
Delete
OK

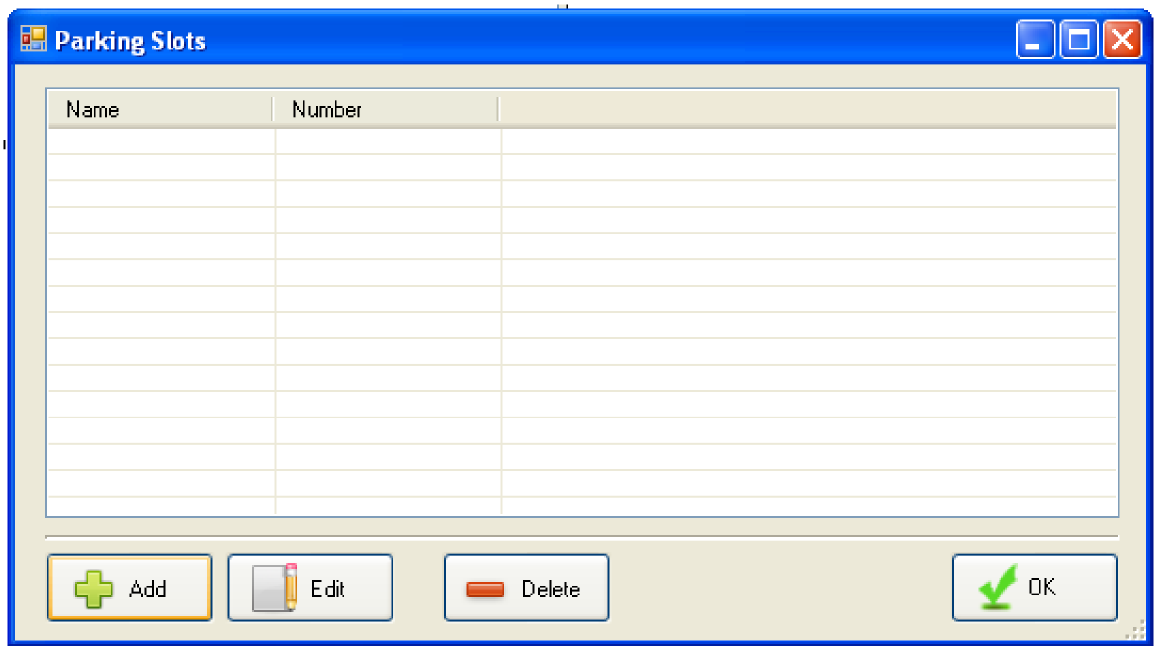


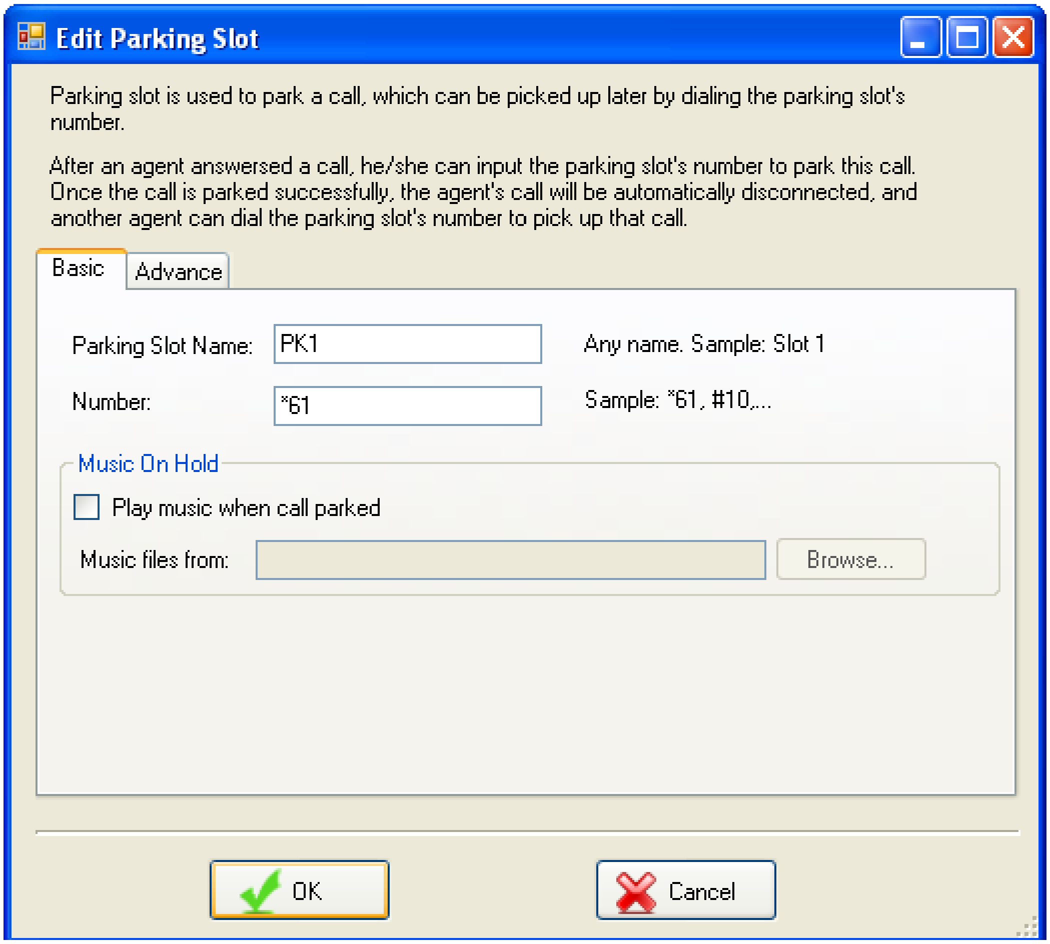
Three extensions 101, 102, 103 are added into ring group rg1. Then we can set up an inbound dialplan, to forward calls to this ring group. When a call comes in and reach this ring group, pbx will ring extensions 101, 102, 103 at same time.

3.8 Call Parking

Call Parking is used to park a call. You must define a call parking slot first to allow the call to park, then later the call can be picked up by another extension or agent.


PCBest Networks SIP PBX v3.12 (Admin
File
Options
Manage
Server
Operations
SIP Accounts(External Lines)
Extensions
Hunt Groups(ACD)
Parking Slots
Ring Groups
IVR Menus
Conference Rooms
Dial Plans
Monitor Groups


Parking Slots
Name
Number
Add
Edit
Delete
OK



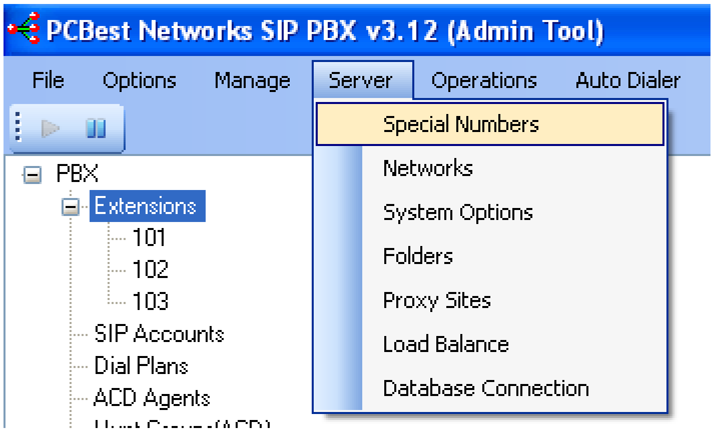
After defined a Parking Slot “PK1”, you can try an incoming call which is transferred into an extension or agent. When extension pressed *61, the call should be parked. Another extension should be able to pick up this call by dialing *61 into PBX.

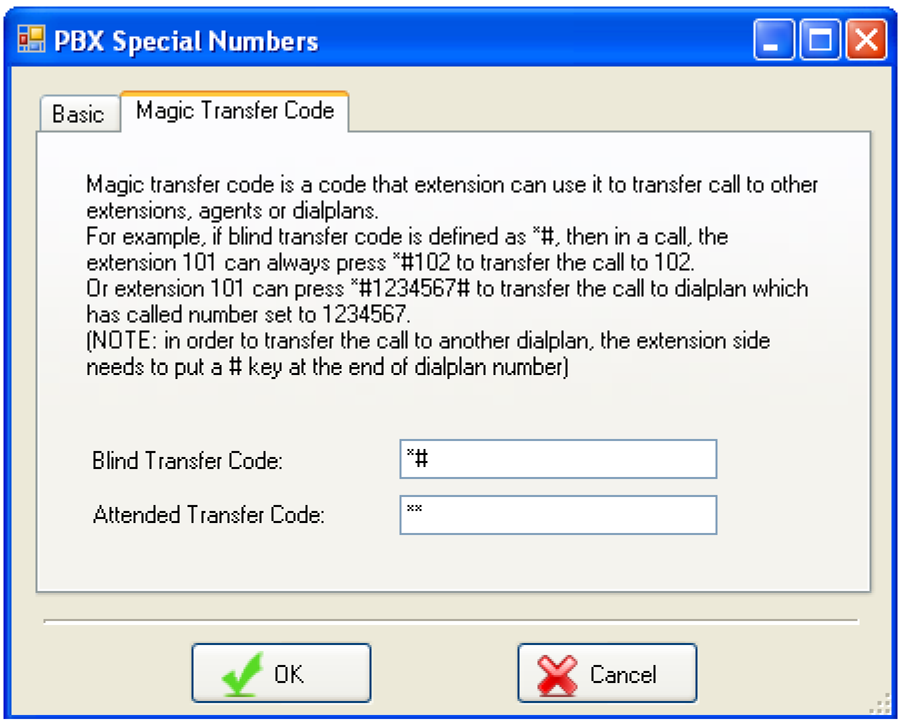
3.9 Magic Transferring Code (ONLY V3)

Magic Transferring Code is used by extensions to transfer current calls to another extension. There are two kinds of transferring:

1. Blind Transfer
2. Attended Transfer

You don’t need to define anything. Magic transferring code default works. Blind transfer code is defined as *#, and Attended transfer code is defined as **.

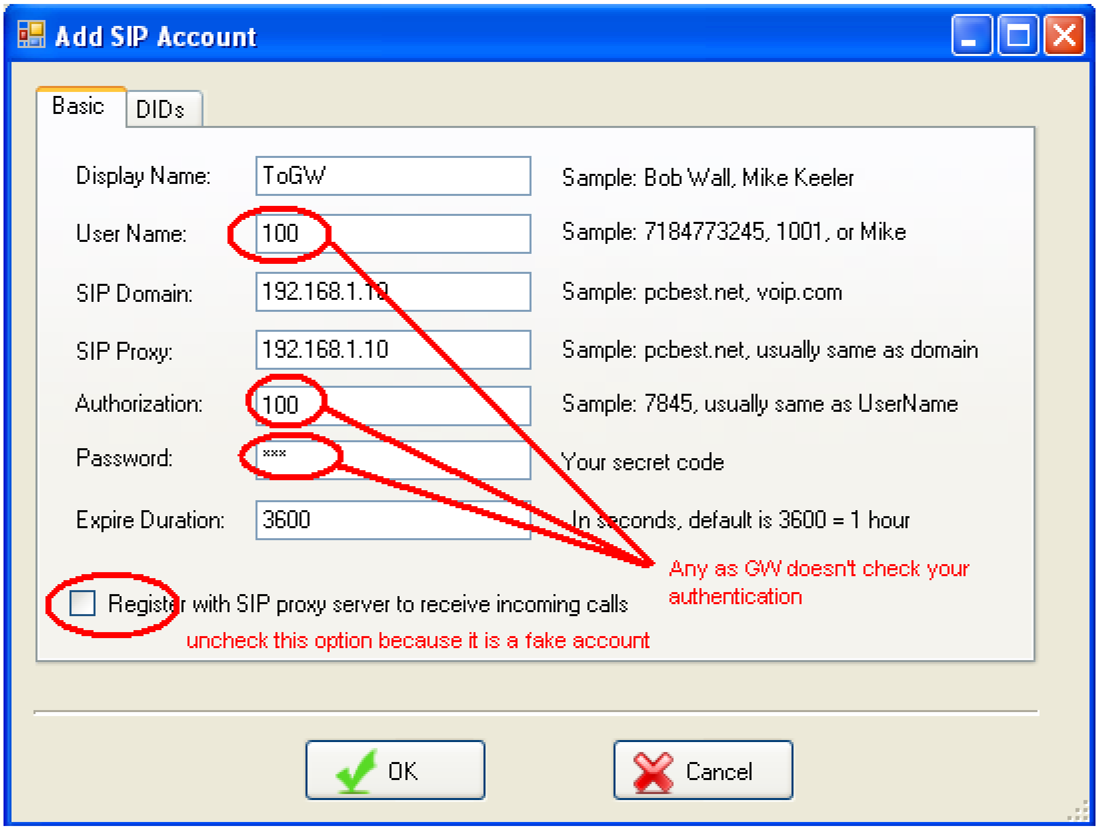




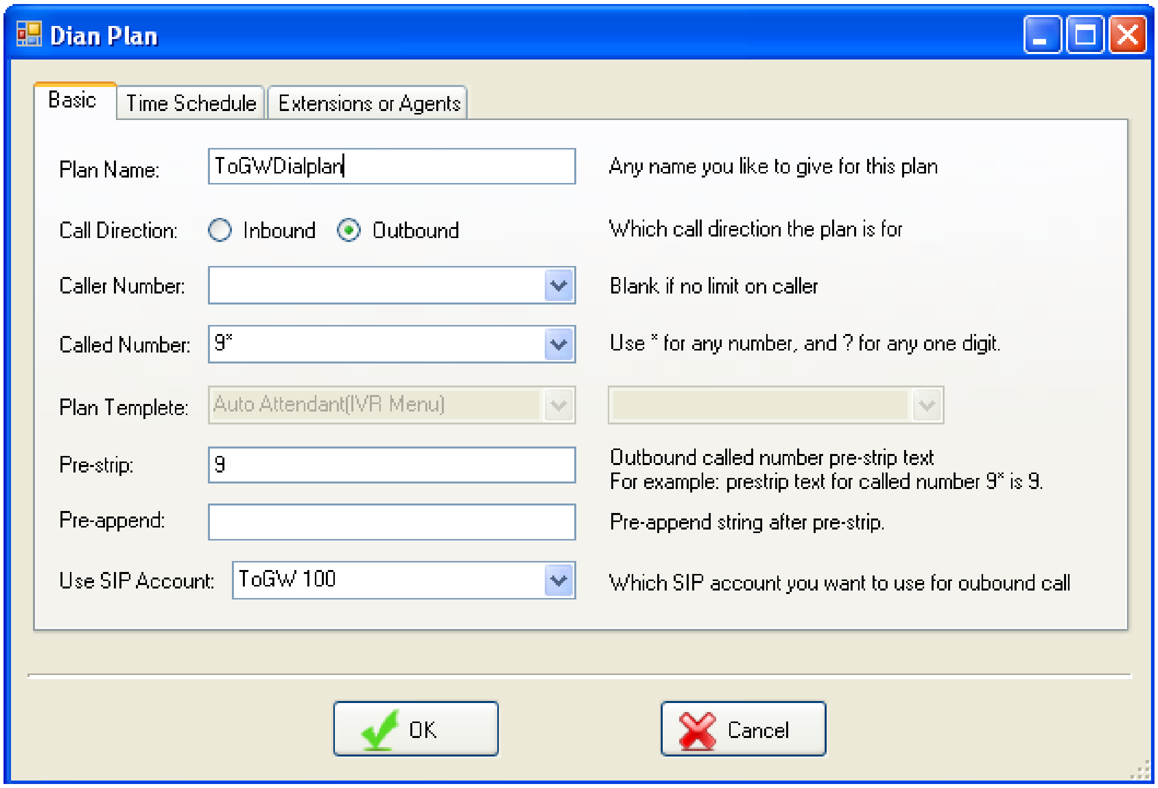
3.10 FXO/FXS or Digital Gateway

PCBest SIP PBX works with most standard FXO/FXS or Digital Gateways. You can configure gateway works as a peer of PCBest SIP PBX.

Assume gateway works at 192.168.1.10, and PCBest SIP PBX runs at 192.168.1.20. On the gateway, you need to forward the incoming calls into IP address 192.168.1.20, and on the PCBest SIP PBX, you need to set up a fake SIP account that points to gateway's IP address:

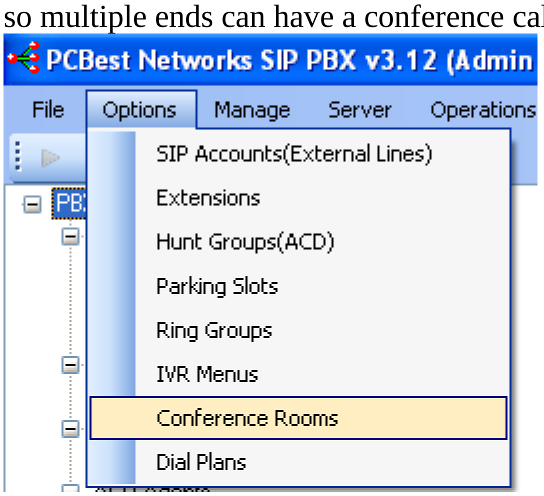


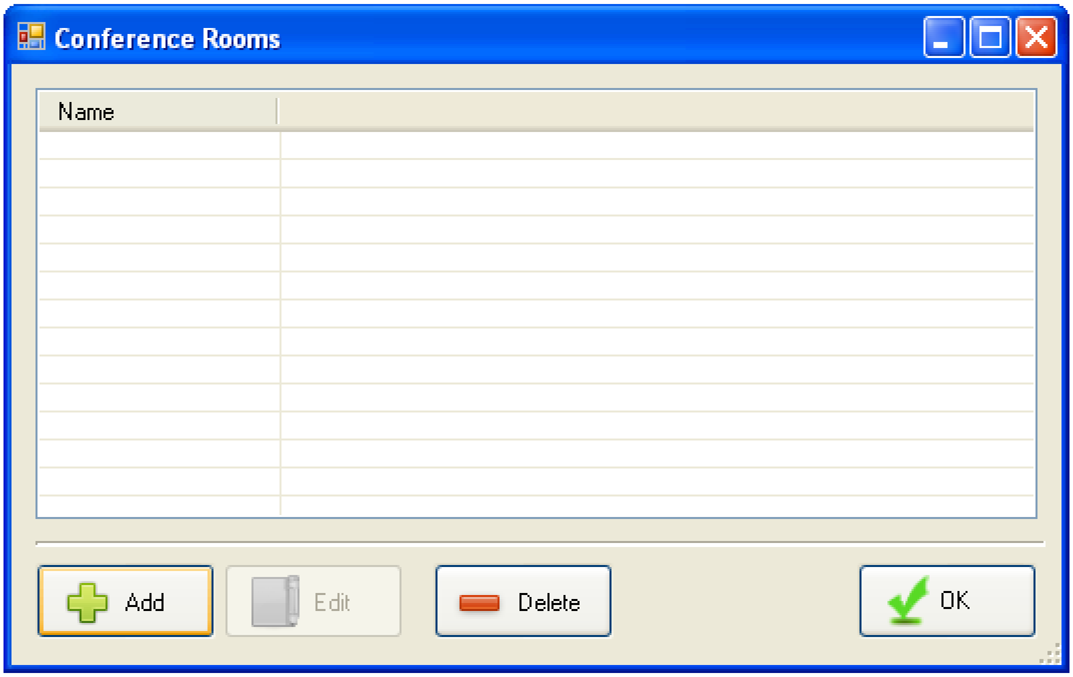
By doing this, you setup a peer which is connected to your gateway. Next step, you need to setup an outbound dialplan to use this sip account to forward extension calls into gateway.



3.11 Conference Room

You can define a conference room, then forward multiple calls into one conference room, so multiple ends can have a conference call.



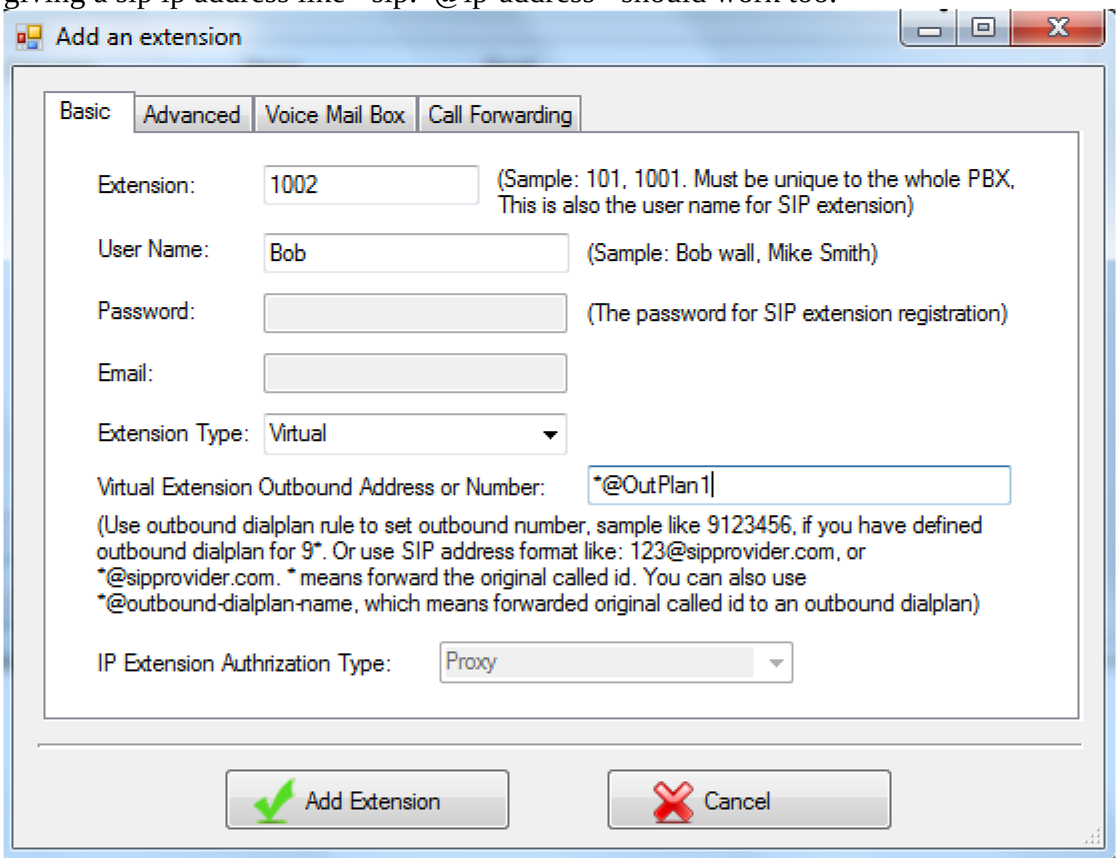


Then you can define a dialplan to forward incoming calls into this conference room.

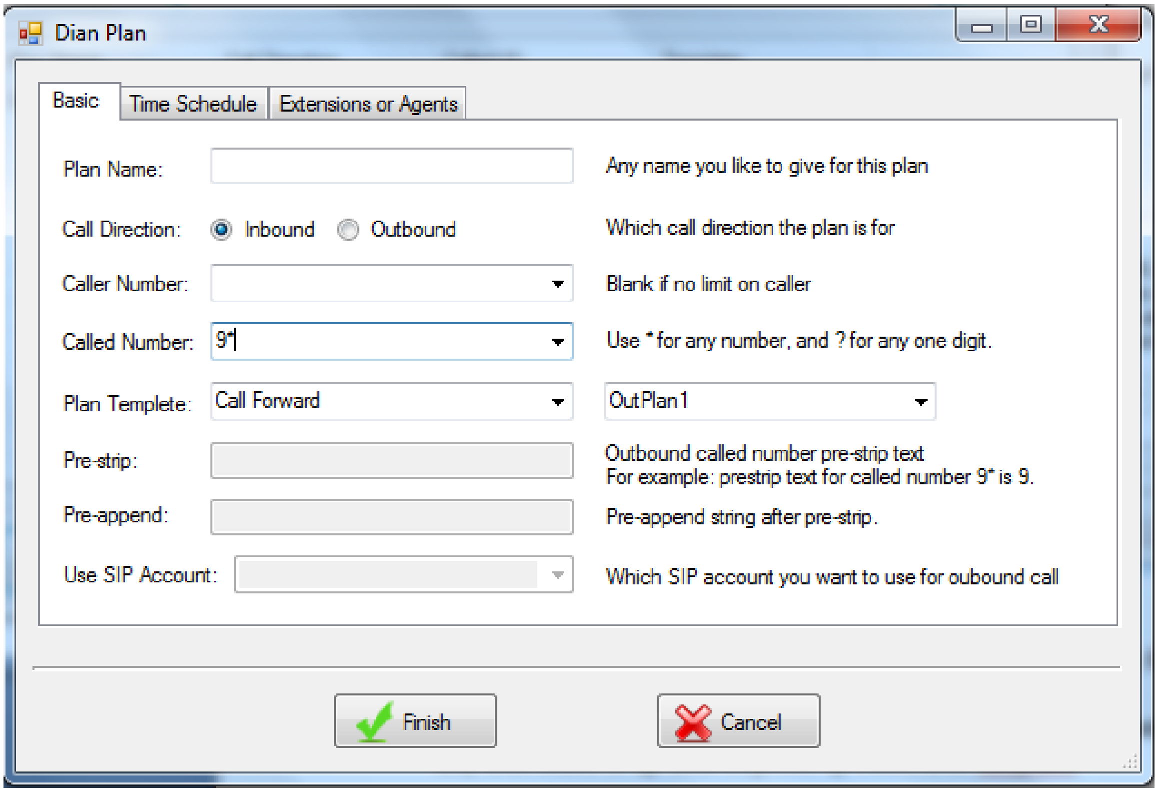
3.12 Inbound 2 Outbound

Sometimes you need to convert an inbound call to outbound call directly.
Because only extensions can call outbound dialplan, so you can achieve this by two ways:
1. Create a virtual extension. In the virtual extension destination address, you can input *, means directly inbound call(dialplan)'s called id to find out proper dialplan. You can give *@outbound-dialplan-name to specify using which dialplan. You can also give sip

address like <sip:*@sipaccount-domian> to route call out by specific sip account. More, giving a sip ip address like <sip:*@ip-address> should work too.

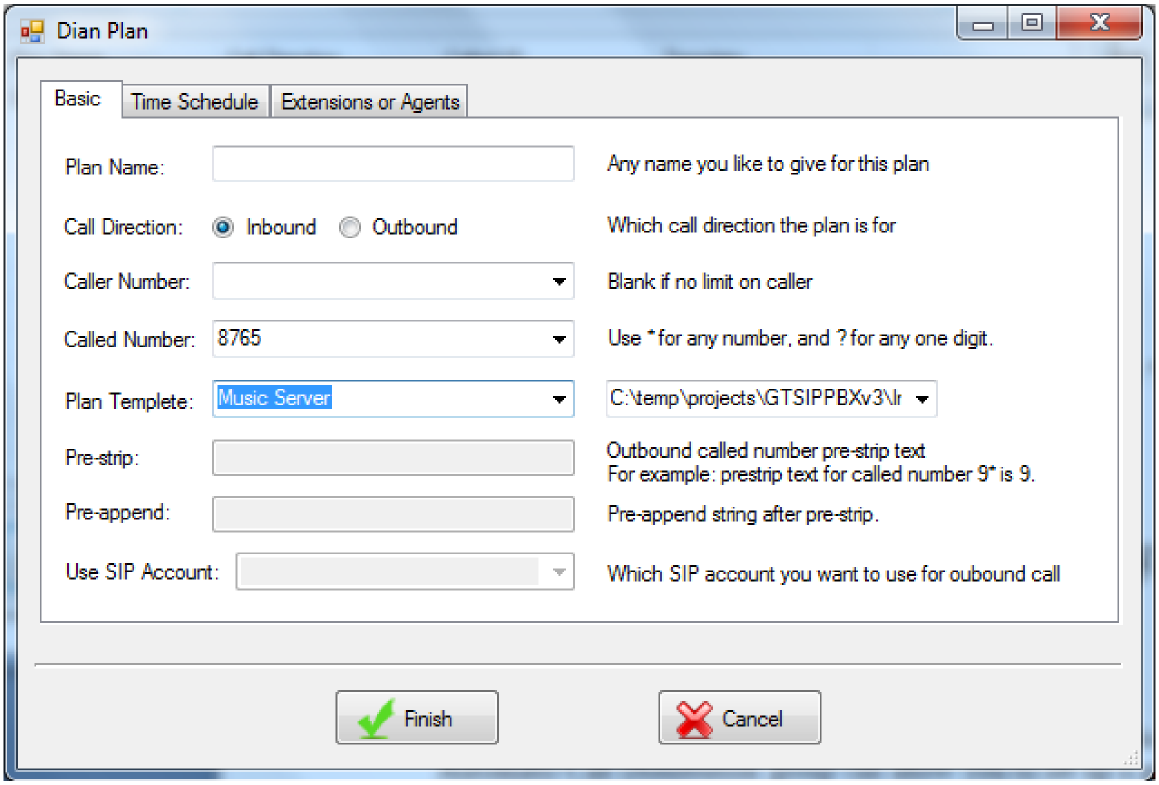


2. Use call forward inbound dialplan
Create an inbound dialplan, set call template to call forward, then choose an outbound dialplan for call forwarding.
Note, for this call forwarding inbound dialplan, please adjust its order in the dialplan list, and make it up and be front of outbound dialplan.



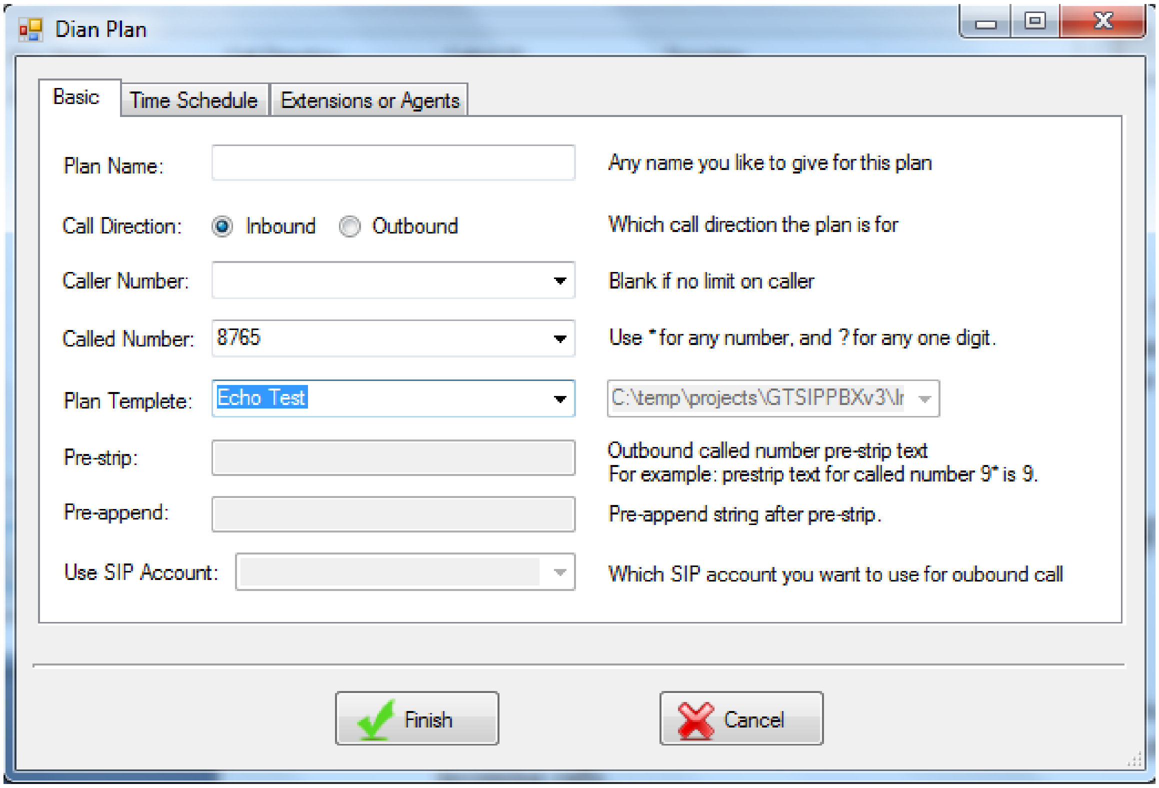
3.13 Setup a music server

Create an inbound dialplan, and choose call plan template to "Music Server", then give the name of music file folder.



3.14 Echo Test for IP extension

Create an inbound dialplan, and choose call plan template to "Echo Test".
IP extensions can call this inbound dialplan to see if voice can be returned back in time.
Sometimes we use this feature to detect network problem like one-way audio.

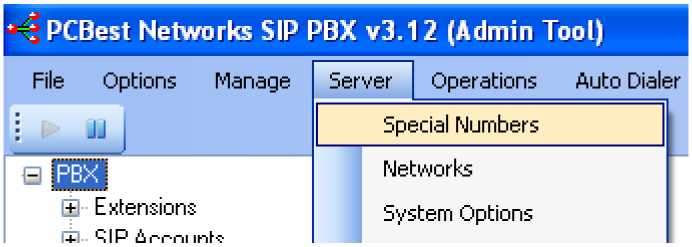


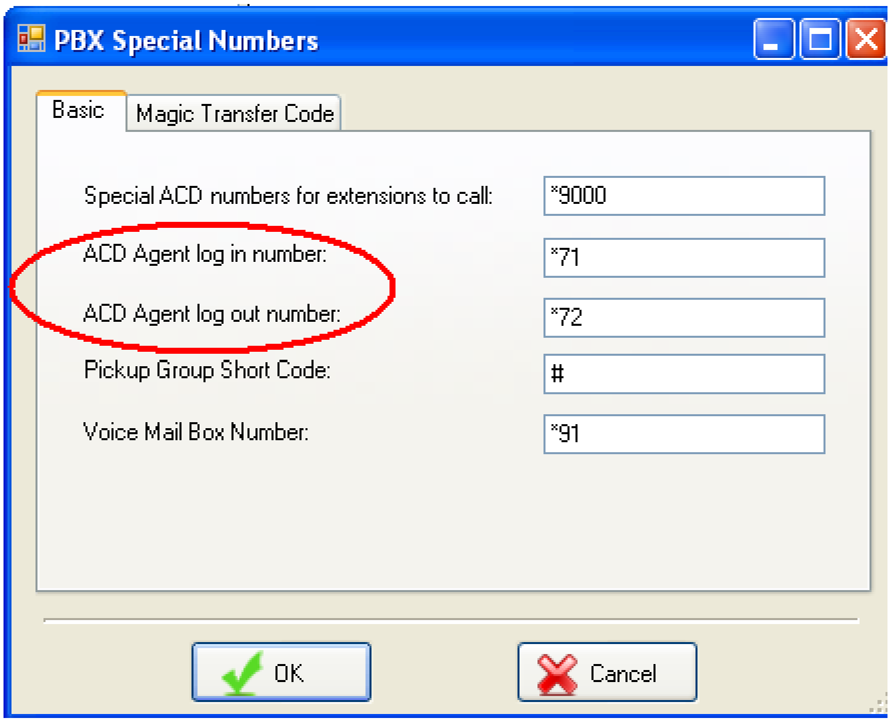
4 PBX Advanced Call Center Features

PCBest SIP PBX can be used as a call center environment. As described in 3.3, Automatic Call Distribution group can allow you to set up a group of agents to answer incoming calls.

4.1 Setting up ACD agents

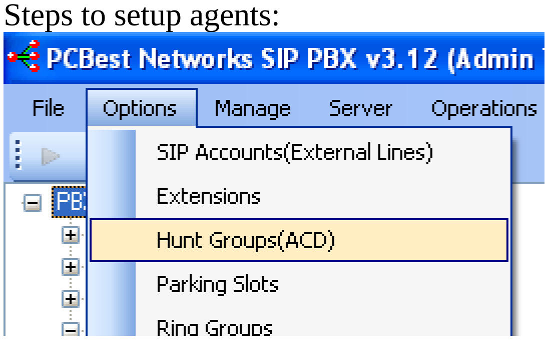
What is an agent? An agent is **NOT** an extension.
An extension is a physical phone, but an agent is a real person to work on an extension. So there may be more than one agent working on the same extension. Usually in a call centre environment, an agent will start to work by login at one of the extension. PBX defines special phone numbers for agents to login and logout at extensions.

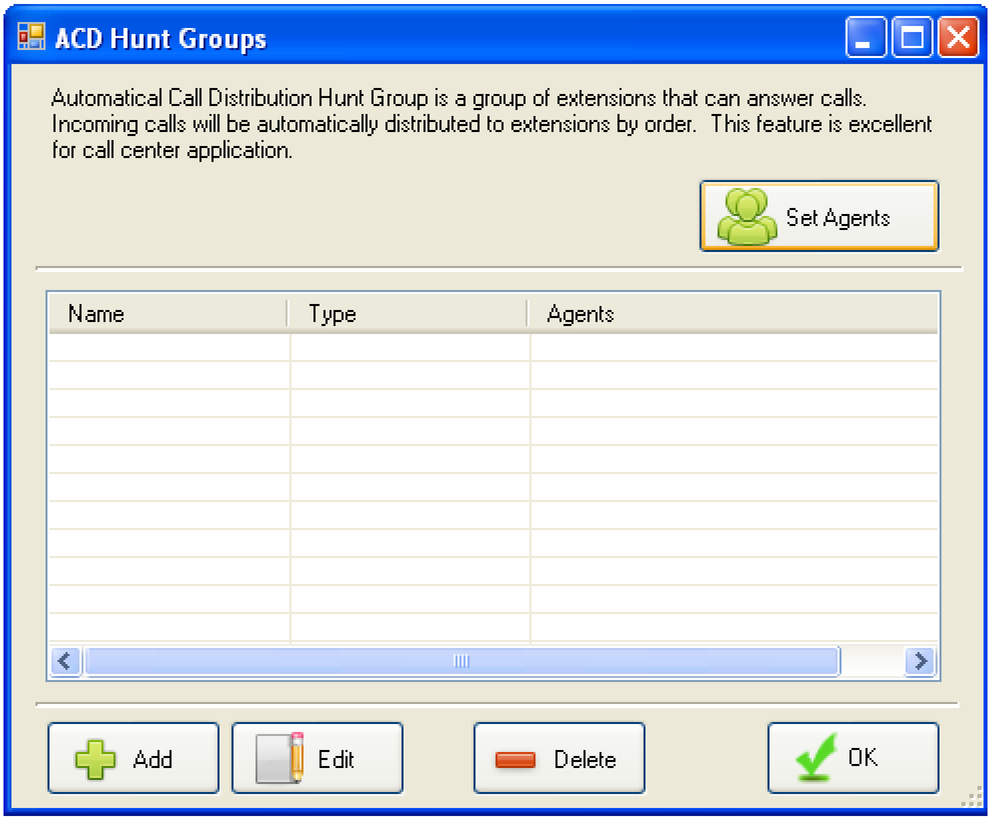


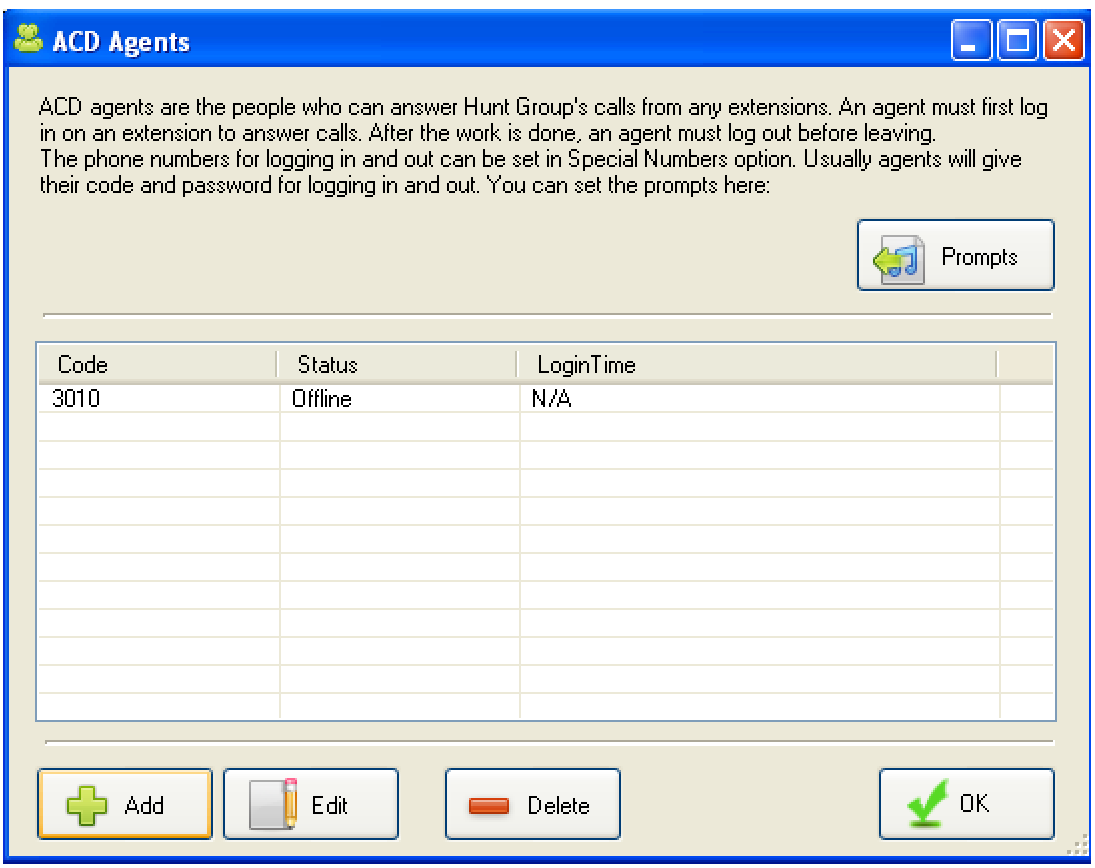


Agents can call above special login and logout numbers from any extension to indicate they are at that extension or not.

Steps to setup agents:



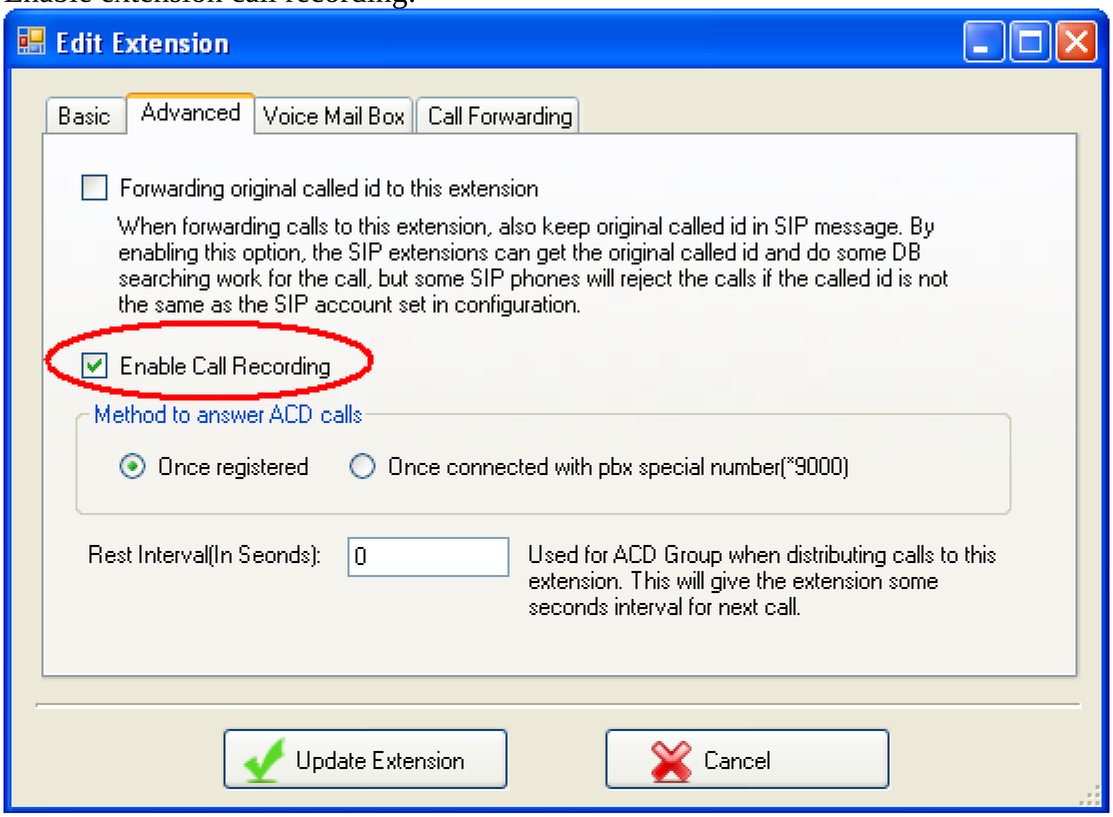

ACD Hunt Groups
Automatical Call Distribution Hunt Group is a group of extensions that can answer calls. Incoming calls will be automatically distributed to extensions by order. This feature is excellent for call center application.
Set Agents
Name
Type
Agents
Add
Edit
Delete
OK


ACD Agents
ACD agents are the people who can answer Hunt Group's calls from any extensions. An agent must first log in on an extension to answer calls. After the work is done, an agent must log out before leaving.
The phone numbers for logging in and out can be set in Special Numbers option. Usually agents will give their code and password for logging in and out. You can set the prompts here:
Prompts
Code
Status
LoginTime
3010
Offline
N/A
Add
Edit
Delete
OK

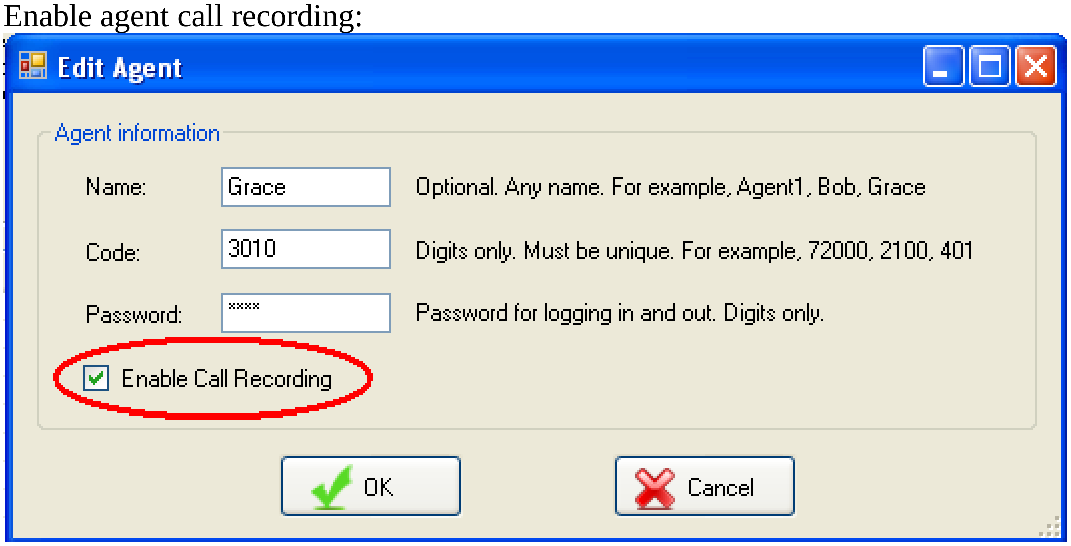
4.2 Enabling Call Recording

Also PCBest SIP PBX allows you to record every calls by enabling recording feature for extensions or agents.

Enable extension call recording:



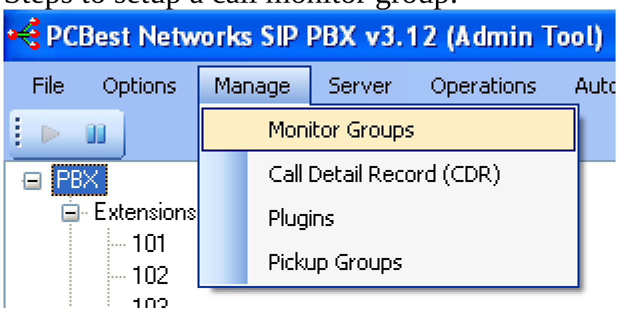
Enable agent call recording:

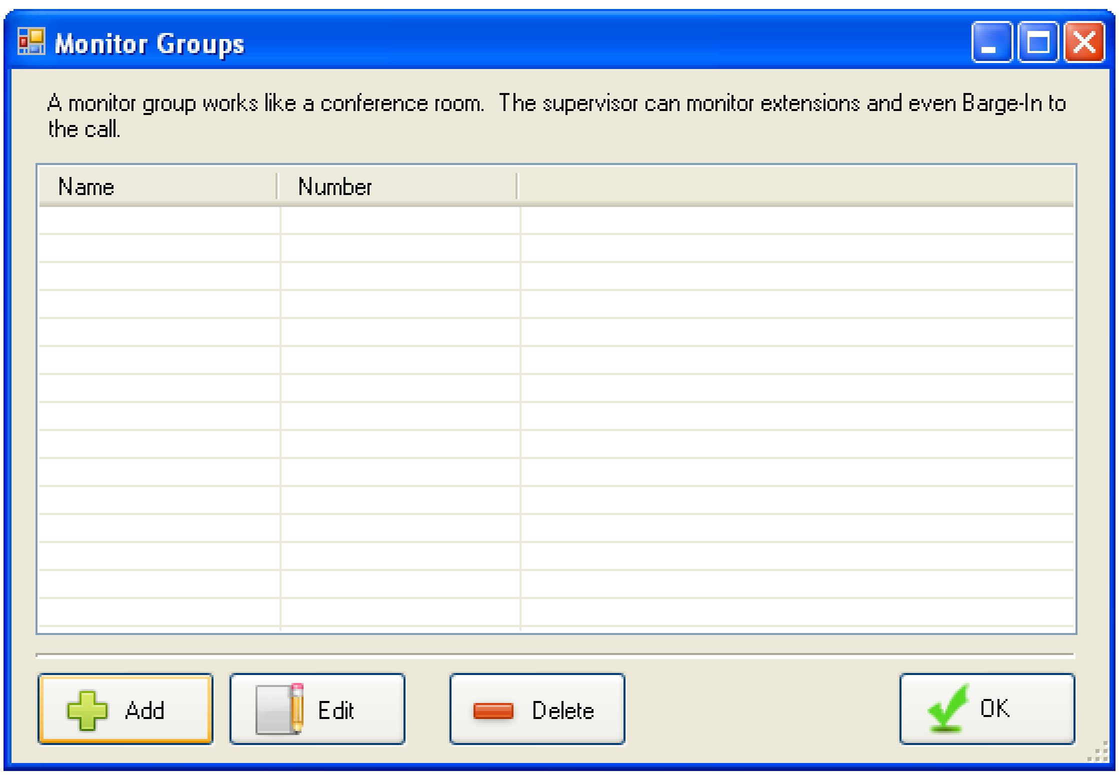


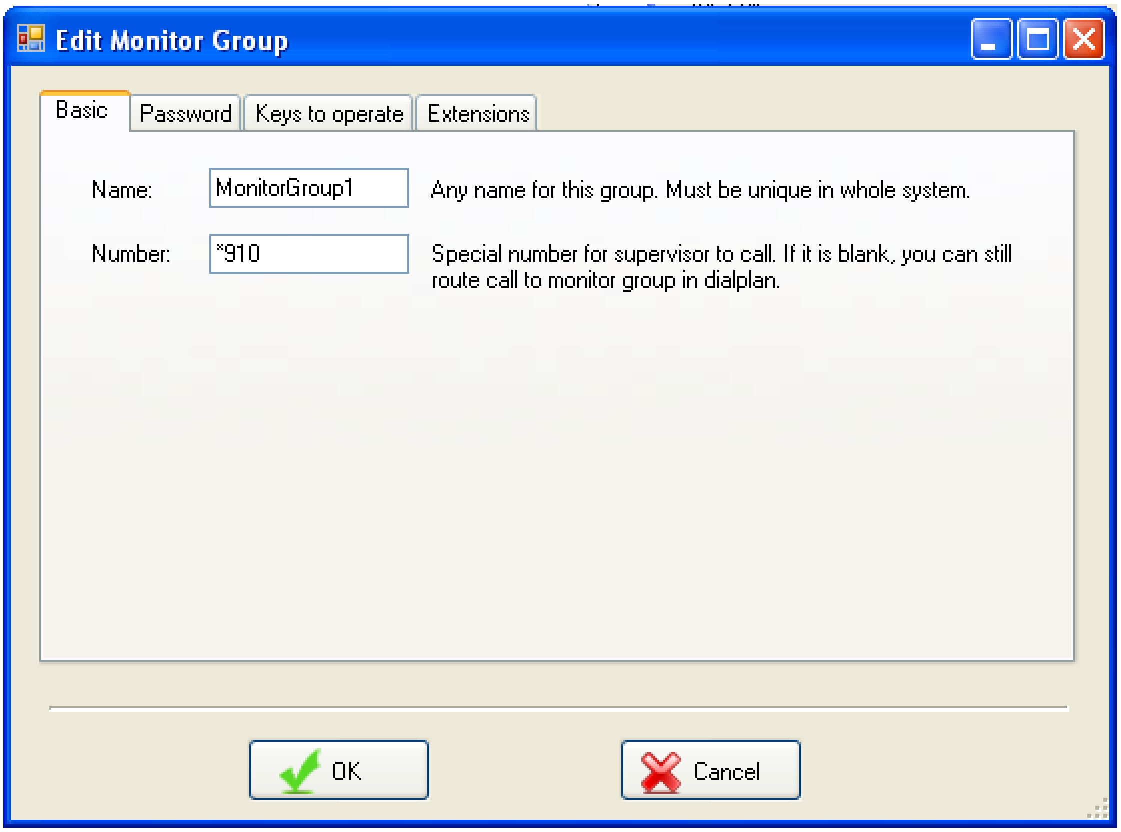
4.3 Supervisor Call Monitoring

In a typical call centre environment, supervisor needs to monitor agent's call in real time. Sometimes supervisor even can give assistance to agent about how to answer the client's call, or even join into the conversation. In order to achieve the call monitoring, you need to setup a call monitoring group. You can regard a call monitoring group as a conference room, so supervisor, agent and client can all join into.

Steps to setup a call monitor group:



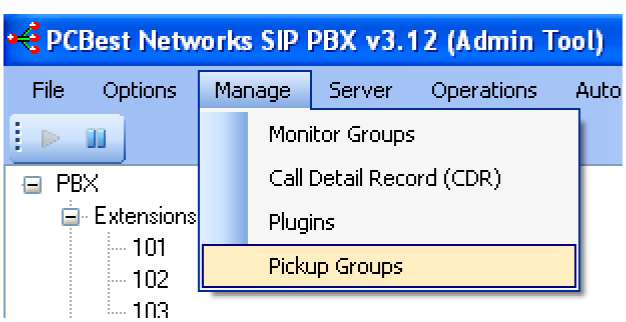


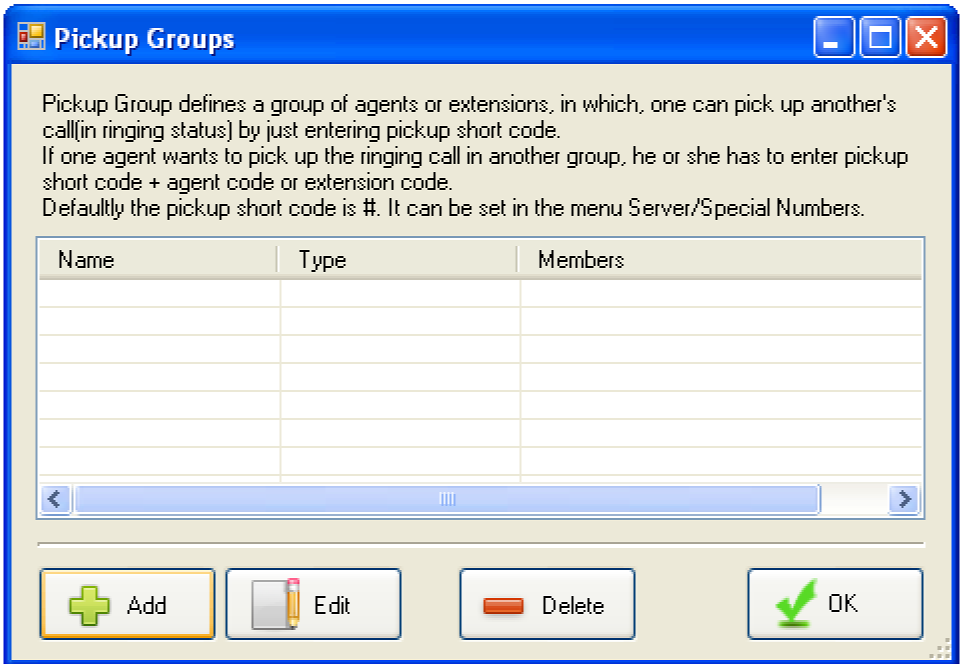


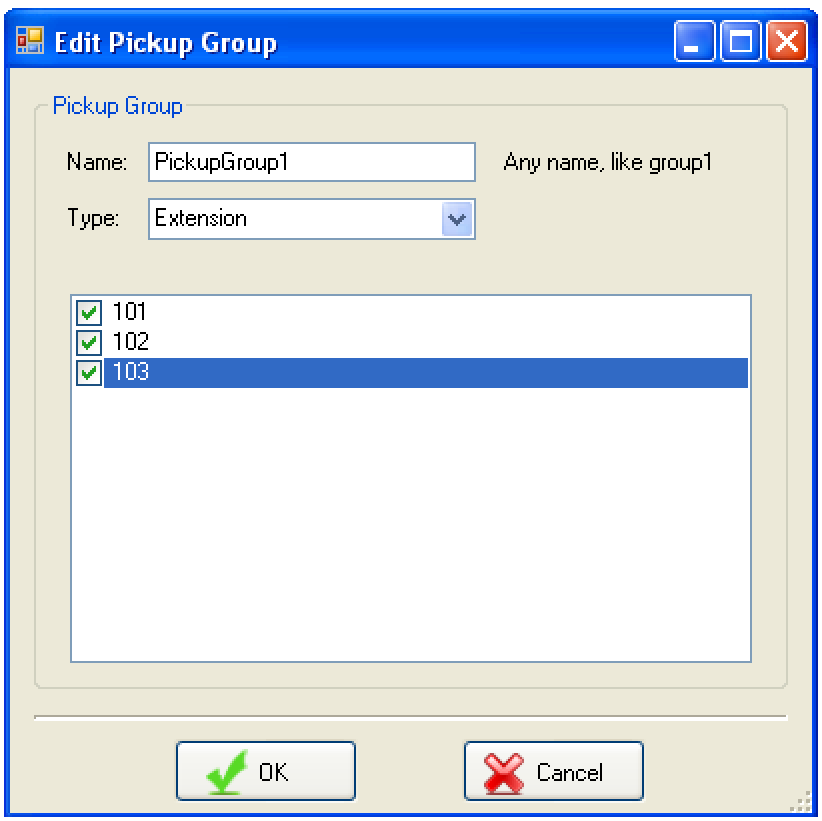
Once you defined a monitor group, please call monitor group number *910 from an supervisor type extension, you will be able to follow the IVR menu to monitor any other extensions.

4.4 Pickup Group

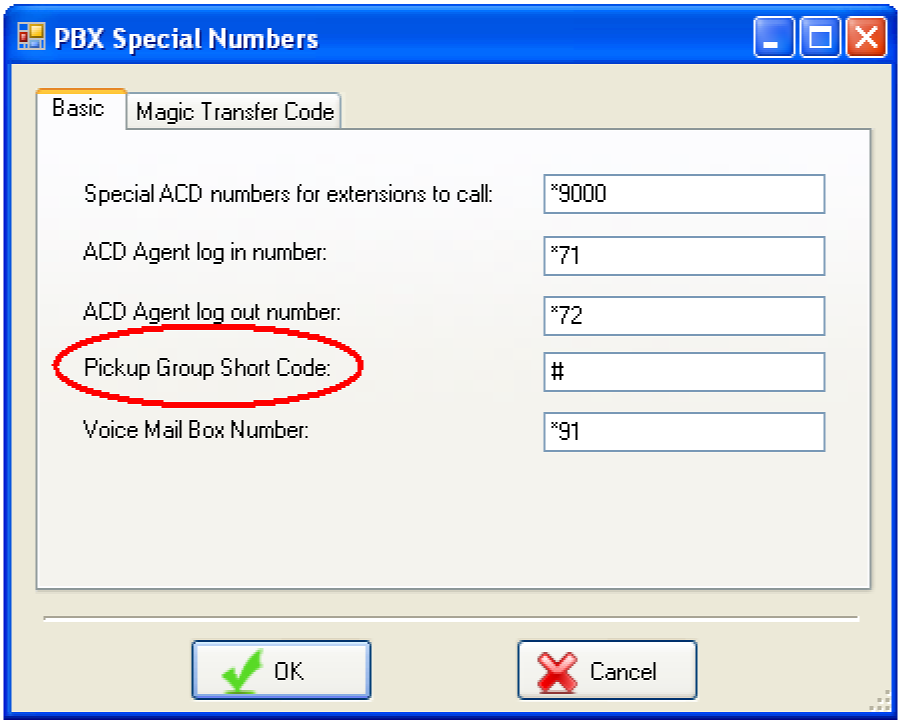
Pickup Group defines a group of agents or extensions, in which, one can pick up another's call(in ringing status) by just entering pickup short code.
If one agent wants to pick up the ringing call in another group, he or she has to enter pickup short code + agent code or extension code.
Defaultly the pickup short code is #. It can be set in the menu Server/Special Numbers.







Pickup Group Short Code is defined in special number:



5 PBX Auto Dialer Feature (Pro Only)

PCBest SIP PBX can do automatic outbound calls, and forward connected calls to an inbound dialplan. Auto Dialer Tasks are outbound jobs from database. You can use it to make outbound calls, then do special routes for connected calls. Typical auto dialer tasks can be:

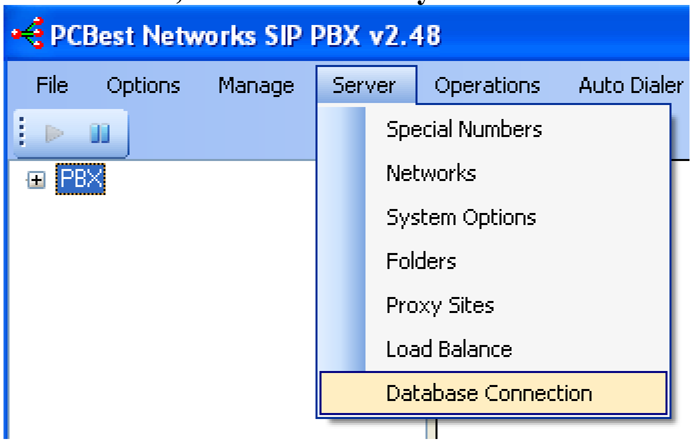
Auto Survey Calls: You can specify an auto dialer task which presents an IVR menu for the connected calls. Once the customer chose an option, then forward the call to another menu, and so on. The customer choices will be record into database like this: IVRMenu1,1;IVRMenu2,2;...

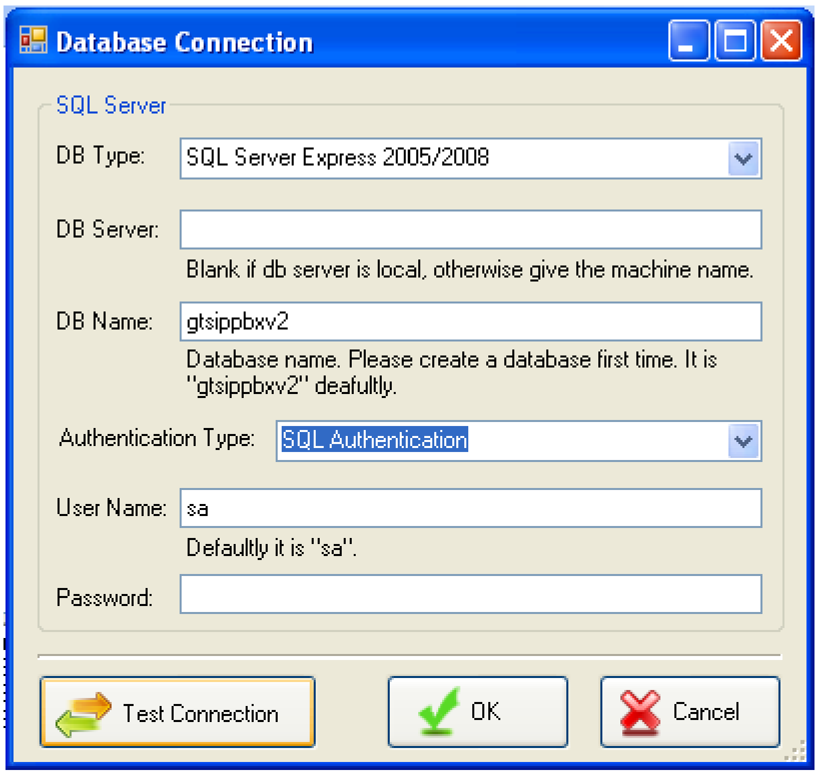
Call Me Back: Your customer can give a phone number to call back on your website. The phone number will be stored into PBX's auto dialer call jobs table. The pbx will call the number, and once the call is connected, then forward the call to an extension(or agent).

CRM, Message Broadcasting, and other applications: Broadcast your messages to a large of phone numbers to increase your sale.

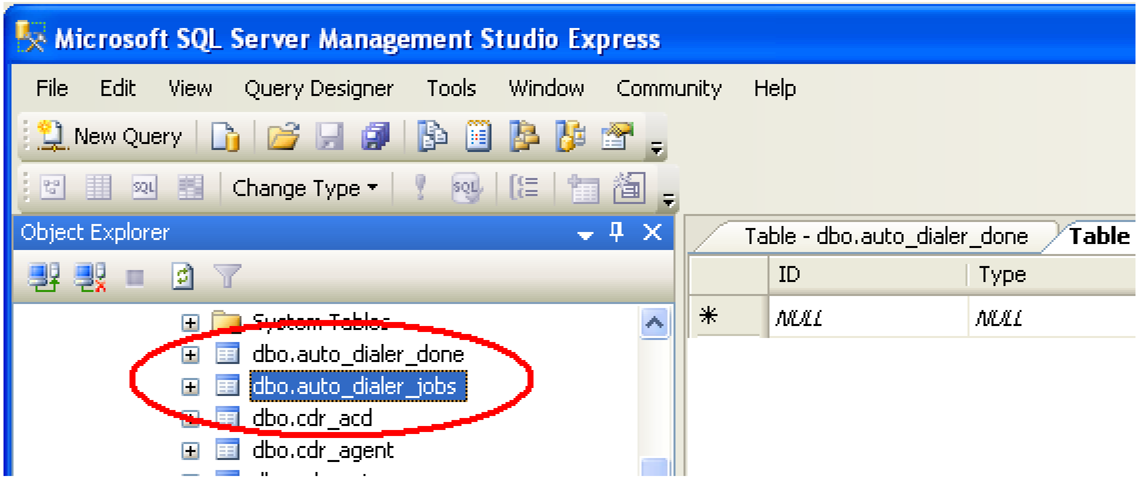
How does it work?

In order to make this feature works, V2 needs setup a Database Connection. V3 doesn't need, because V3 always works with database.



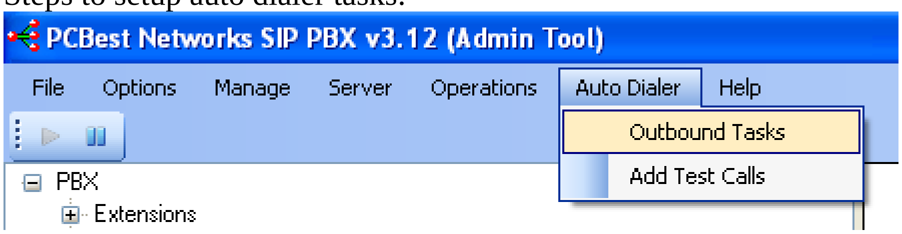


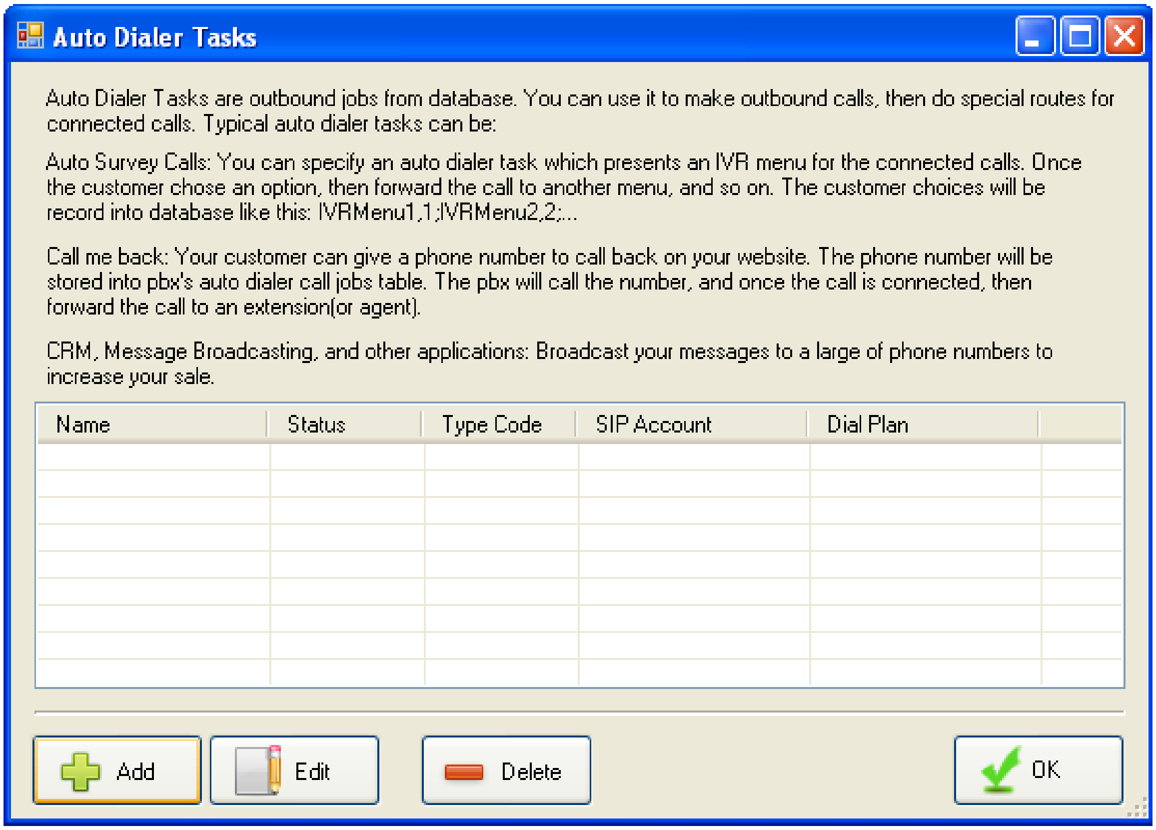
Once the PBX connected with the database, it will create some tables that it needs. Please look at two tables auto_dialer_jobs, and auto_dialer_done.

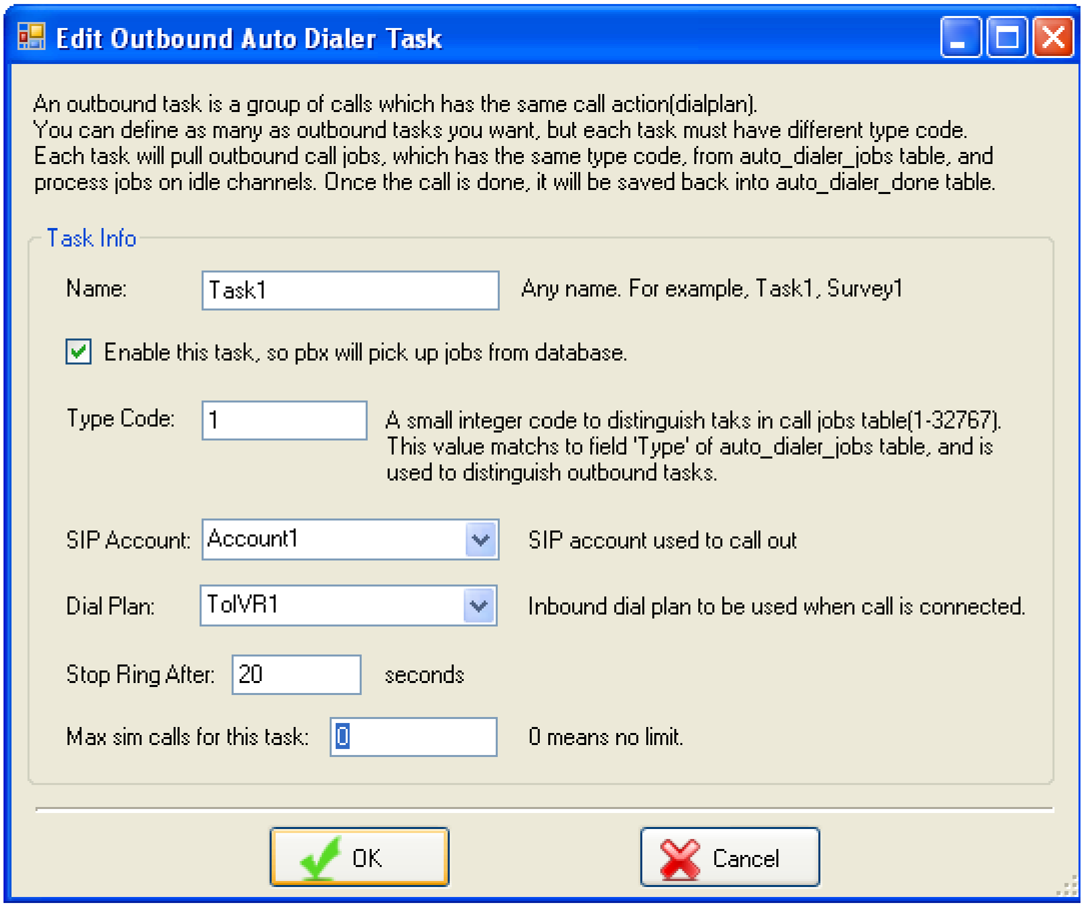


PBX will try to check auto_dialer_jobs every 2 seconds, to pull out outbound records, then dial the numbers out, then write the result back into auto_dialer_done table.

Steps to setup auto dialer tasks:

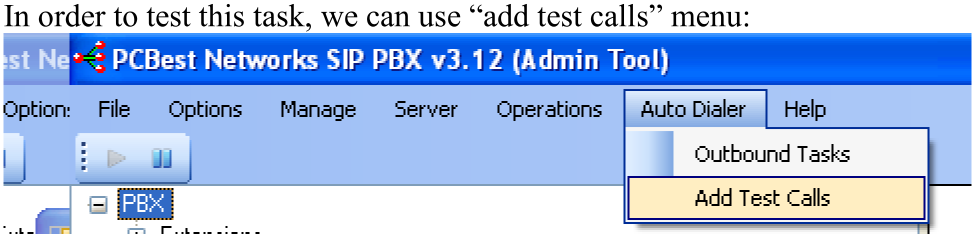


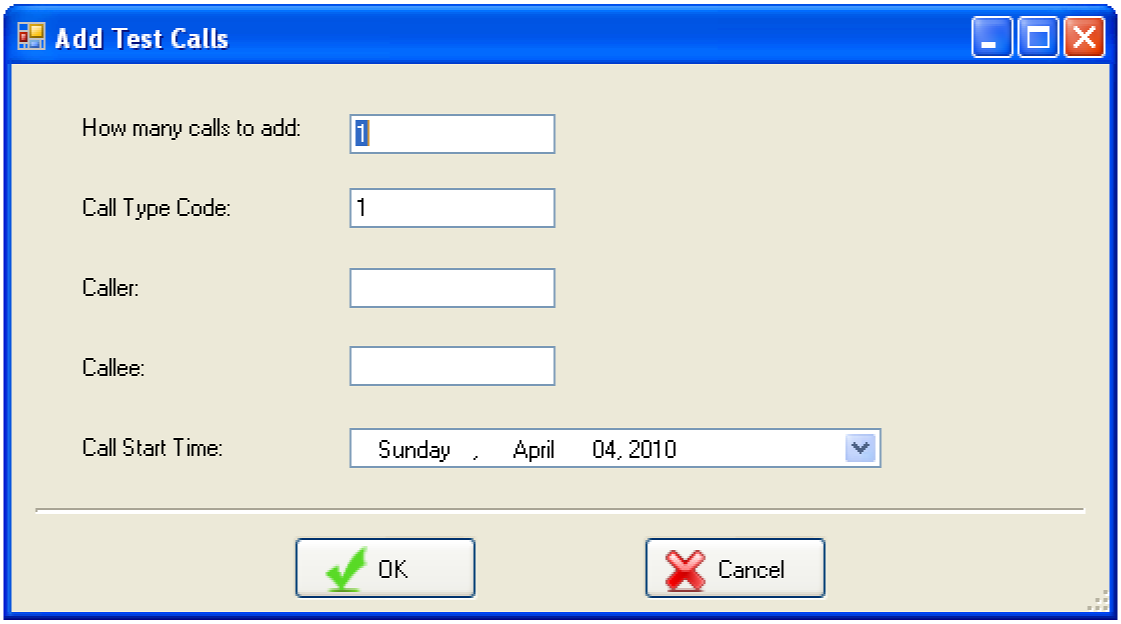

Auto Dialer Tasks
Auto Dialer Tasks are outbound jobs from database. You can use it to make outbound calls, then do special routes for connected calls. Typical auto dialer tasks can be:
Auto Survey Calls: You can specify an auto dialer task which presents an IVR menu for the connected calls. Once the customer chose an option, then forward the call to another menu, and so on. The customer choices will be record into database like this: IVRMenu1,1;IVRMenu2,2;...
Call me back: Your customer can give a phone number to call back on your website. The phone number will be stored into pbx's auto dialer call jobs table. The pbx will call the number, and once the call is connected, then forward the call to an extension(or agent).
CRM, Message Broadcasting, and other applications: Broadcast your messages to a large of phone numbers to increase your sale.
Name
Status
Type Code
SIP Account
Dial Plan
Add
Edit
Delete
OK



Above sample defines auto dialer “Task1”, which has type code 1, and use SIP account “account1” to dial out. After the call is connected, it will use dialplan ToIVR1 to handle the call.

In order to test this task, we can use “add test calls” menu:

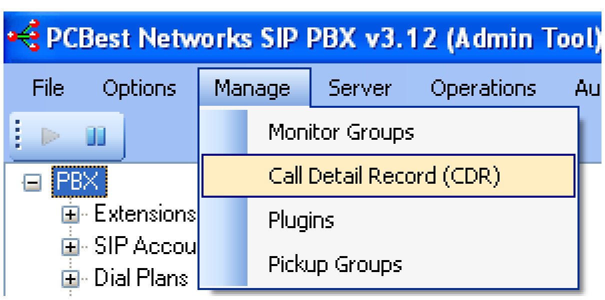




Give the type code 1, and caller and callee numbers, then click OK. PBX should be able to pick up the call job, and dial out to the number.

6 PBX Other Configurations

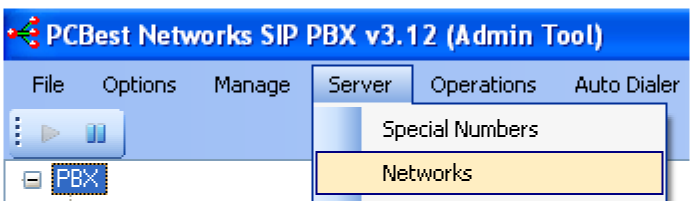
6.1 CDR



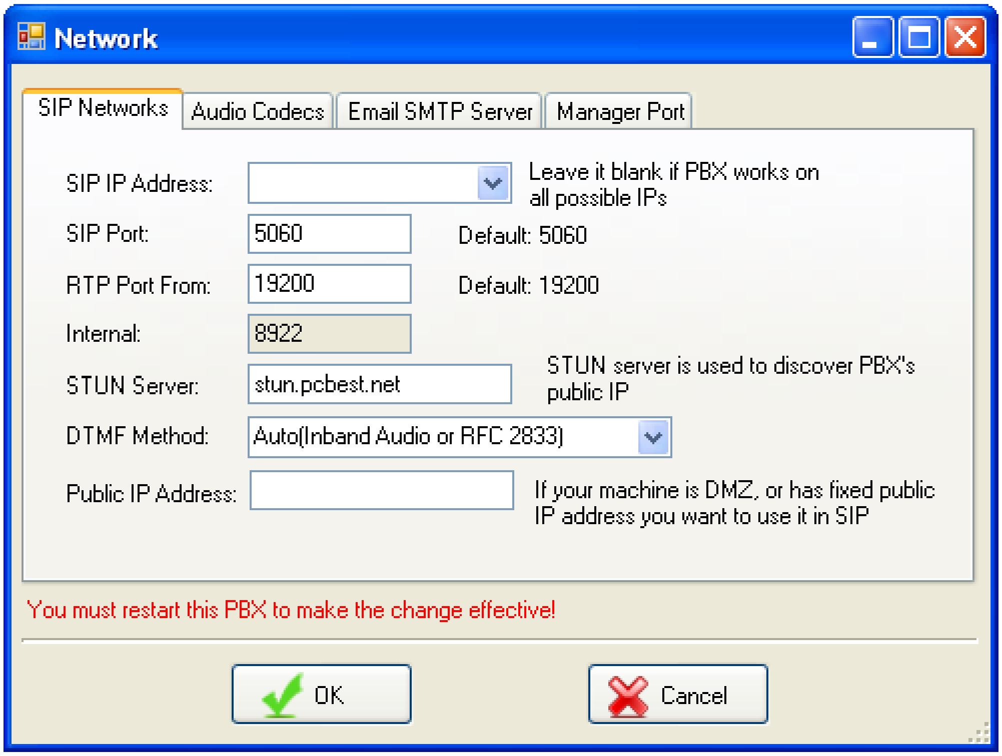
You can write CDR into database: (**Note** V2 must setup a database connection first)


Call Detail Record(CDR)
PBX CDR
Enable
Write to text file
Write to database
ACD CDR
Enable
Write to text file
Write to database
Extension CDR
Enable
Write to text file
Write to database
Agent Login and Logout Record
Enable
Write to text file
Write to database
OK
Cancel

6.2 Networks


PCBest Networks SIP PBX v3.12 (Admin Tool)
File
Options
Manage
Server
Operations
Auto Dialer
Special Numbers
Networks
PBX

SIP Networks Tab:



SIP IP Address: The local IP address that PBX should work on. Usually it is blank, so PBX can work on all possible NIC interfaces or IPs. If you do have multiple IP addresses, and want the PBX only work on one of them, please use drop box to select.

SIP Port: The port number that PBX works on for SIP protocol. Default it is 5060, but you can change it something else. For example, some countries block 5060 to disable VoIP calls. You can use other port number to get around.

RTP Port From: The starting RTP port number. Sometimes you may need to open your firewall for RTP(audio) transmit. Keep in mind, PBX will use a range of RTP port for communication. Basically one channel will use 4 ports(although it only use actually one, but we separate them with enough space), so one 8 channels PBX will need ports open from 19200 to 19232 (19200 + 4*8).

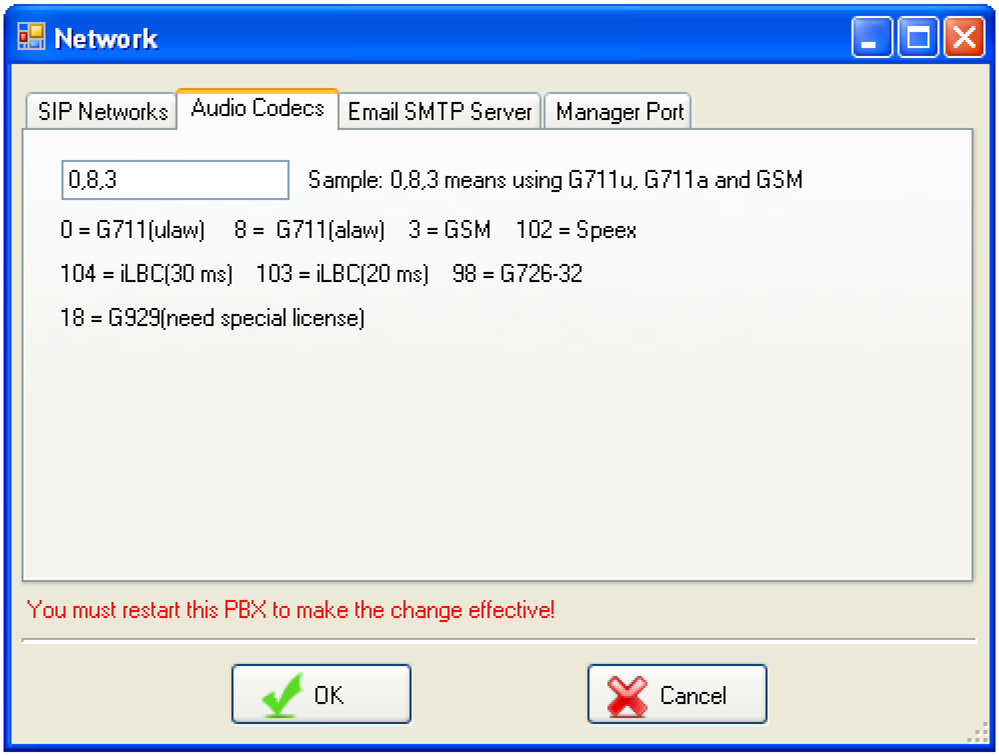
Internal: PBX uses this internal port for internal messages and events communication. It is not changeable.

STUN Server: PBX uses STUN server to discover the actual public IP address of network, to go through possible NAT issue. Please contact your SIP service provider for STUN server setting.

DTMF Method: Usually it is auto, so PBX will automatically figure out the DTMF method. Unless you know the details about this setting, you can change it.

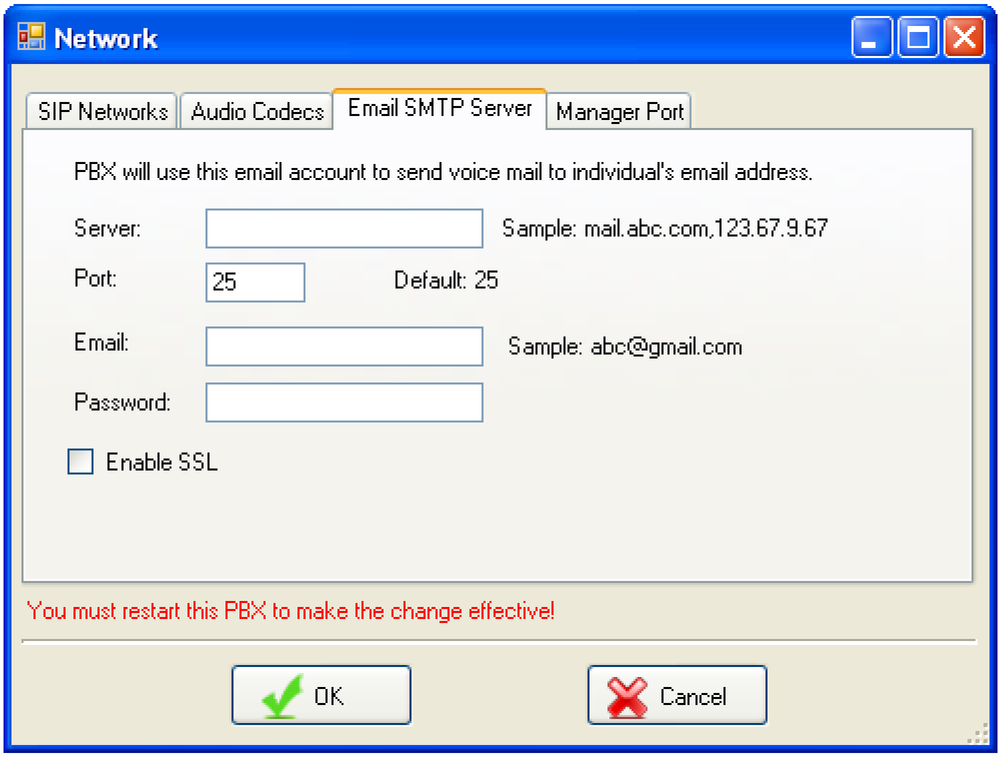
Public IP Address(V3 Only): In some case, for example, DMZ, you know your PBX are working on specific public IP address, so you can specify this field so PBX won't use internal IP address or ignore STUN server to get public IP.

Audio Codec Tab:



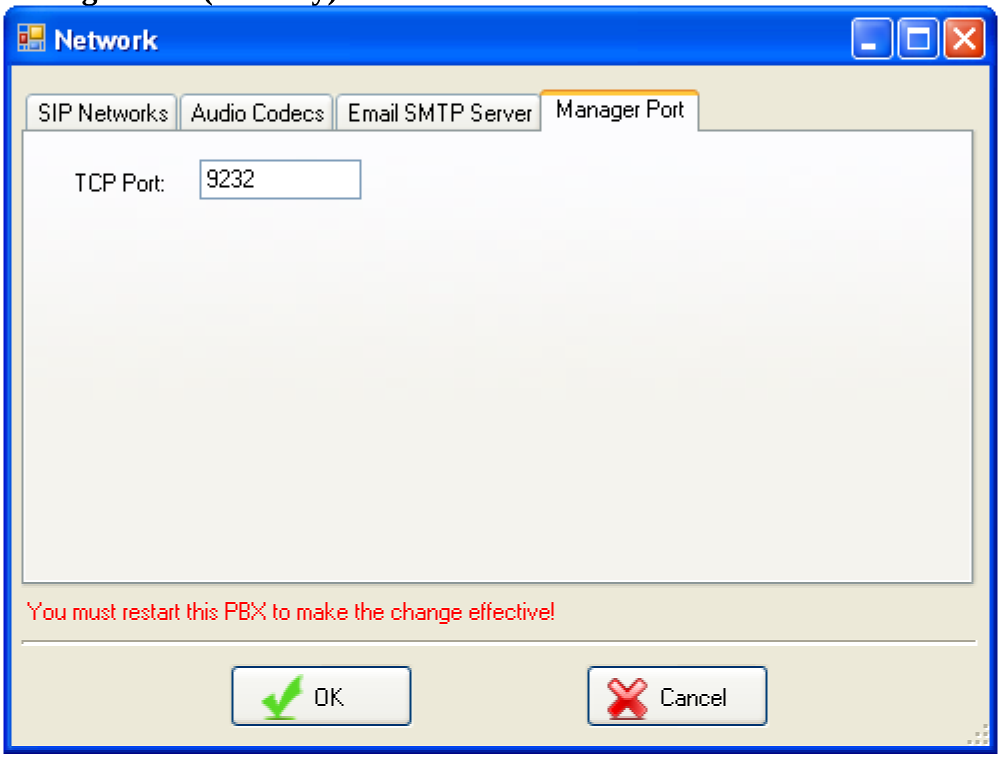
You can specify the PBX which audio codec in SIP SDP negotiation. When negotiating the audio codec, PBX will try to use the audio codec that is in the front of the list.
In above sample, the audio codec is 0,8,3. It means that g711 mulaw first, then g711 alaw, then GSM.

Email SMTP Server:



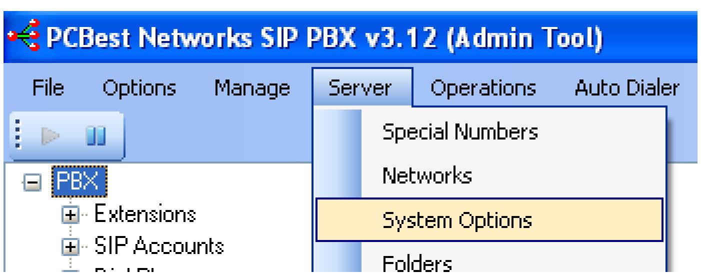
Server: Email server address. It can be an IP address or domain name.
Port: Email server port number.
Email: Email address that is used by PBX to send out email.
Password: Password for above email address.
Enable SSL: if it uses SSL.

Manager Port (V3 Only):

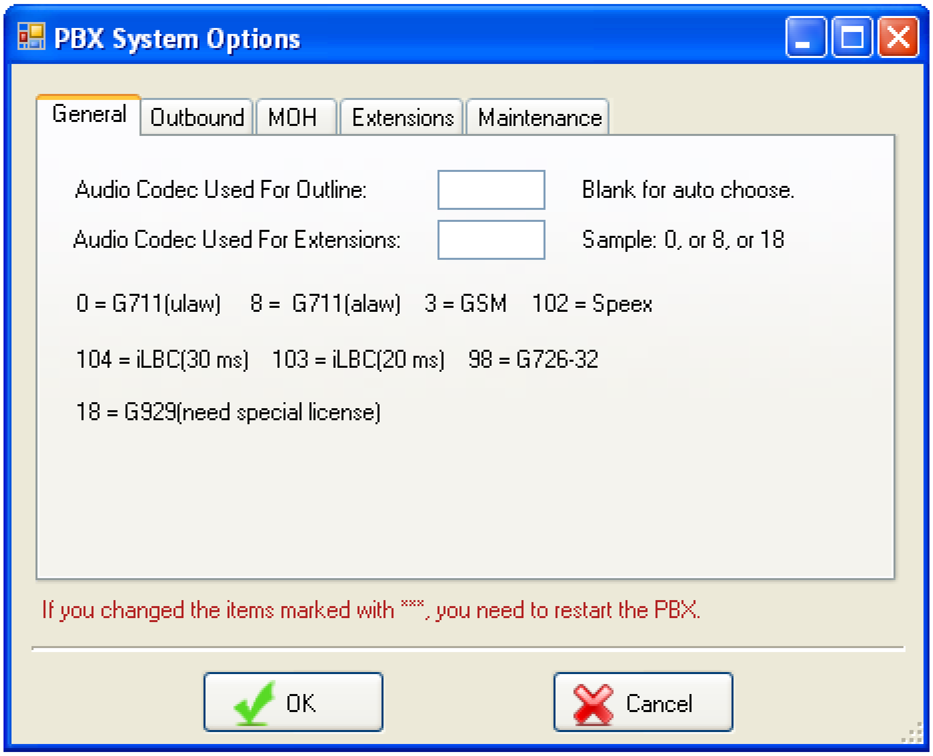


Manager port is used to for manager client to connect. PBX has a sample in SDK named "ManagerClient", which shows how to develop .NET application to receive events from PBX, or control PBX. Please refer to 7.2 about details.

6.3 System Options

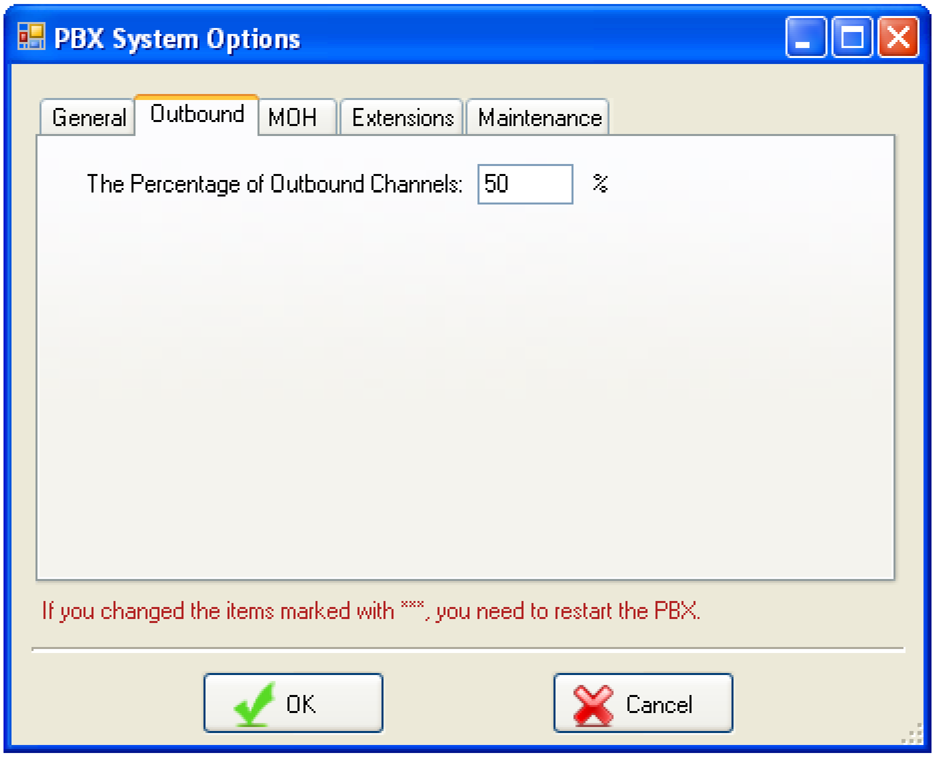


General Tab:



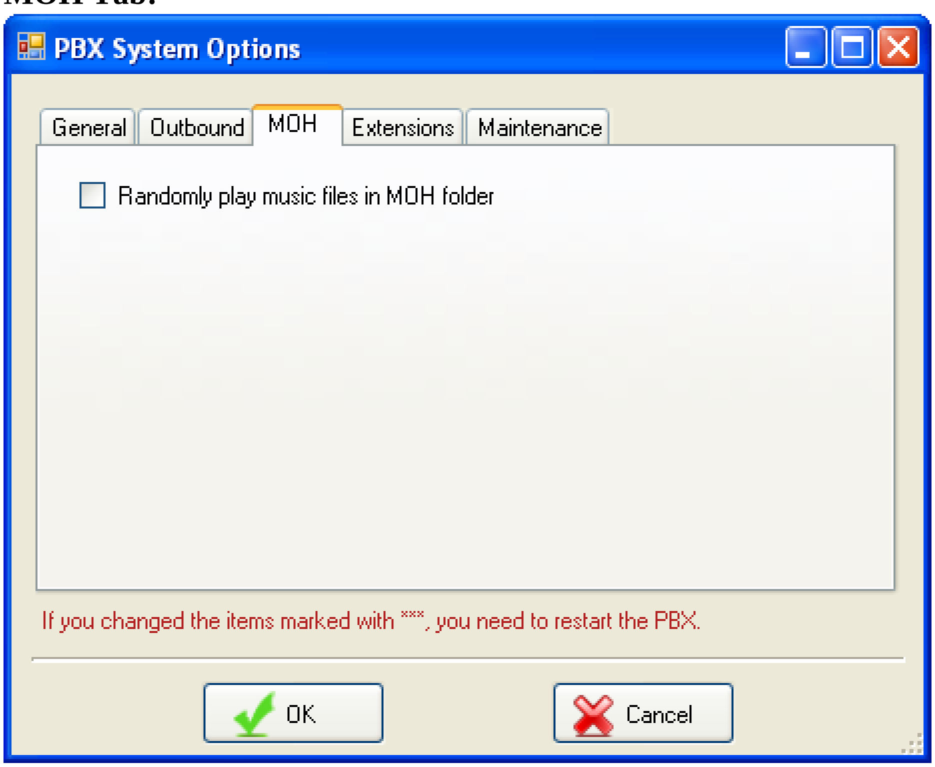
A typical example is that sometimes, you want low bandwidth audio codec using on the public network, but high quality audio codec on intranet.
Above dialog give you an option to specify the outline codec and internal codec.
For example, you can specify:
Outline: 18
Extensions: 0
It means PBX will do audio codec converting from g711 to g729 when extension calls out. In another word, PBX will use g711 to handle extension calls, and use g729 for outline.

Outbound Tab:

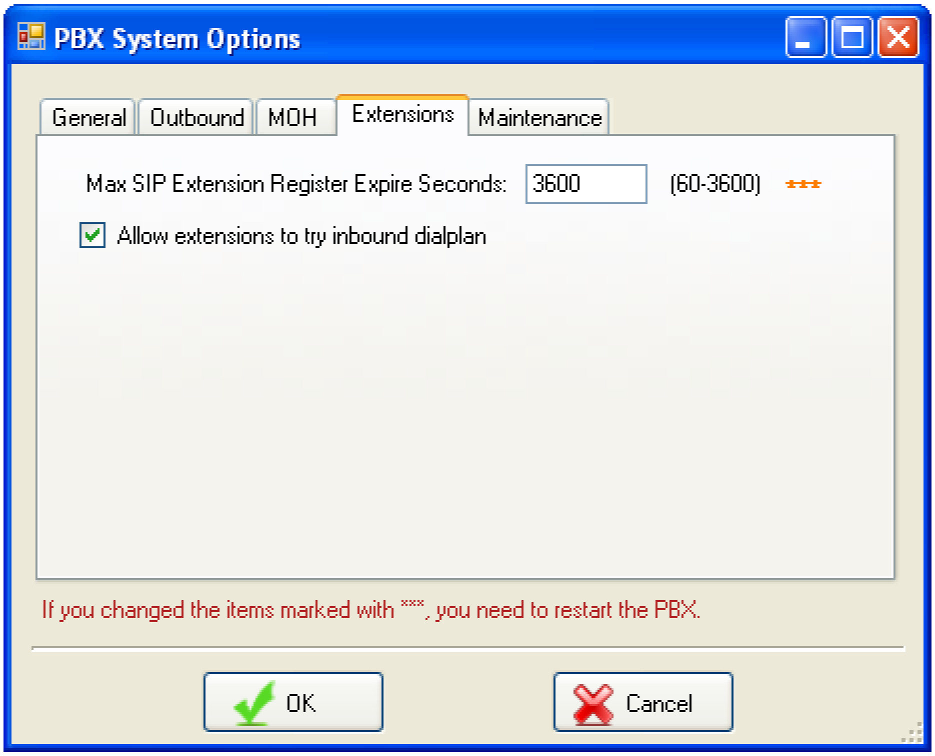


Percentage of outbound channels is for outbound calls. PBX default uses half channels for outbound, and keep half channels for inbound calls.

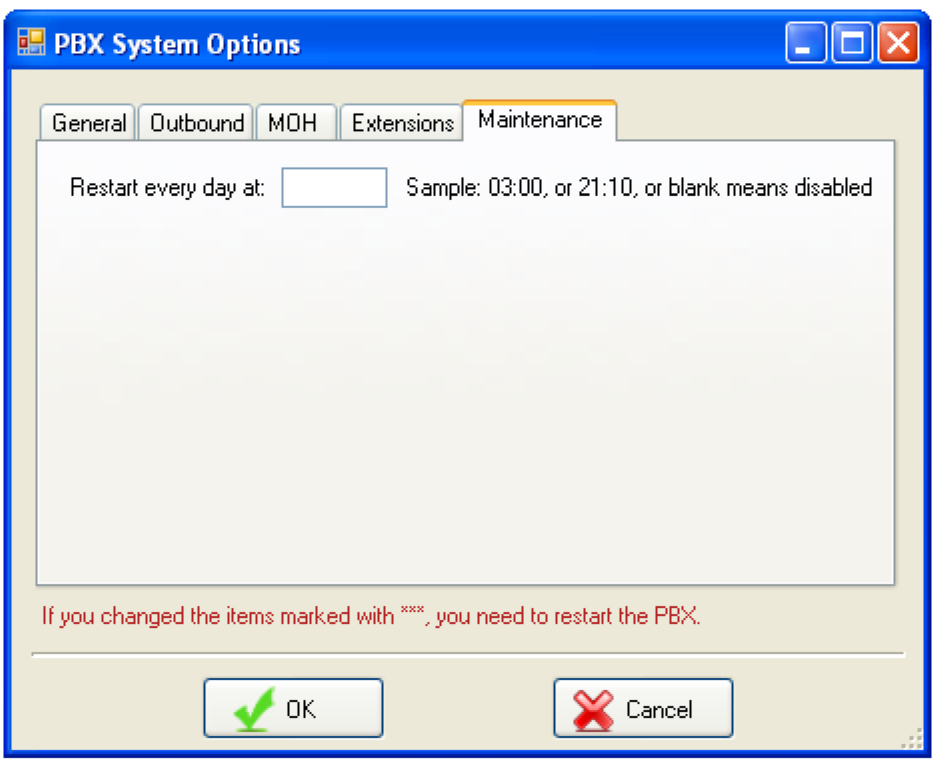
MOH Tab:



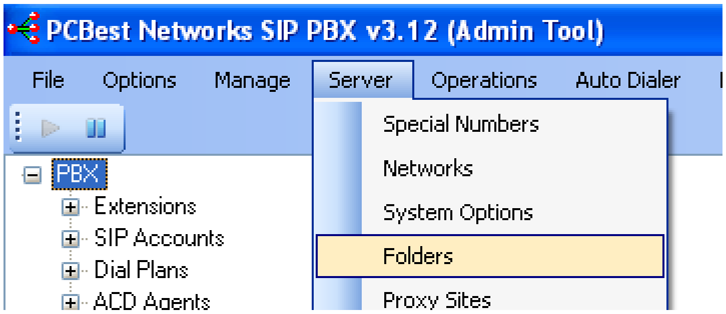
Extensions Tab:

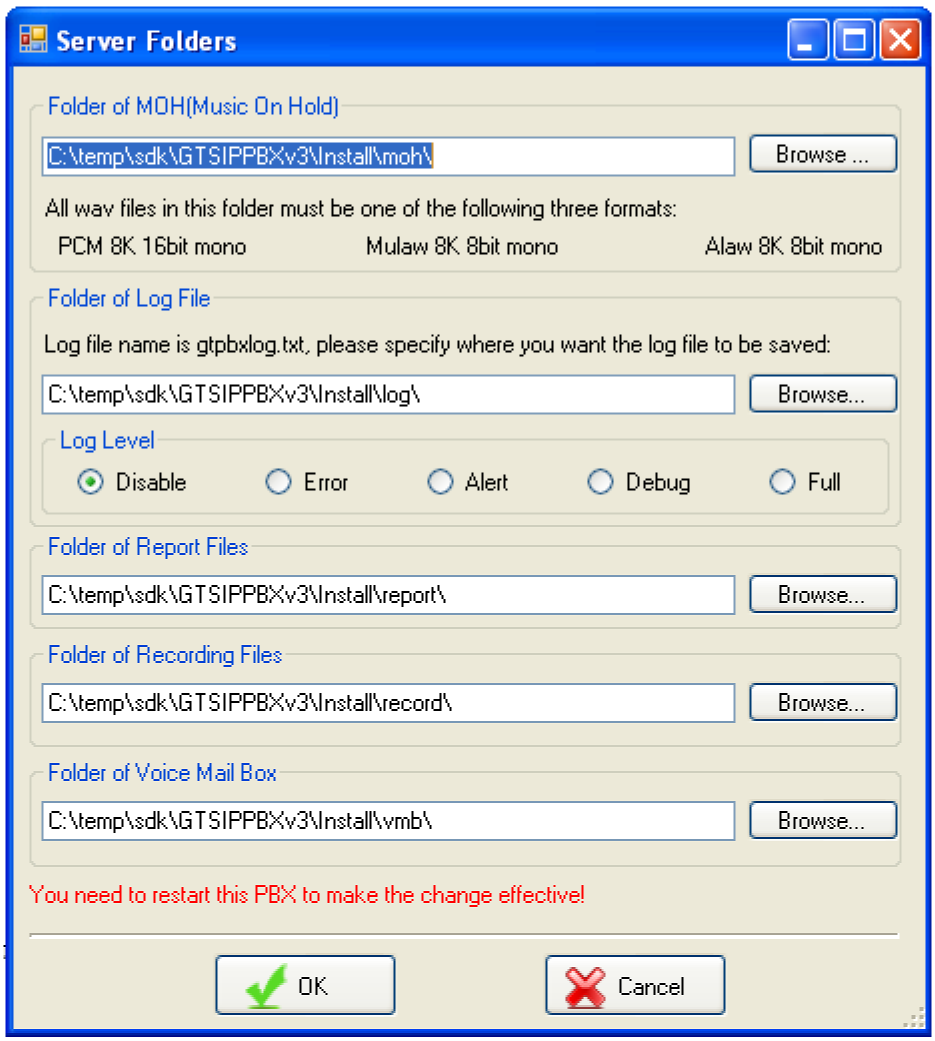


Maintenance:



6.4 Folders and Logs





Enable Log: Please set log level to Full, and after restarting PBX, you should be able to find log files in log folder.

7 PBX Developments (Pro Only)

7.1 Plugin

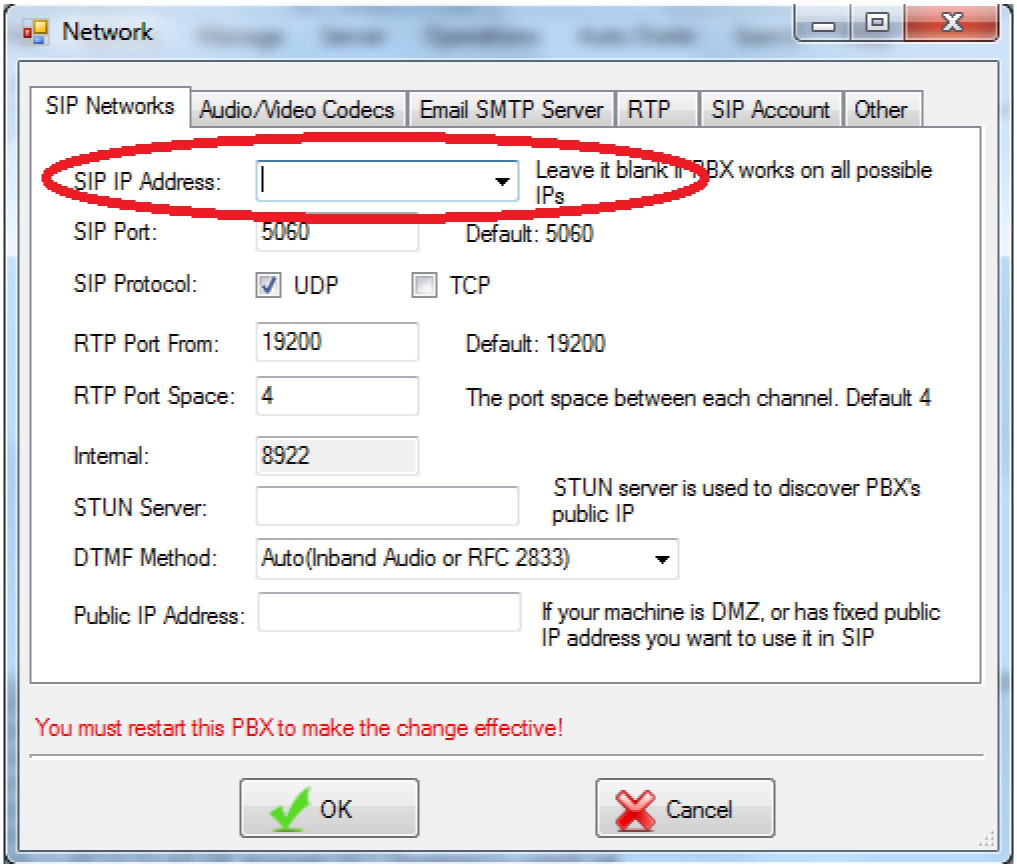
7.2 Manager Client Application (V3 only)

7.3 Database Development (V3)

8 Session Border Controller (SBC)

Sometimes you have need to transfer calls between LAN and WAN. PCBest SIP PBX has flexible features to allow you do so.

First you will need to listen on all NICs for SIP address, by setting it to blank:



Then we can set up the individual cases.

8.2 WAN to LAN

8.3 LAN to WAN

9 PBX Database Structure

9.2 cfg_sipaccounts

ID: the index of this record in DB table

DisplayName: Display Name of the SIP account. It is different than UserName. DisplayName usually can be set to anything for your preference, like your own name.

UserName: the account name that SIP provider gives

DomainServer: the domain name that SIP provider gives, usually it is SIP provider's website name, or IP address

ProxyServer: in the most of case, leave it blank or the same as DomainServer if SIP provider didn't give.

AuthName: it is the same as UserName usually, unless SIP provider has a different one.
Password: the password of this sip account

ExpireSec: how many seconds to register on the server. After that period, PBX will re-register the account and keep it alive for the SIP provider.

RegWithProxyServer: If register with SIP provider to receive inbound calls. 1: register, 0: not register

DIDS: The DID number of this SIP account.

Disabled: 1 = disabled, 0 = not disabled

ModTag: If modified by GUI program and not updated to PBX service. 1 = Modified, 0 = not modified

AcceptOtherID: When calling out, if this SIP account accepts other ID

UseLocalIPInFrom: If it can use local IP address in From(for outbound calls) rather than DomainName

MappedExten: mapped extension id

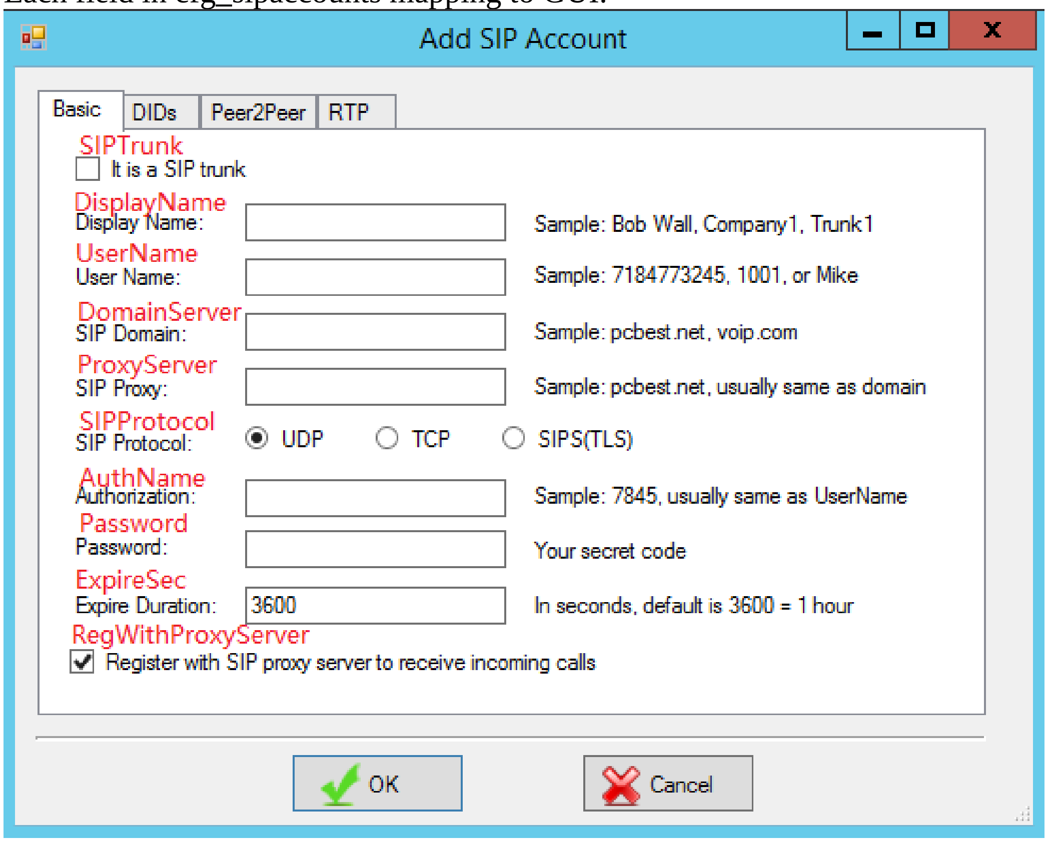
AppendExtenID: if append the extension id to from

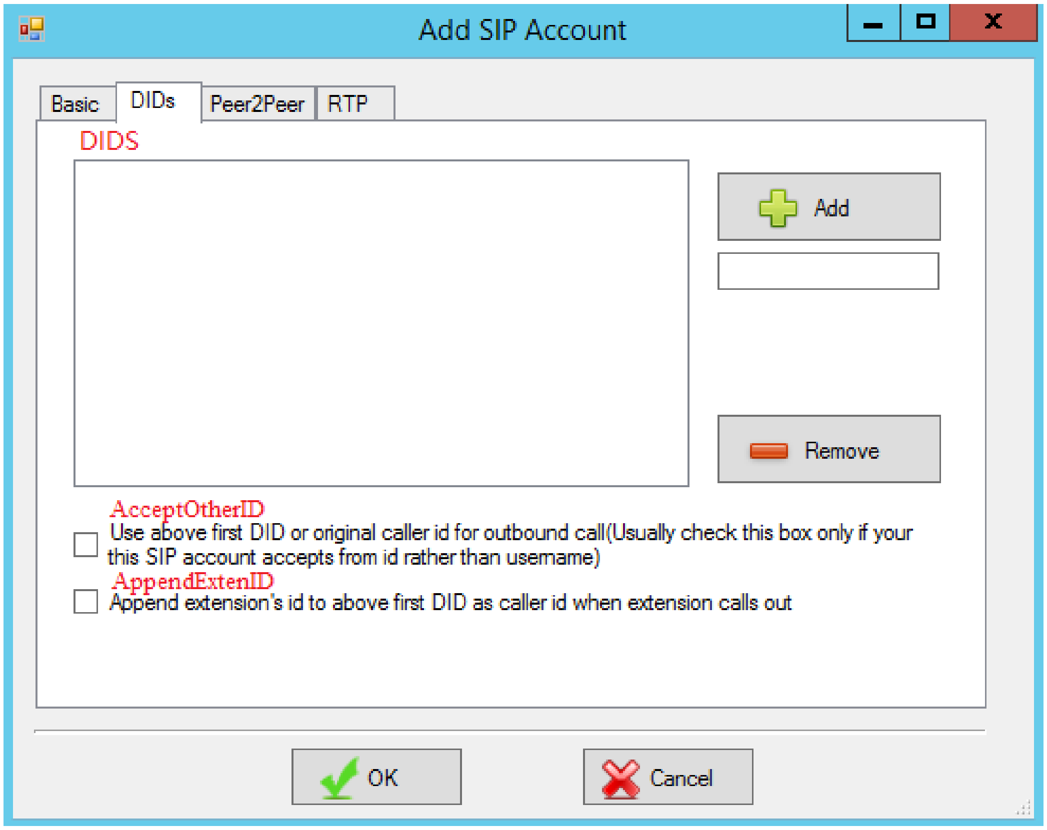
SIPProtocol: 0 = RTP, 1 = TCP, 2=SIPS(SIP on TLS)

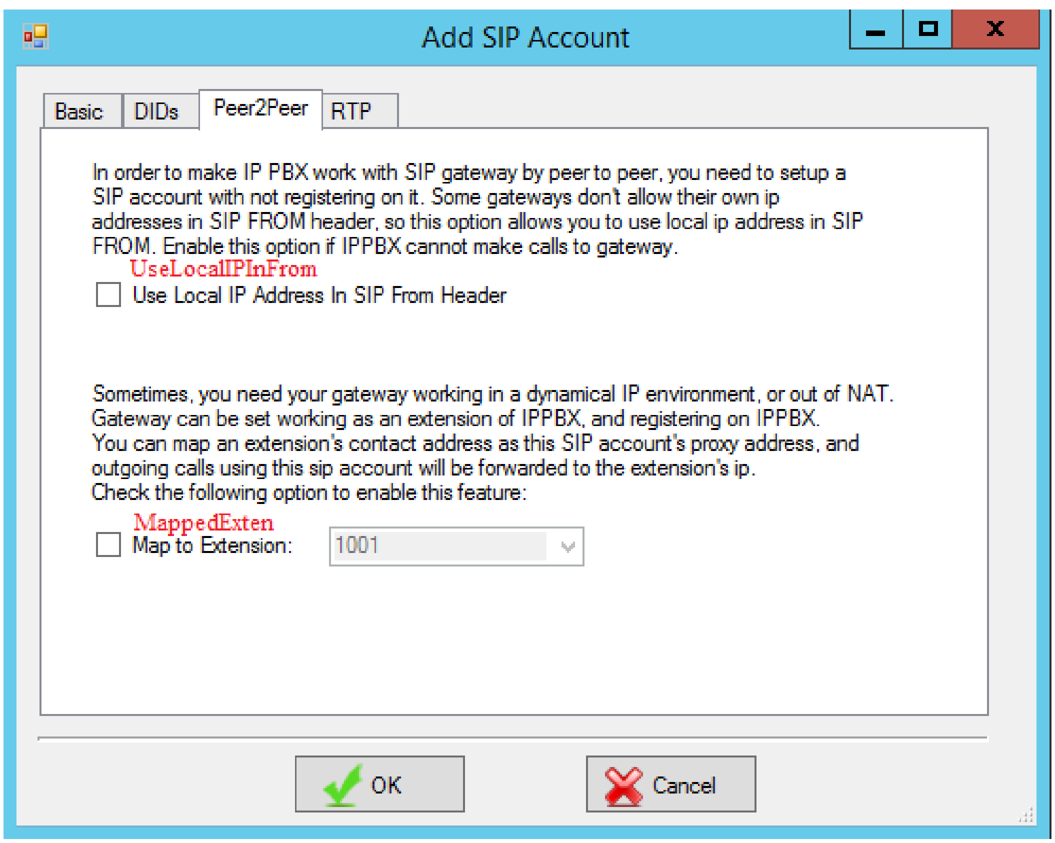
SIPTrunk: if it is a SIP trunk. 1 = yes, 0 = no

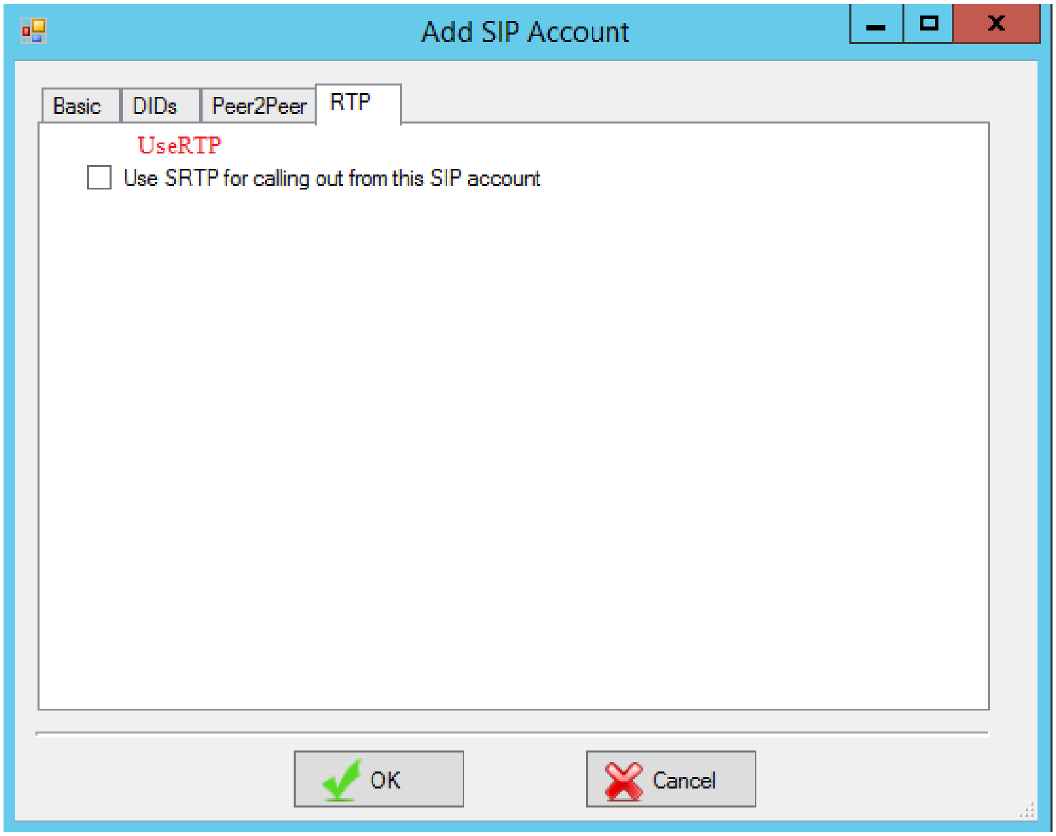
UseSRTP: if use SRTP. 1 = use, 0 = no

Each field in cfg_sipaccounts mapping to GUI:




Add SIP Account
Basic
DIDs
Peer2Peer
RTP
DIDS
Add
Remove
AcceptOtherID
Use above first DID or original caller id for outbound call(Usually check this box only if your this SIP account accepts from id rather than username)
AppendExtenID
Append extension's id to above first DID as caller id when extension calls out
OK
Cancel


Add SIP Account
Basic
DIDs
Peer2Peer
RTP
In order to make IP PBX work with SIP gateway by peer to peer, you need to setup a SIP account with not registering on it. Some gateways don't allow their own ip addresses in SIP FROM header, so this option allows you to use local ip address in SIP FROM. Enable this option if IPPBX cannot make calls to gateway.
UseLocalIPInFrom
Use Local IP Address In SIP From Header
Sometimes, you need your gateway working in a dynamical IP environment, or out of NAT. Gateway can be set working as an extension of IPPBX, and registering on IPPBX.
You can map an extension's contact address as this SIP account's proxy address, and outgoing calls using this sip account will be forwarded to the extension's ip.
Check the following option to enable this feature:
MappedExten
Map to Extension:
1001
OK
Cancel



9.3 cfg_extensions

UserName: Extension number, like 101, 1201. It must be unique to the whole PBX. This is also the ser name for SIP phone extension.

RealName: Like Bob wall, Mike Smith. Used to mark extension’s name.

Password: The password for SIP extension registration.

Email: Extension’s email address

AltPhoneNumber: outbount caller id

RegSDKTime: *Internal use only, record extension register time in Unix format*

RegisterTime: *Internal use only, record extension register time*

RegisterExpire: *Internal use only, record extension register expire seconds*

ContactAddr: *Internal use only, record registered extension contact address*

RegFromID: *Internal use only, record registered extension from ID*

RegToID: *Internal use only, record registered extension to ID*

UAName: *Internal use only, record registered extension user agent name*

NATType: *Internal use only, record registered extension NAT type*

MsgWaitingCount: *Internal use only, record registered extension waiting VM count*

MsgTotalCount: *Internal use only, record registered extension total count of messages*

MsgAccount: *Internal use only, record registered extension message account*

VoiceMsg: *Internal use only, record registered extension voice message*

PriorityLevel: extension type, 1 = normal, 8 = supervisor, 16 = virtual

VirtualExtenDestAddr: Virtual extension outbound address or number

ACDCallMethod: Method to answer ACD calls. 0 = Once registered. 1 = Once Connected with PBX special number(*9000).

RingTimeoutSec: How many seconds to forward calls after ringing. 0 = no ringing timeout

CallForwardingPlan: if it is "", it means it is this extension's voice mail box

RecordCall: 1 = Enable call recording, 0 = not recording

AcceptOtherID: if the extension accepts other called id rather than extension's username

RestSeconds: Reset interval in seconds for ACD group call distribution

VMBOn: 1 = enable Voice Mail Box, 0 = disabled

VMBPrompt: VMB prompt audio path

VMBEmail: Email address to receive the voice mail

VMBMaxLength: Maximum length of each voice mail in seconds.

VMBPassword: Voice Mail password, only digit, maximum 4 digits.

ModTag: If this line has been modified.

AuthType: 0 = Proxy, 1 = WWW, 2 = NONE

OnlyAgentLogin: ACD agent must login to use this extension. Usually only check in a call center.

MappedContactAddr: *Internal use only, record registered mapped network address*

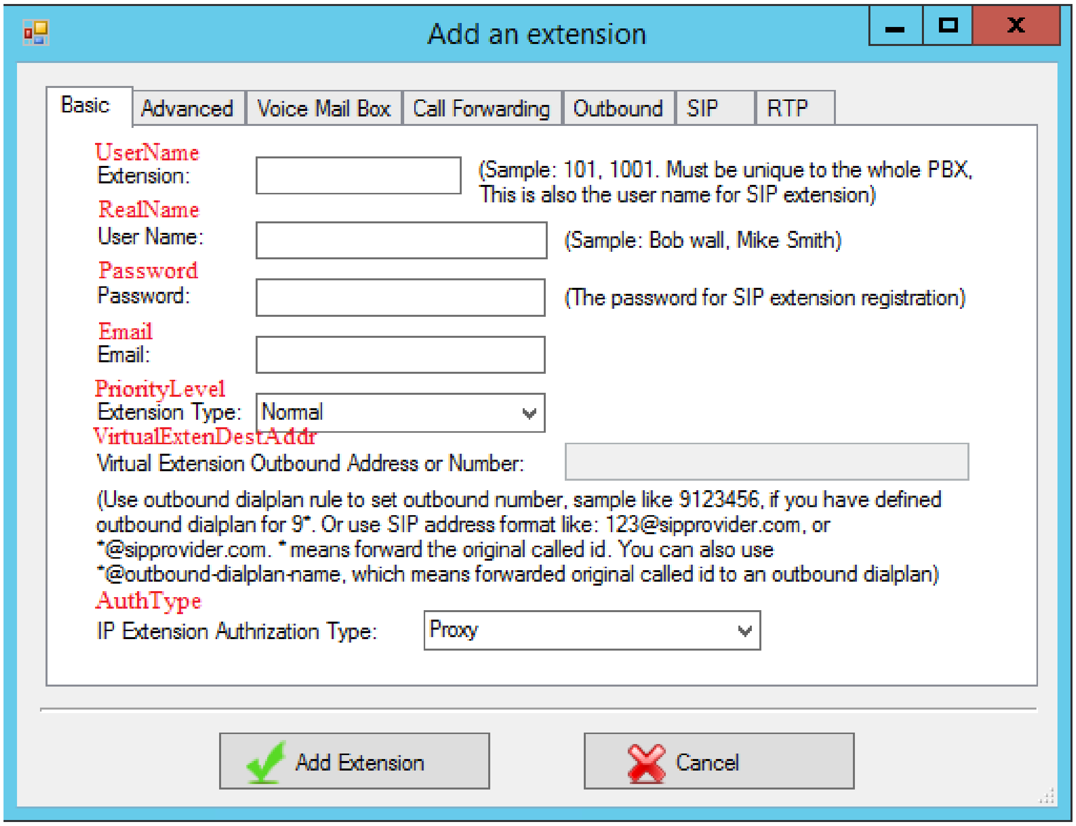
RegSrcIP: *Internal use only, record registered extension source IP*

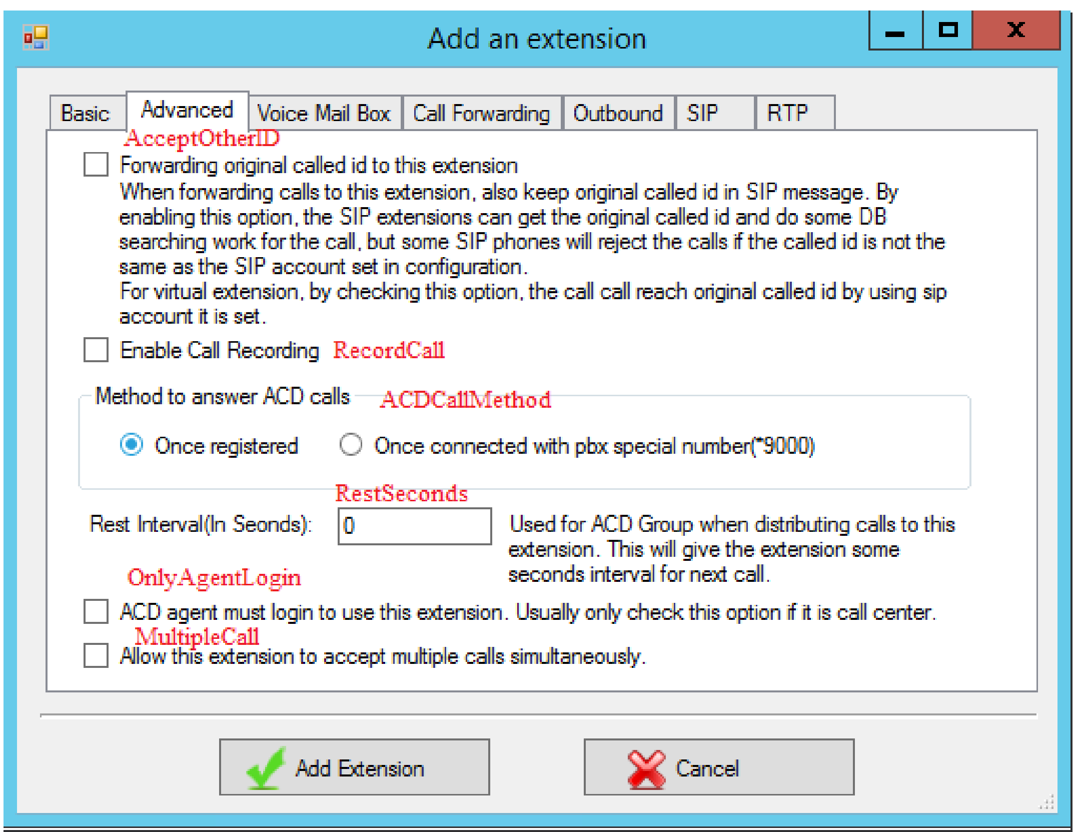
RegSrcPort: *Internal use only, record registered extension source Port*

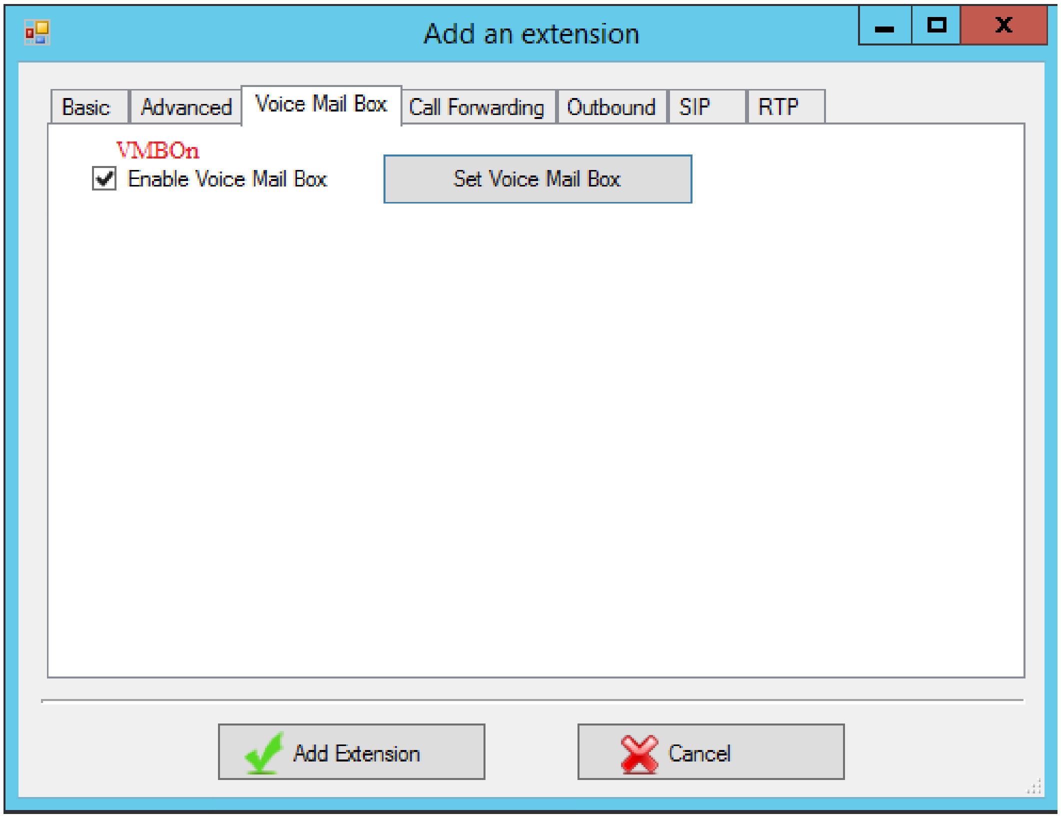
MultipleCall: Allow this extension to accept multiple calls simultaneously.

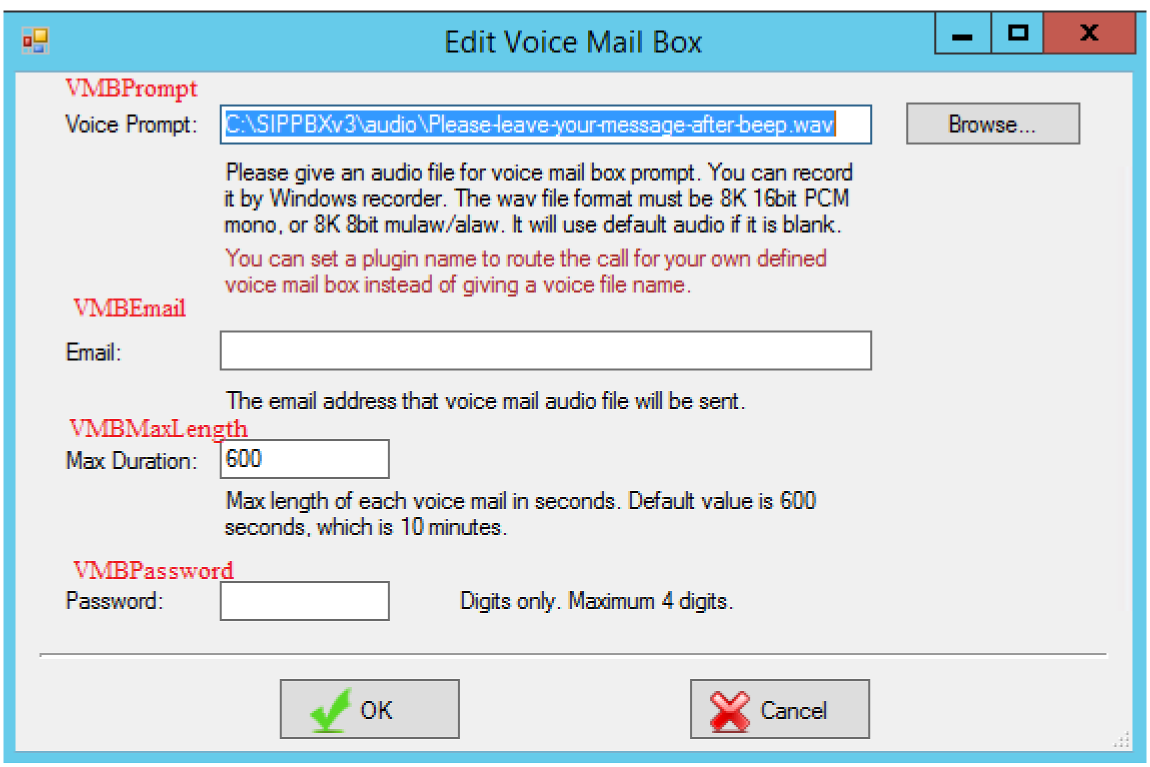
MaxRegExpSec: Maximum SIP Registration Expiration in seconds

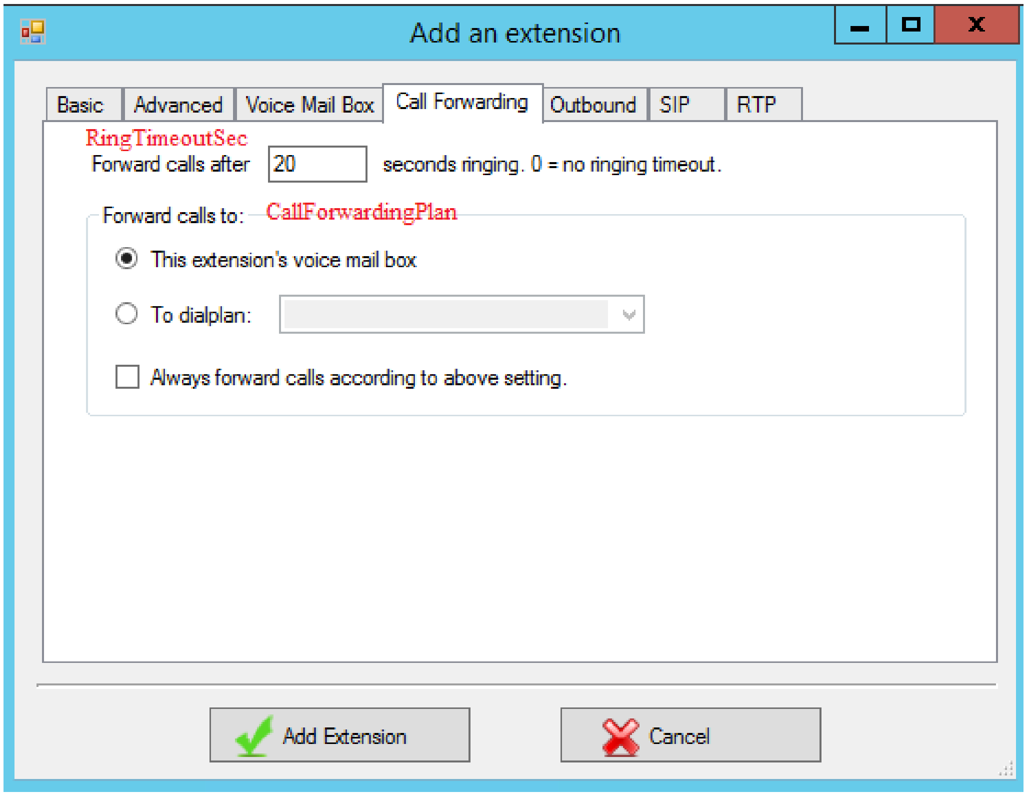
UseSRTP: Extension is enabled to use SRTP. 0 = no, 1 = yes, for every call

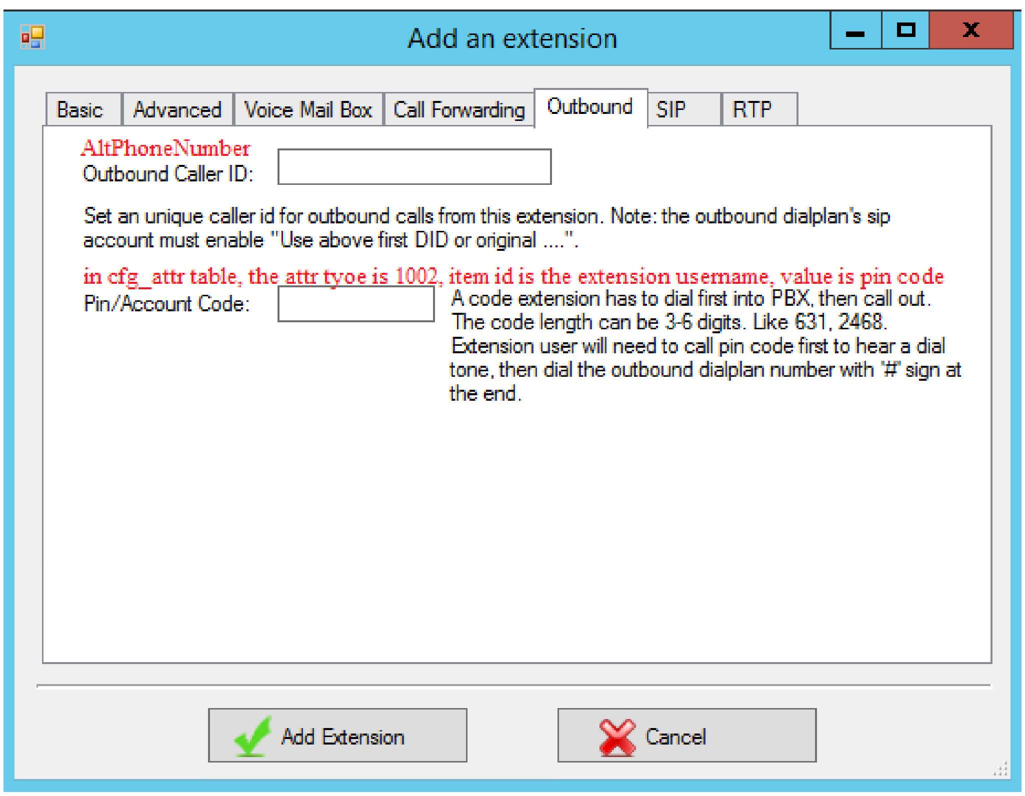

Add an extension
Basic
Advanced
Voice Mail Box
Call Forwarding
Outbound
SIP
RTP
UserName
Extension:
(Sample: 101, 1001. Must be unique to the whole PBX, This is also the user name for SIP extension)
RealName
User Name:
(Sample: Bob wall, Mike Smith)
Password
Password:
(The password for SIP extension registration)
Email
Email:
PriorityLevel
Extension Type:
Normal
VirtualExtenDestAddr
Virtual Extension Outbound Address or Number:
(Use outbound dialplan rule to set outbound number, sample like 9123456, if you have defined outbound dialplan for 9*. Or use SIP address format like: 123@sipprovider.com, or *@sipprovider.com. * means forward the original called id. You can also use *@outbound-dialplan-name, which means forwarded original called id to an outbound dialplan)
AuthType
IP Extension Authrization Type:
Proxy
Add Extension
Cancel

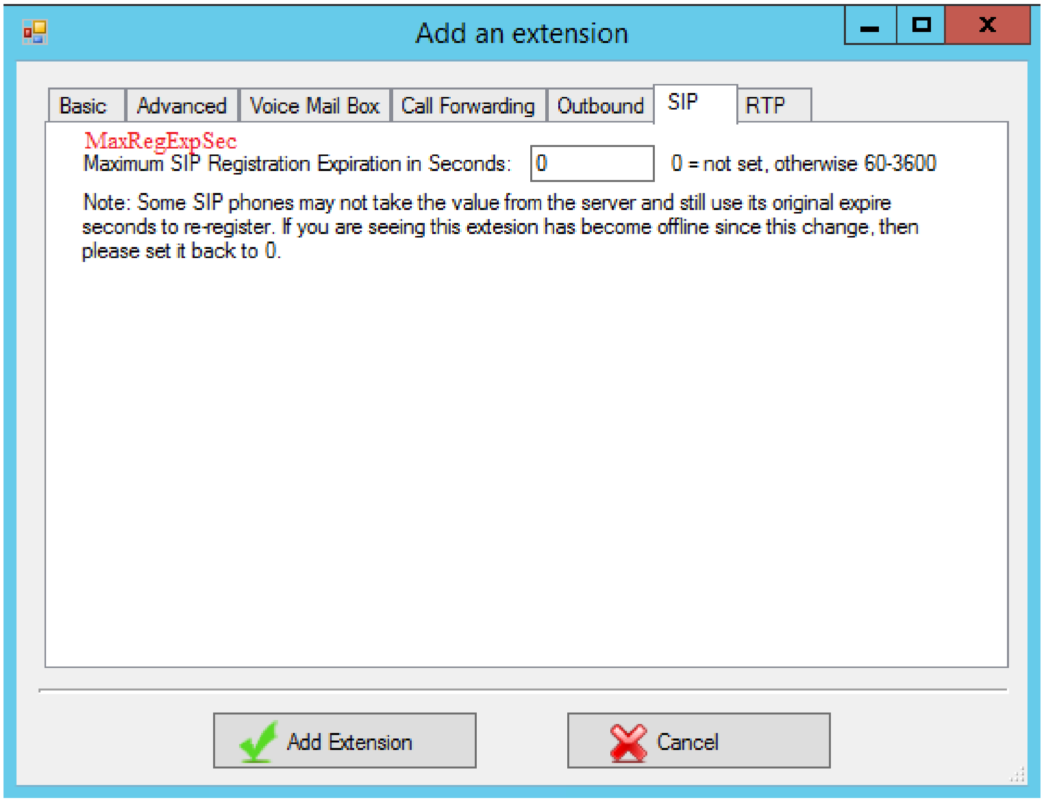

Add an extension
Basic
Advanced
Voice Mail Box
Call Forwarding
Outbound
SIP
RTP
AcceptOtherID
Forwarding original called id to this extension
When forwarding calls to this extension, also keep original called id in SIP message. By enabling this option, the SIP extensions can get the original called id and do some DB searching work for the call, but some SIP phones will reject the calls if the called id is not the same as the SIP account set in configuration.
For virtual extension, by checking this option, the call call reach original called id by using sip account it is set.
Enable Call Recording
RecordCall
Method to answer ACD calls
ACDCallMethod
Once registered
Once connected with pbx special number(*9000)
RestSeconds
Rest Interval(In Seonds):
0
Used for ACD Group when distributing calls to this extension. This will give the extension some seconds interval for next call.
OnlyAgentLogin
ACD agent must login to use this extension. Usually only check this option if it is call center.
MultipleCall
Allow this extension to accept multiple calls simultaneously.
Add Extension
Cancel

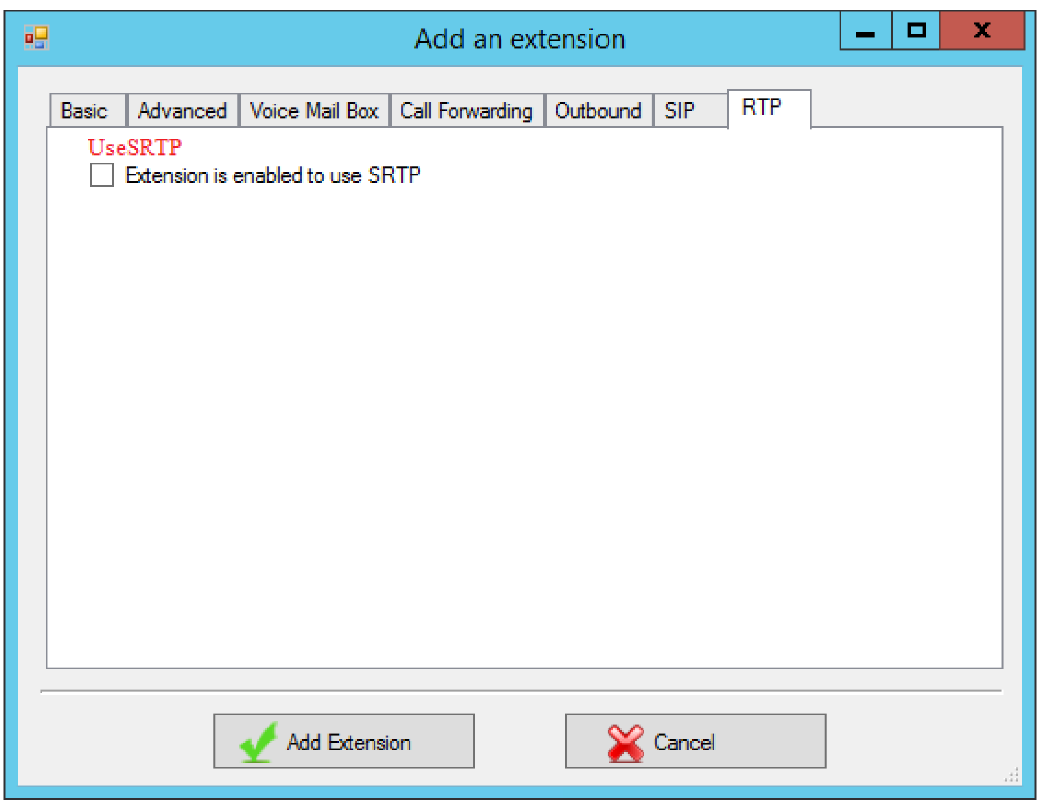

Add an extension
Basic
Advanced
Voice Mail Box
Call Forwarding
Outbound
SIP
RTP
VMBOn
Enable Voice Mail Box
Set Voice Mail Box
Add Extension
Cancel


Edit Voice Mail Box
VMBPrompt
Voice Prompt:
C:\SIPPBXv3\audio\Please-leave-your-message-after-beep.wav
Browse...
Please give an audio file for voice mail box prompt. You can record it by Windows recorder. The wav file format must be 8K 16bit PCM mono, or 8K 8bit mulaw/alaw. It will use default audio if it is blank.
You can set a plugin name to route the call for your own defined voice mail box instead of giving a voice file name.
VMBEmail
Email:
The email address that voice mail audio file will be sent.
VMBMaxLength
Max Duration:
600
Max length of each voice mail in seconds. Default value is 600 seconds, which is 10 minutes.
VMBPassword
Password:
Digits only. Maximum 4 digits.
OK
Cancel


Add an extension
Basic
Advanced
Voice Mail Box
Call Forwarding
Outbound
SIP
RTP
RingTimeoutSec
Forward calls after
20
seconds ringing. 0 = no ringing timeout.
Forward calls to:
CallForwardingPlan
This extension's voice mail box
To dialplan:
Always forward calls according to above setting.
Add Extension
Cancel


Add an extension
Basic
Advanced
Voice Mail Box
Call Forwarding
Outbound
SIP
RTP
AltPhoneNumber
Outbound Caller ID:
Set an unique caller id for outbound calls from this extension. Note: the outbound dialplan's sip account must enable "Use above first DID or original ....".
in cfg_attr table, the attr tyoe is 1002, item id is the extension username, value is pin code
Pin/Account Code:
A code extension has to dial first into PBX, then call out. The code length can be 3-6 digits. Like 631, 2468. Extension user will need to call pin code first to hear a dial tone, then dial the outbound dialplan number with '#' sign at the end.
Add Extension
Cancel


Add an extension
Basic
Advanced
Voice Mail Box
Call Forwarding
Outbound
SIP
RTP
MaxRegExpSec
Maximum SIP Registration Expiration in Seconds:
0
0 = not set, otherwise 60-3600
Note: Some SIP phones may not take the value from the server and still use its original expire seconds to re-register. If you are seeing this extesion has become offline since this change, then please set it back to 0.
Add Extension
Cancel


Add an extension
Basic
Advanced
Voice Mail Box
Call Forwarding
Outbound
SIP
RTP
UseSRTP
Extension is enabled to use SRTP
Add Extension
Cancel

9.4 cfg_dialplans

table to record dialplan settings.

DialPlanName: Dialplan Name

CallDirection: Dialplan call direction, for inbound or outbound

Caller: The caller id to match this plan

Called: The called id to match this plan

CallPlan: Plan Template

DestAddress: Plan template name or destination

OutboundPreStrip: Outbound called number prestrip before dialing out

OutboundPrepend: Outbound called number prepend before dialing out

OutboundSIPAcct: SIP account index. The first SIP account index is recorded in first 8 bits, the second is recorded in 0xff00, and the third is at 0xff0000.

OutboundCallerID: not used

ExtenPriorityLevel: not used

TimeLimited: 1 = enabled time schedule, 0 = disable time schedule

TimeStartHour: time schedule start hour

TimeStartMinute: time schedule start minute

TimeEndHour: time schedule end hour

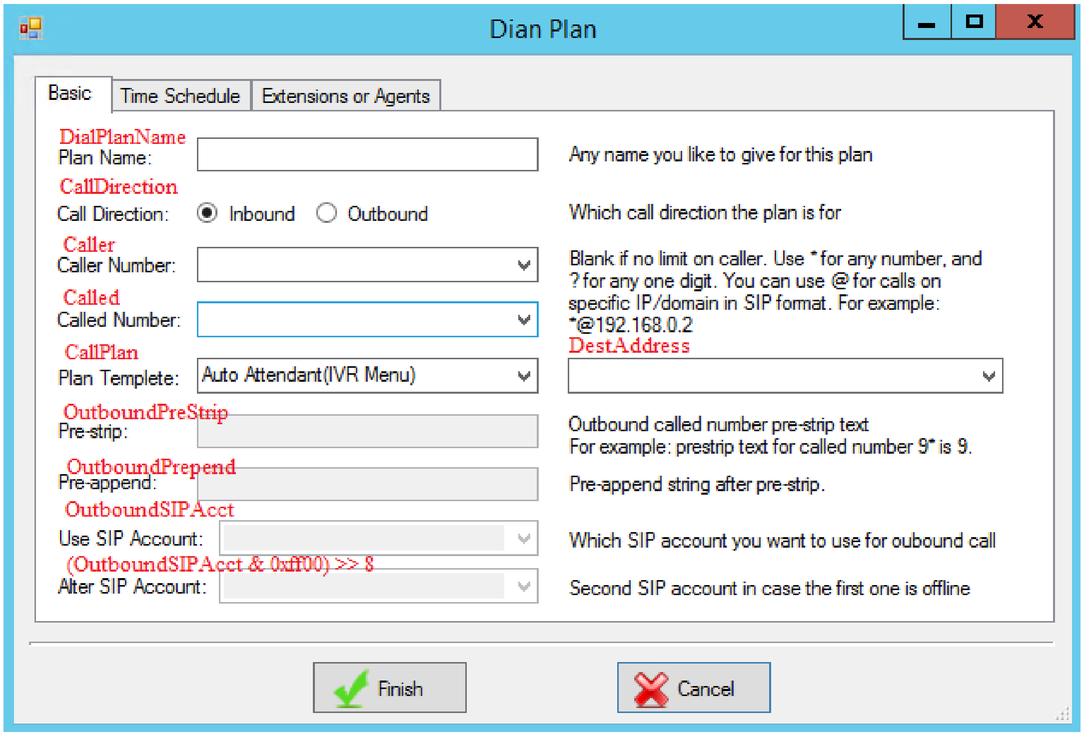
TimeEndMinute: time schedule end minute

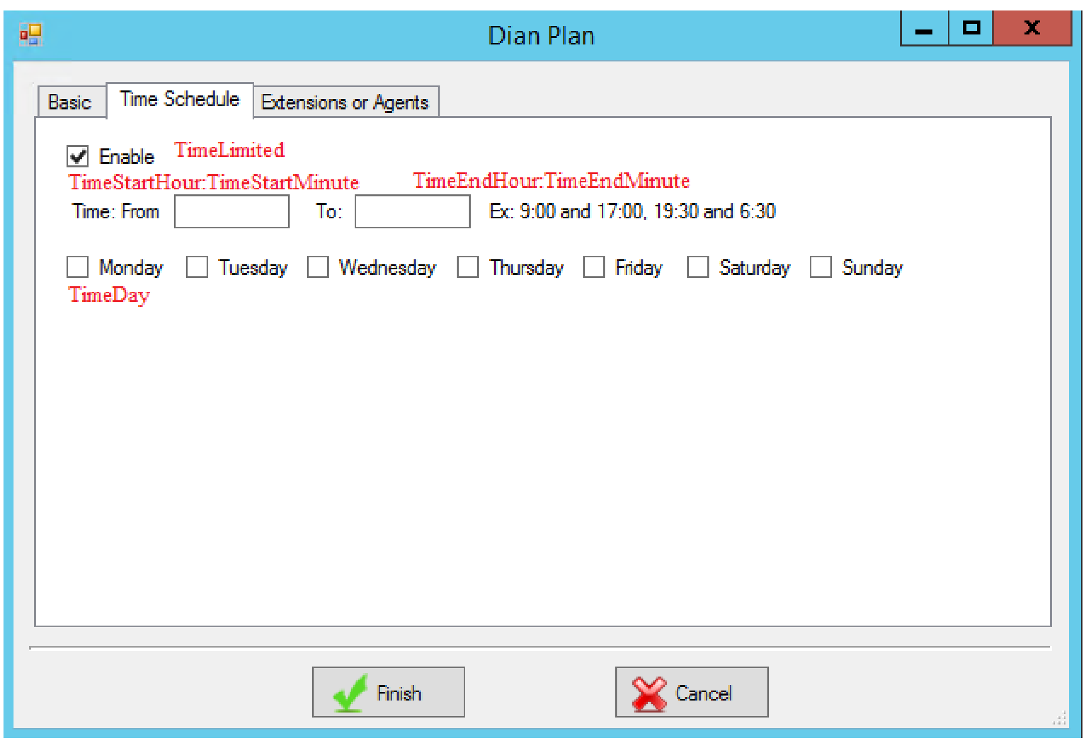
TimeDay: time schedule day of week

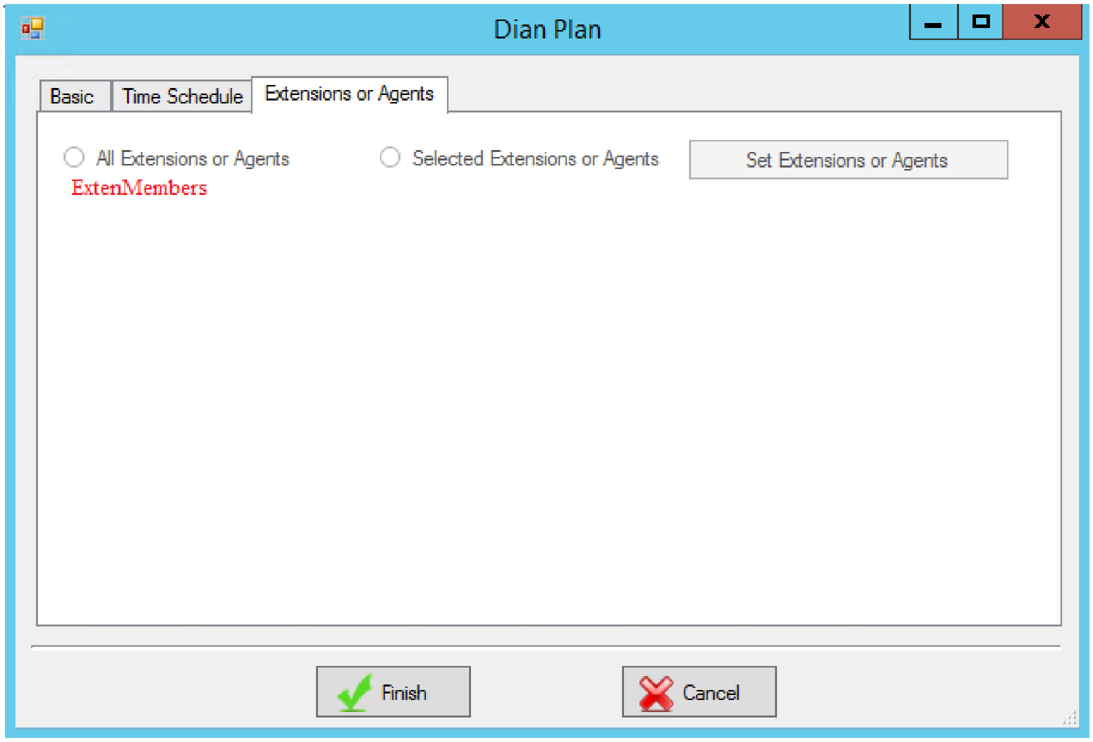
ExtenMembers: selected extensions or agents which can use this dialplan

ModTag: If this dialplan has been changed by GUI and not refreshed into PBX

DialPlanIndex: internal use, for index of a dialplan







9.5 cfg_huntgroups

table to record all hunt groups (ACD groups)

Name: The name of hunt group

Type: The type of hunt group, Linear/Circular/Most Idle/Most Skill

PlayMOH: If play music on hold when waiting

MOHDir: The directory saving MOH music files

DialplanDTMF: The DTMF which caller can press to go a dialplan

DialplanName: the dialplan name above DTMF routes the call to

WaitTimeout: Caller wait timeout in seconds

WaitTimeoutTo: The call is routed to VMB or Dialplan

VMBDTMF: The DTMF to leave a voice message

VMBOn: Voice Mailbox is on

VMBPrompt: Voice Mailbox Prompt

VMBEmail: Voice Mailbox Email

VMBMaxLength: The maximum length of a voice mail, in seconds

VMBPassword: Voice mailbox password

AgentType: 0 =Extension, 1 = Agent login, 2 = ACD login

Agents: Agent list

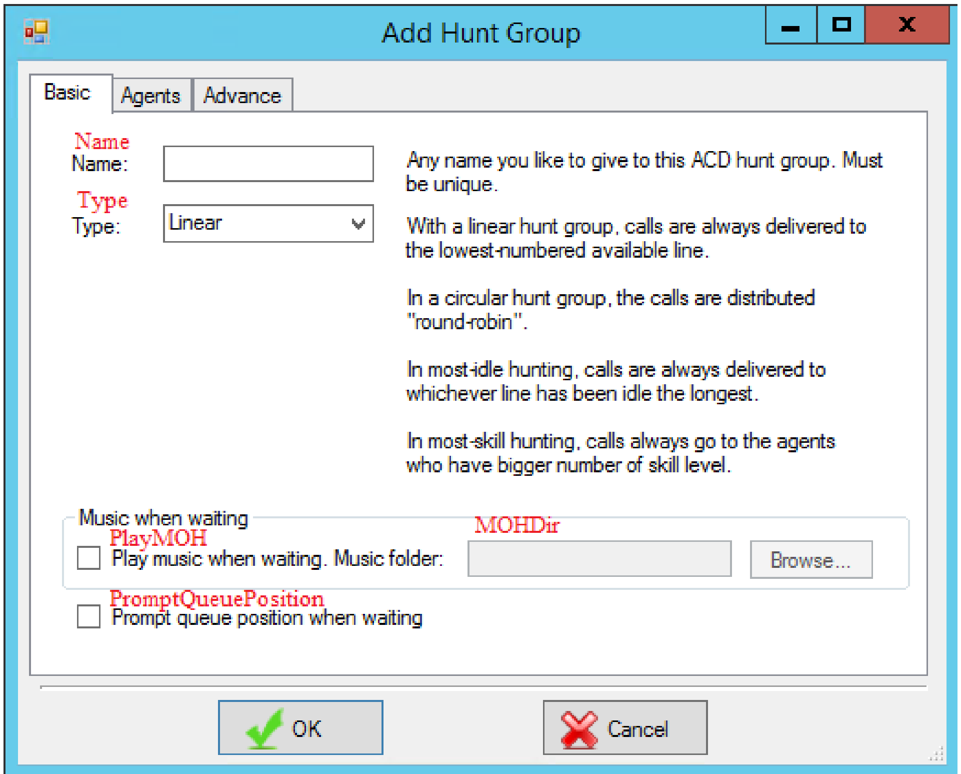
ModTag: If the record has been changed by GUI, and not refreshed/taken effect into PBX service

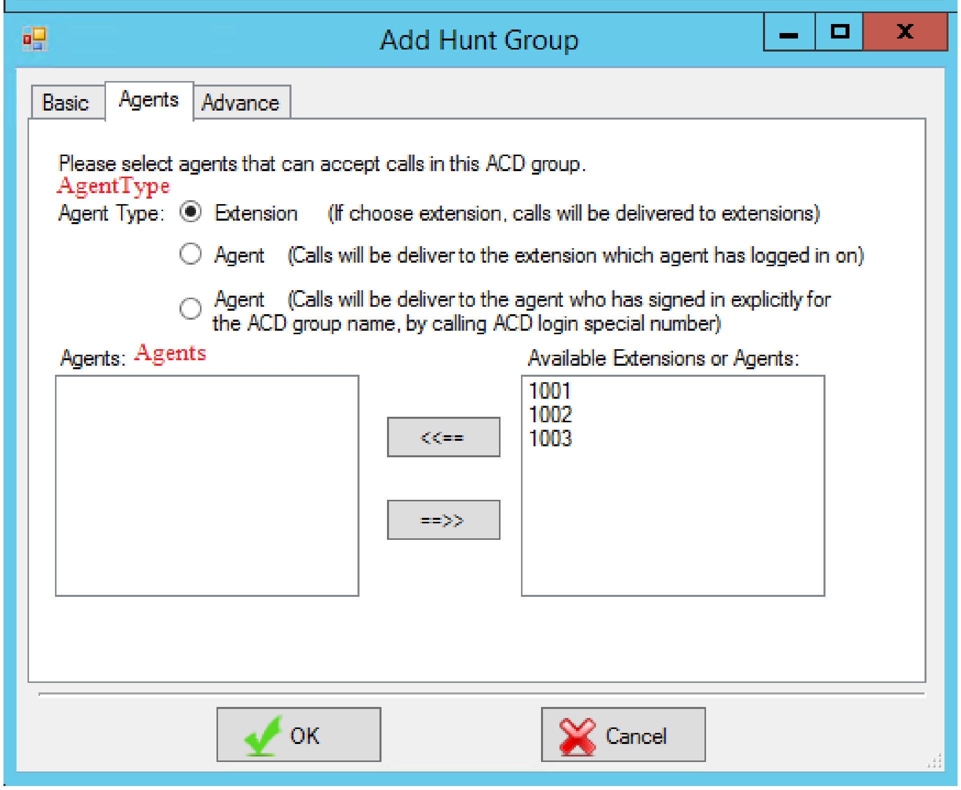
MaxNumOfCalls: The max number of calls in queue

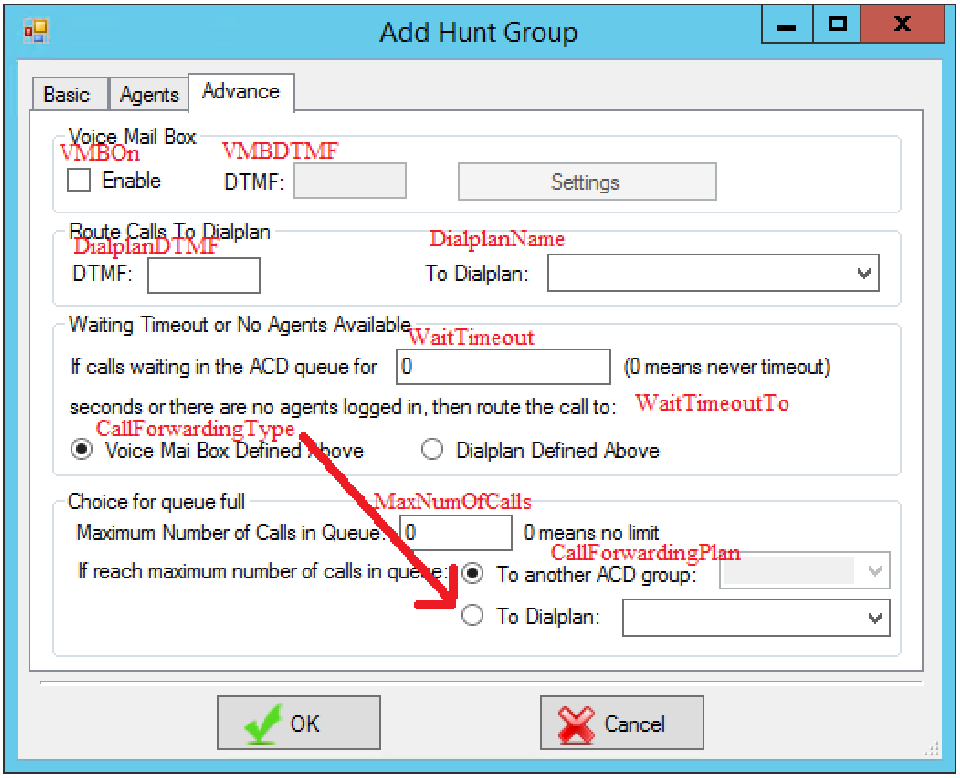
CallForwardingType: If reaching maximum number of calls, forwarding type, 0 = to another ACD group, 1 = to dialplan

CallForwardingPlan: If to dialplan, the name of it

PromptQueuePosition: If prompt the position in queue when waiting


Add Hunt Group
Basic
Agents
Advance
Name
Name:
Any name you like to give to this ACD hunt group. Must be unique.
Type
Type:
Linear
With a linear hunt group, calls are always delivered to the lowest-numbered available line.
In a circular hunt group, the calls are distributed "round-robin".
In most-idle hunting, calls are always delivered to whichever line has been idle the longest.
In most-skill hunting, calls always go to the agents who have bigger number of skill level.
Music when waiting
PlayMOH
Play music when waiting. Music folder:
MOHDir
Browse...
PromptQueuePosition
Prompt queue position when waiting
OK
Cancel


Add Hunt Group
Basic
Agents
Advance
Please select agents that can accept calls in this ACD group.
AgentType
Agent Type:
Extension (If choose extension, calls will be delivered to extensions)
Agent (Calls will be deliver to the extension which agent has logged in on)
Agent (Calls will be deliver to the agent who has signed in explicitly for the ACD group name, by calling ACD login special number)
Agents:
Agents
Available Extensions or Agents:
<<==
==>>
1001
1002
1003
OK
Cancel



9.6 cfg_parkingslots

Name: The name of the parking slot

DTMFStr: The DTMF or number to park the caller's call to this parking slot.

PlayMOH: If play music on hold when waiting

MOHDir: The directory that has the music on hold audio files

DialplanDTMF: The DTMF to redirect the call to a dialplan

DialplanName: The name of the dialplan

WaitTimeout: The maximum wait time for a caller in parking slot

WaitTimeoutTo: 0 = Voice Mailbox, 1 = Dialplan defined above

VMBDTMF: The DTMF to leave a message

VMBOn: 1 = Voice Mailbox is enabled, 0 = no VMB

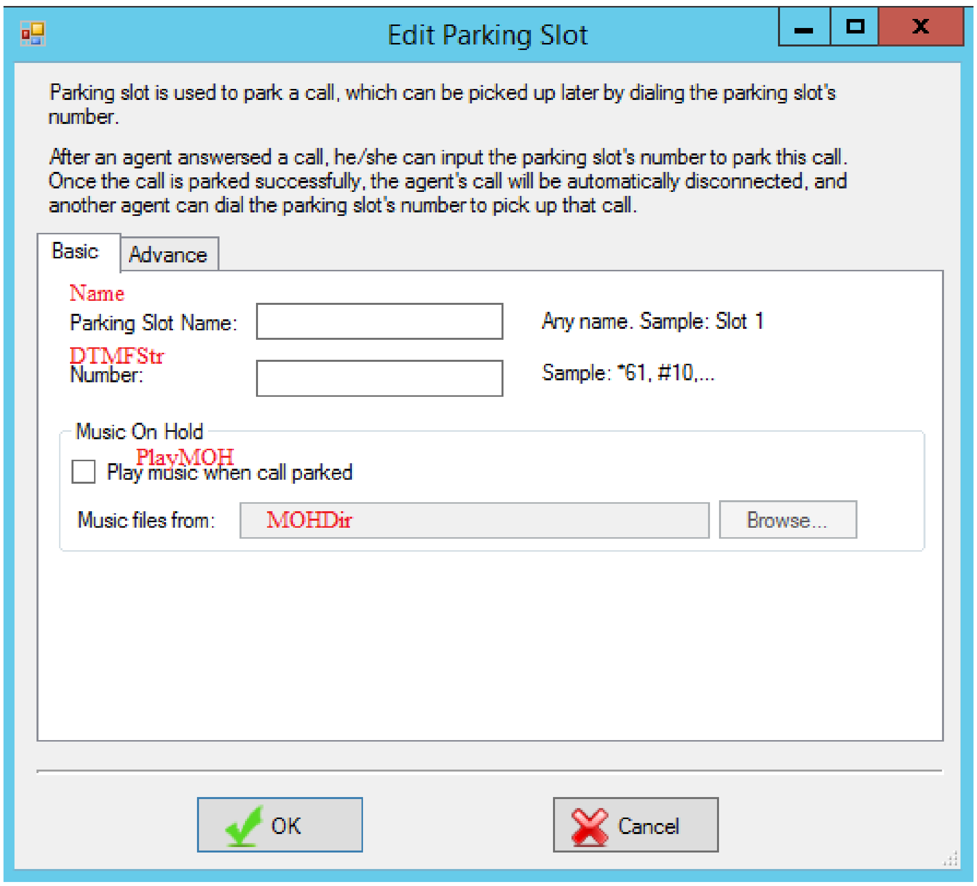
VMBPrompt: Voice Mailbox Prompt

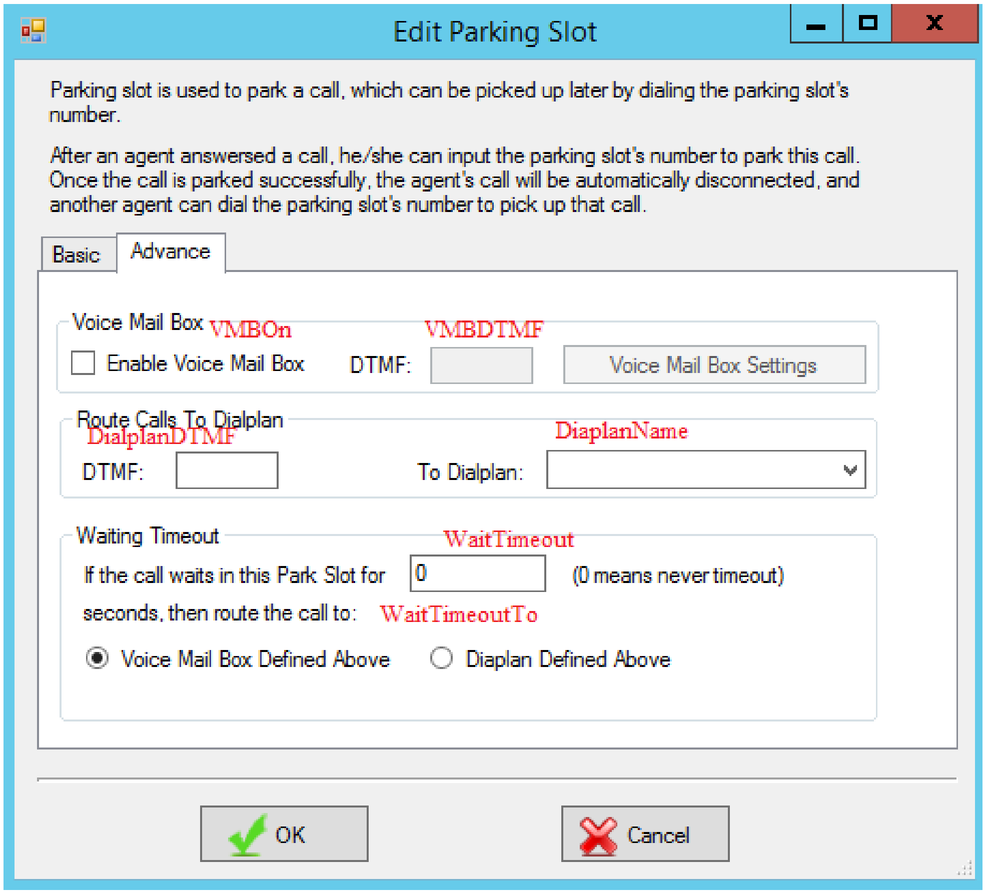
VMBEmail: Voice Mailbox email

VMBMaxLength: The maximum length of Voice Mailbox

VMBPassword: The password of Voice Mailbox

ModTag: 1 = changed by GUI, not taken effect by PBX service





9.7 cfg_ringgroups

Name: The name of this ring group

Type: 0 = ring all destionations at same time, 1 = ring destinations by order

PlayMOH: If play music on hold when waiting

MOHDir: The directory that has the music on hold audio files

VMBOn: 1 = Voice Mailbox is enabled, 0 = no VMB

VMBPrompt: Voice Mailbox Prompt

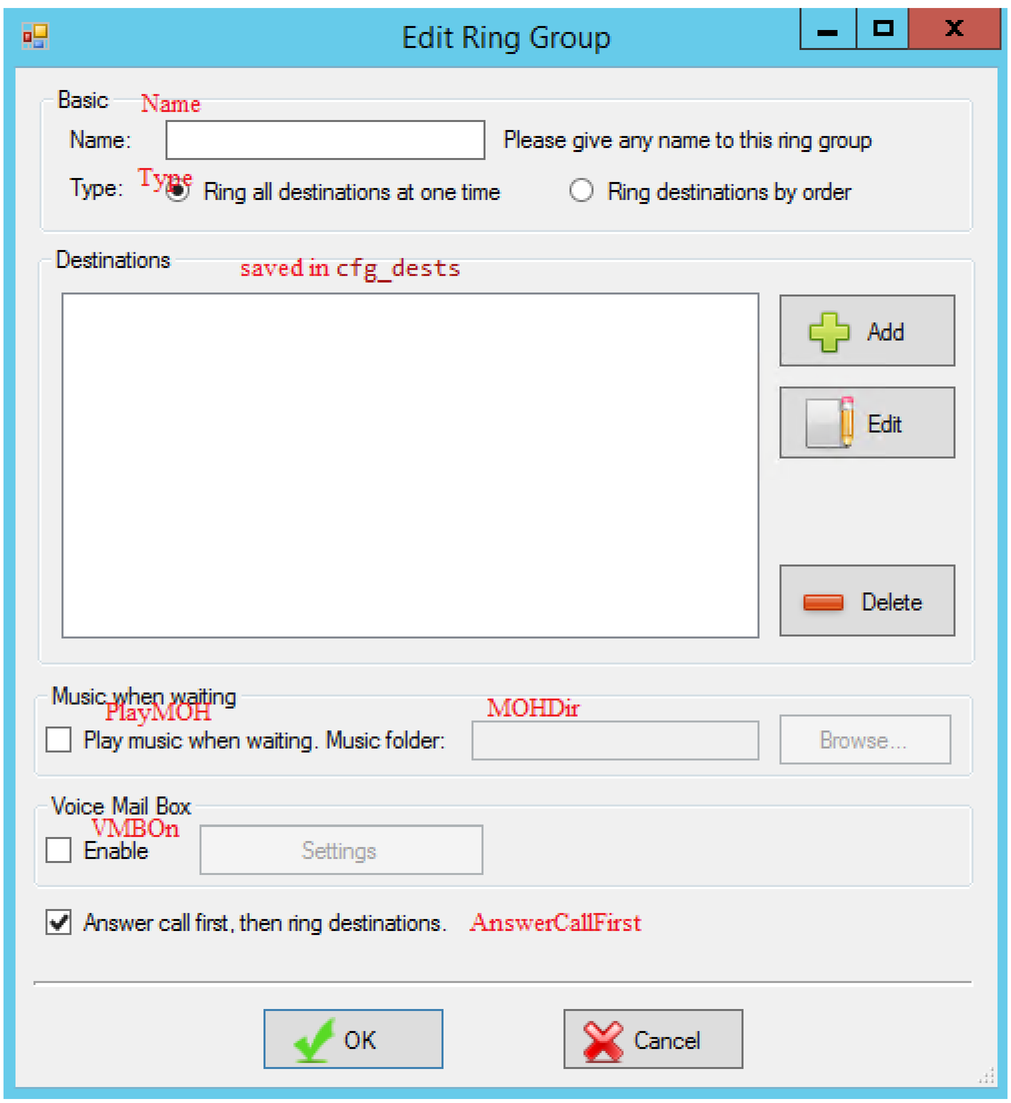
VMBEmail: Voice Mailbox email

VMBMaxLength: The maximum length of Voice Mailbox

VMBPassword: The password of Voice Mailbox

AnswerCallFirst: 1 = answer the call first, then ring destionations

ModTag: 1 = changed by GUI, not taken effect by PBX service



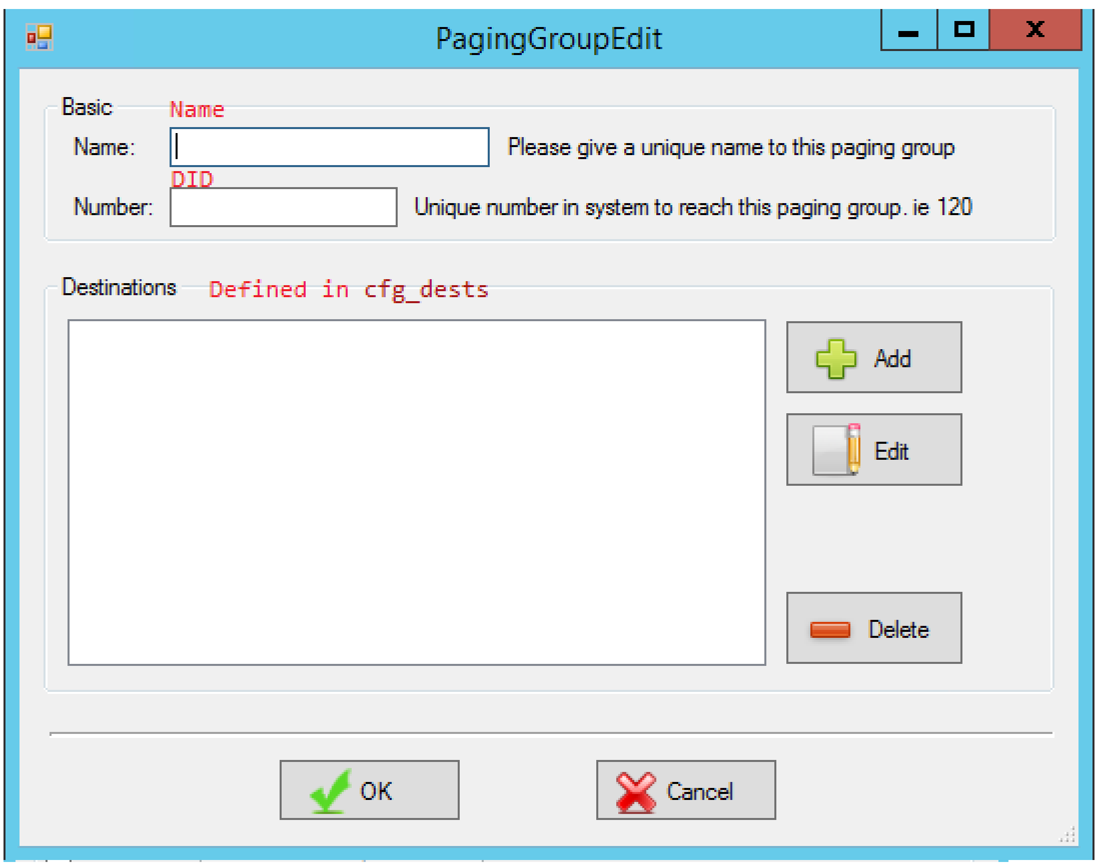
9.8 cfg_paginggroups

Name: The name of Paging Group

DID: The unique number in the system to reach this paging group, ie 120

UseGroupName: Not used

ModTag: 1 = changed by GUI, not taken effect by PBX service



9.9 cfg_monitorgroups

Name: The name of monitor group

Number: Special number for supervisor to call. If it is a blank, you can still route the call to monitor group by dialplan.

PasswordPrompt: The prompt audio file of password

Password: The password of this monitor group

KeyBargeIn: When monitoring, press this key to speak

KeyBargeOut: When monitoring, press this key to jump out of conversation, and choose another extension to continue

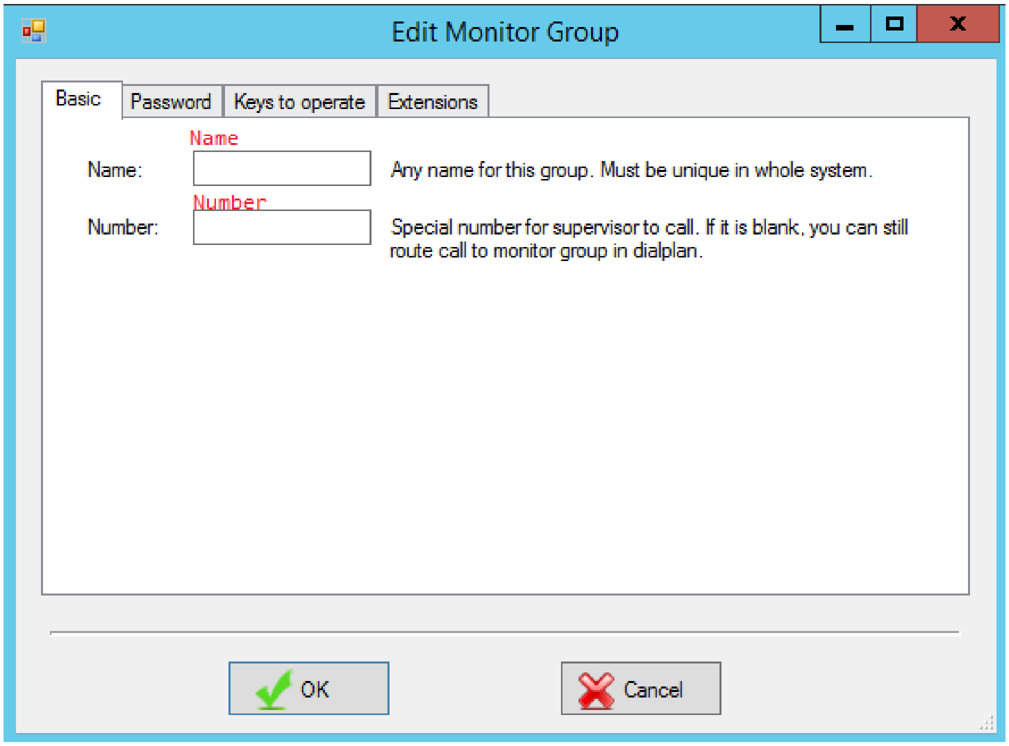
KeyWhisper: When monitoring, press this key to whisper to agent

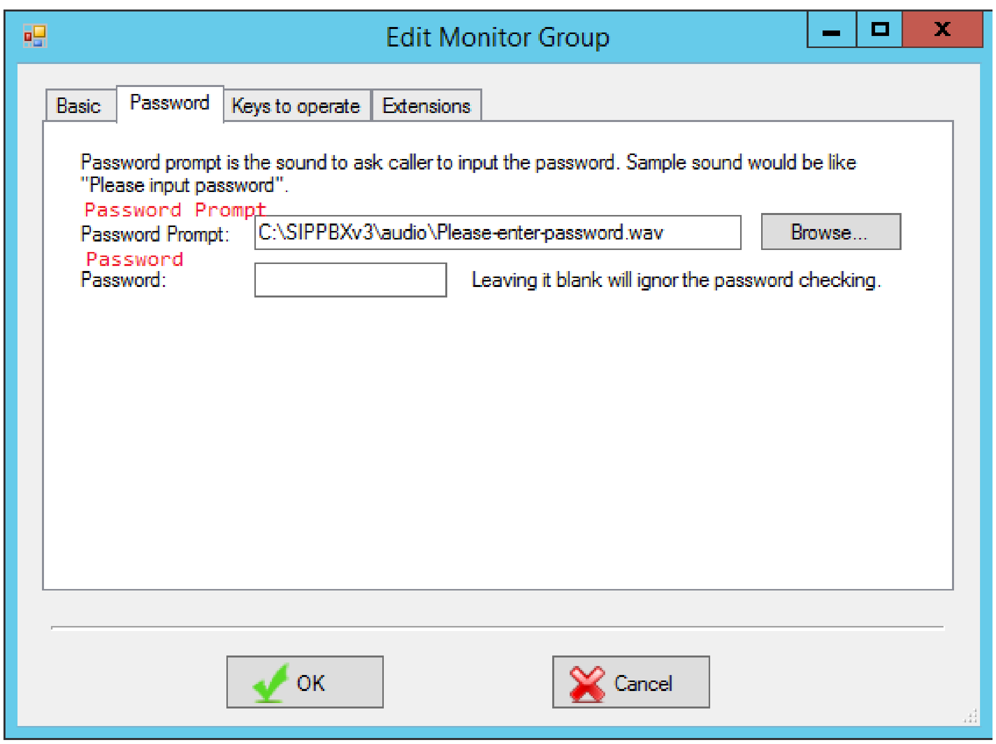
ExtenPrompt: Prompt for inputting extension

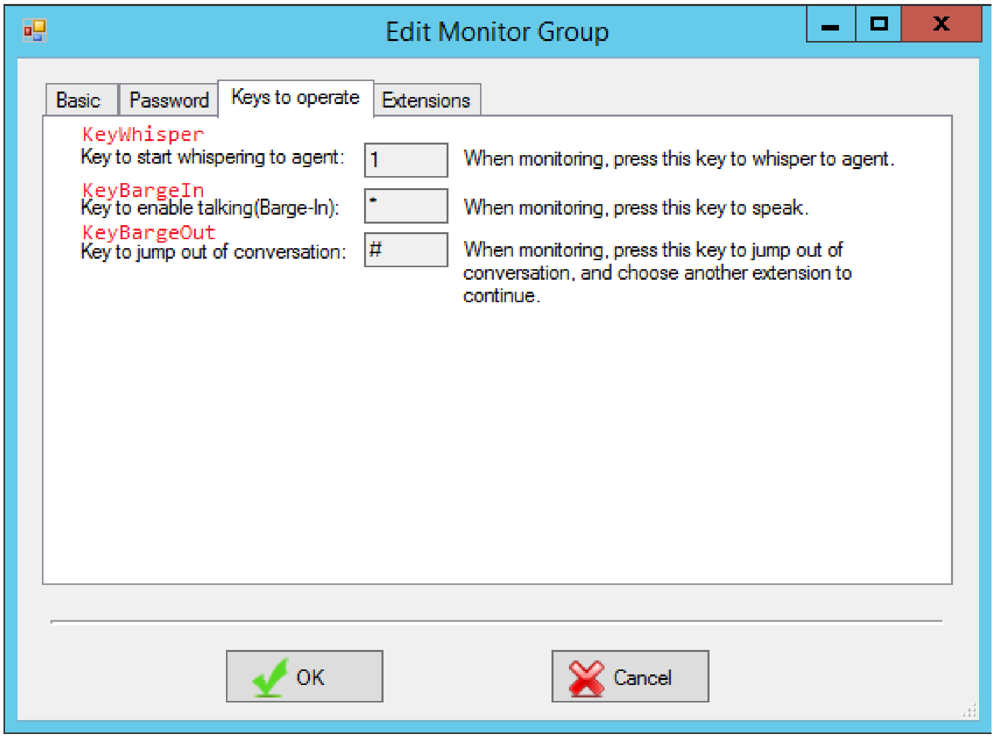
ExtenAll: if monitor all extensions

Extensions: List of extensions

ModTag: 1 = changed by GUI, not taken effect by PBX service


Edit Monitor Group
Basic
Password
Keys to operate
Extensions
Name
Name:
Any name for this group. Must be unique in whole system.
Number
Number:
Special number for supervisor to call. If it is blank, you can still route call to monitor group in dialplan.
OK
Cancel


Edit Monitor Group
Basic
Password
Keys to operate
Extensions
Password prompt is the sound to ask caller to input the password. Sample sound would be like "Please input password".
Password Prompt
Password Prompt:
C:\SIPPBXv3\audio\Please-enter-password.wav
Browse...
Password
Password:
Leaving it blank will ignor the password checking.
OK
Cancel



9.10 cfg_agents

Name: The name of the agent

Code: The agent code, digits only, must be unique. For example, 72000, 2100, 401

Password: The password for agent to login

RecordCall: 1 = record this agent's call, 0 = not record

AtExten: Internal data, to record the extension number that the agent is logging in on

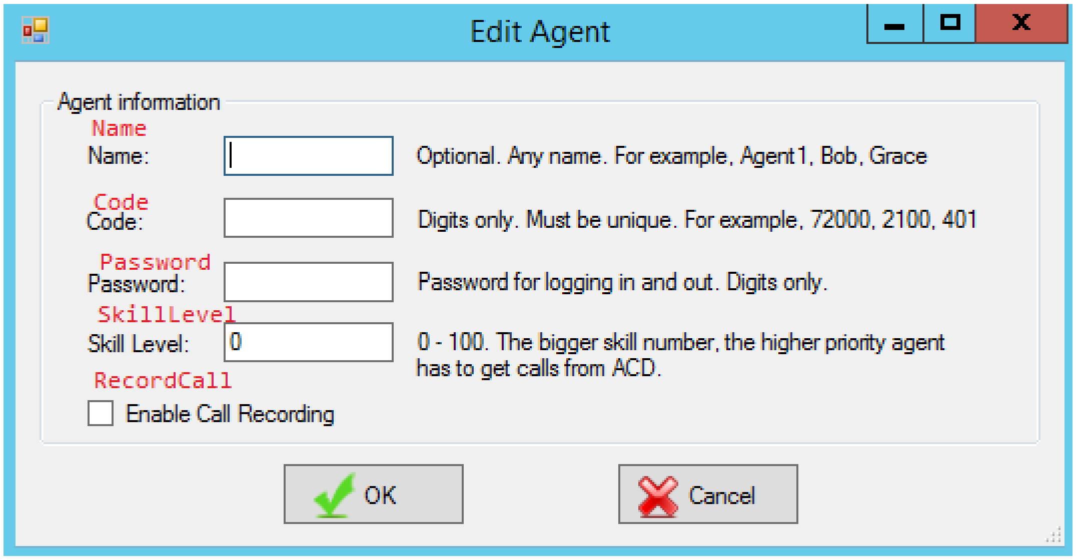
LogInTime: Internal data, to record login time

LogOutTime: Internal data, to record logout time

ModTag: 1 = changed by GUI, not taken effect by PBX service

SkillLevel: 0-100, the bigger skill number, the higher priority to get the calls from ACD

Paused: not used or applied



9.11 cfg_ivrsubitems

DTMFStr: the DTMF string to active submenu

IVRMenuAction: the action of submenu

IVRMenuSoundFile: the sound prompt file of submenu

IVRMenuTransferTo: the transfer destination according to the action

BelongTo: parent menu’s name, in the cfg_ivrs.

9.12 cfg_ivrs

MenuName: The name of IVR menu

Action: not used

MenuSound: the prompt of this ivr menu

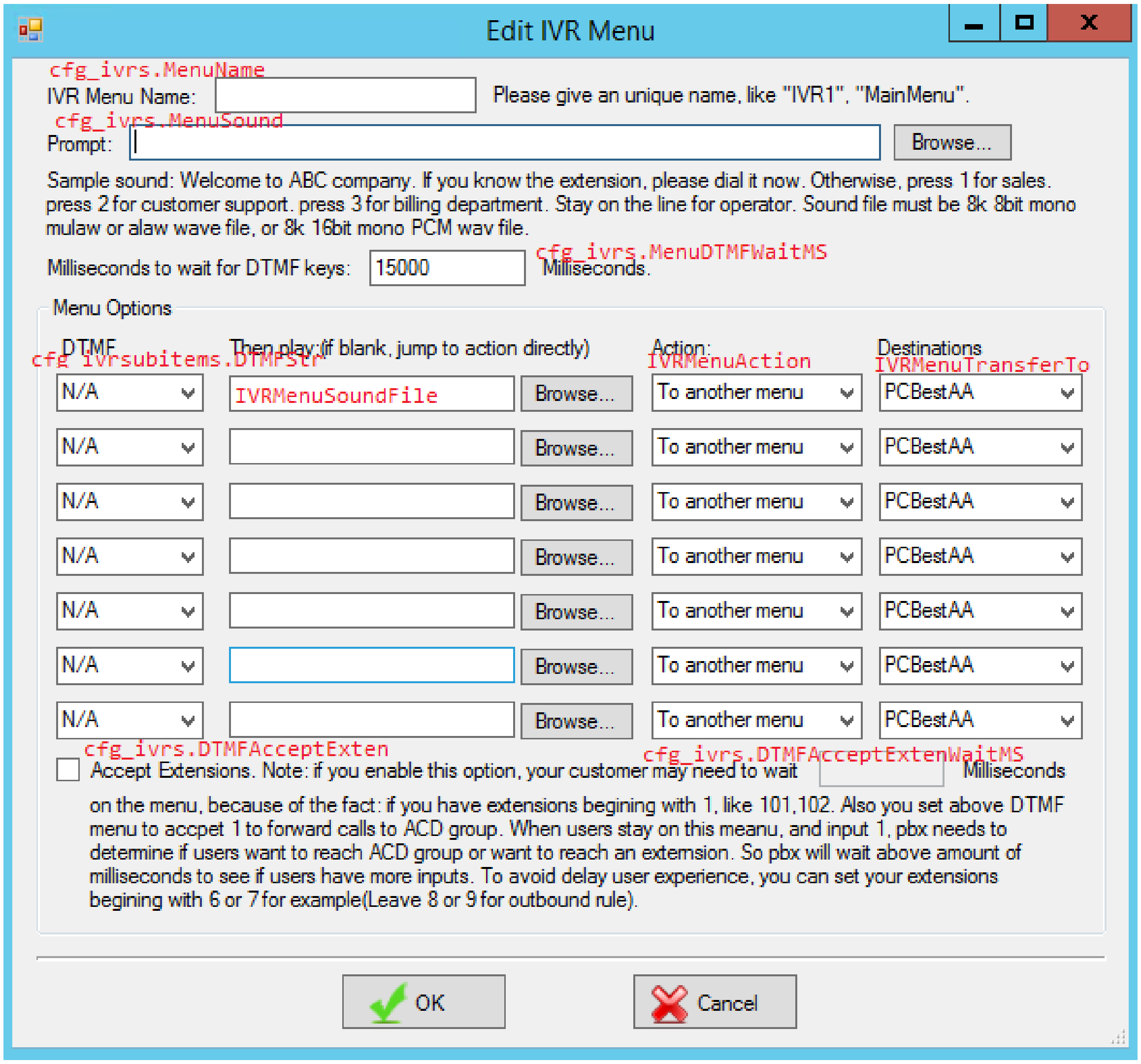
TransferTo: not used

MenuDTMFWaitMS: how many milliseconds to wait

DTMFAcceptExtenWaitMS: waiting milliseconds

DTMFAcceptExten: if accept extension number in this IVR menu

ModTag: 1 = changed by GUI, not taken effect by PBX service



9.13 cfg_autodialertasks

Name: the name of this auto dialer tasks

Enabled: 1 =enabled, 0 = disabled

TypeCode: job type code, between 1-32767. This value matches the field ‘Type’ of auto_dialer_jobs table. Give a unique value for each task.

SIPAcct: The sip account to be used for outbound call

DialPlan: The dialplan to run after the call is connected

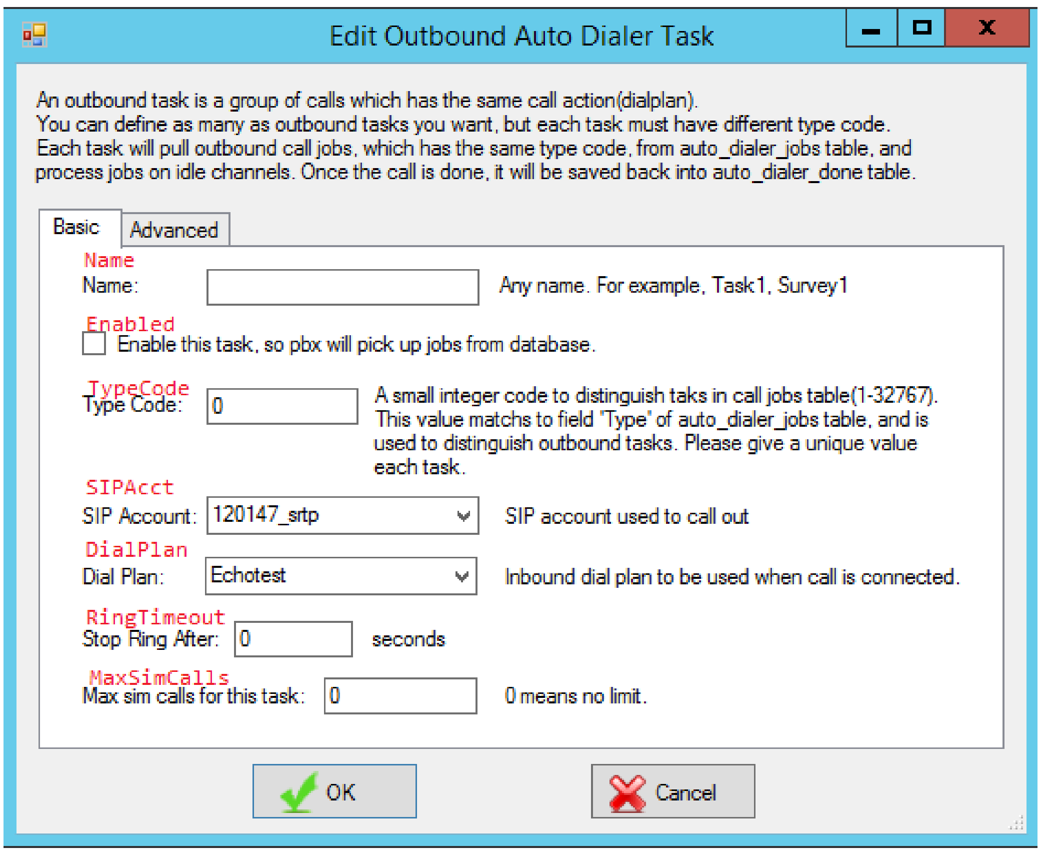
RingTimeout: in seconds for ringing

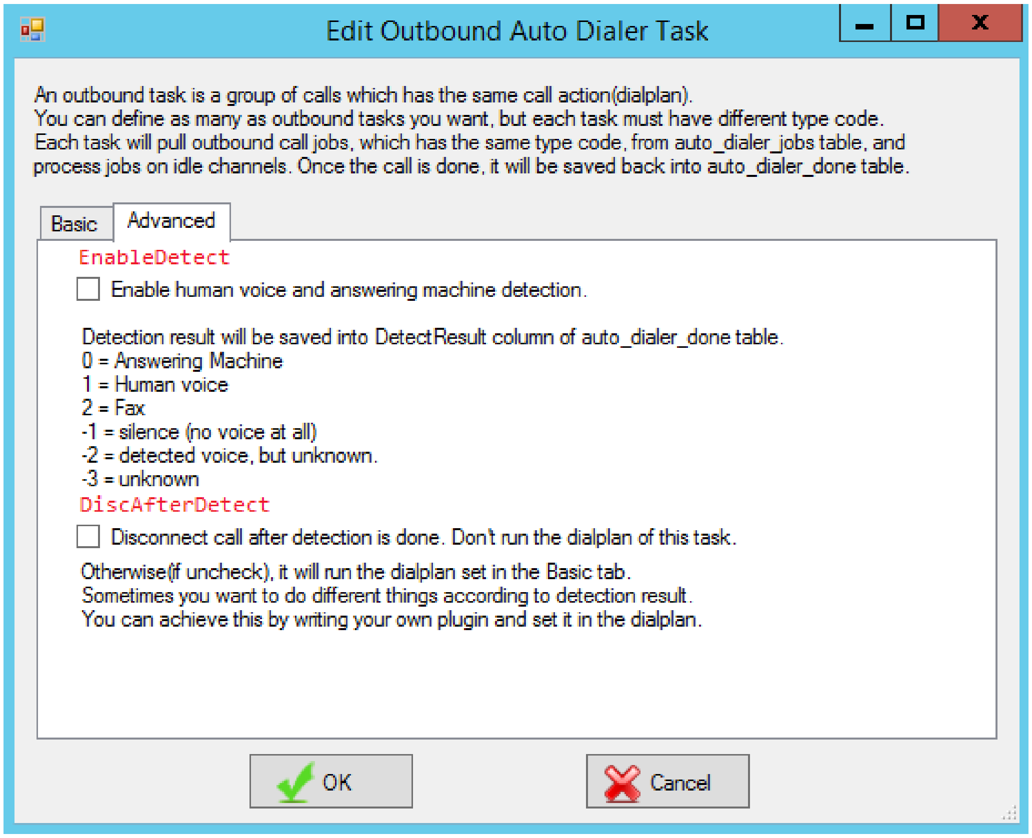
MaxSimCalls: How many calls simultaneously for this task

ModTag: 1 = changed by GUI, not taken effect by PBX service

EnableDetect: Enable human voice and answering machine detection

DiscAfterDetect: Disconnect call after detection





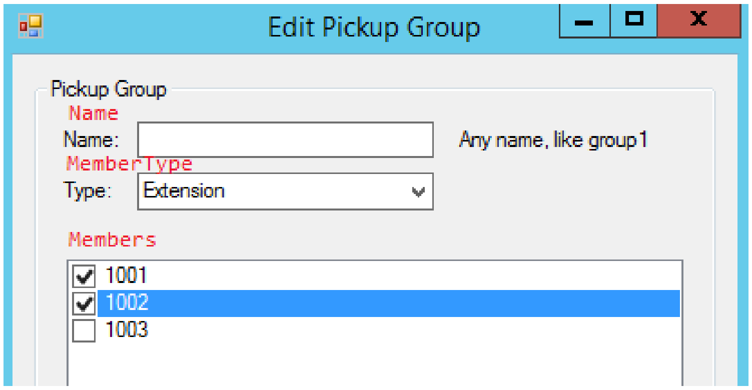
9.14 cfg_pickupgroups

Name: the name of this pickup group

MemberType: 0 = extension, 1 = agent

Members: a list of extensions or agents

ModTag: 1 = changed by GUI, not taken effect by PBX service



9.15 cfg_conferencerooms

Name: The name of this conference room

ModTag: 1 = changed by GUI, not taken effect by PBX service. 0 = already read into PBX service

MaxCallNum: Max number of users allowed in this conference room

JoinPrompt: The prompt when user join into the conference

LeavePrompt: The prompt when user leave the conference

MOHDir: The directory for Music on hold sound files

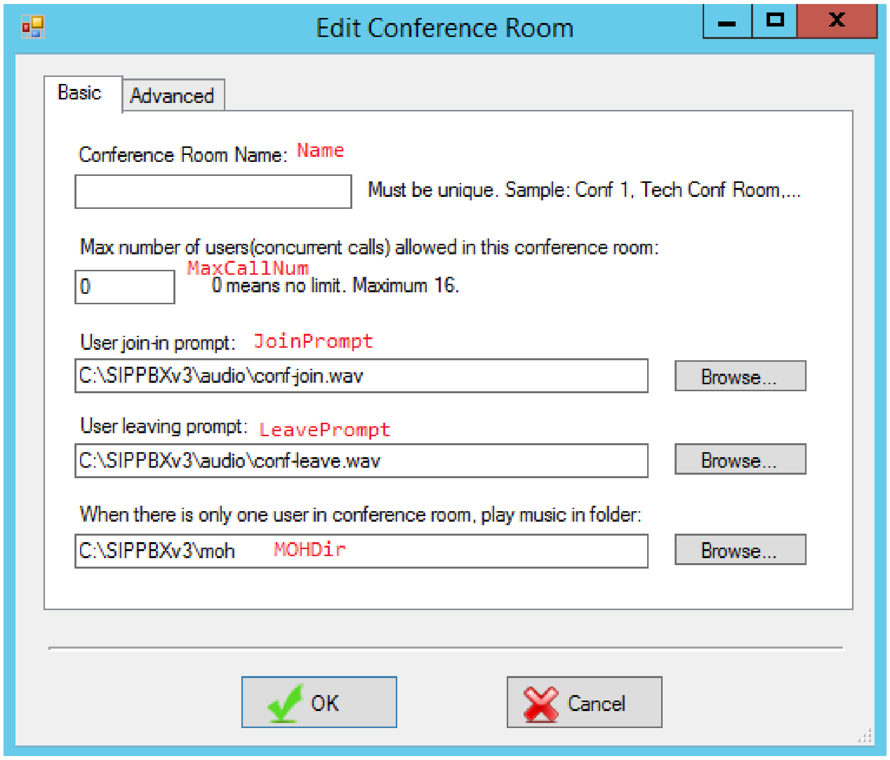
DiscCall: Disconnect the last user when others have left

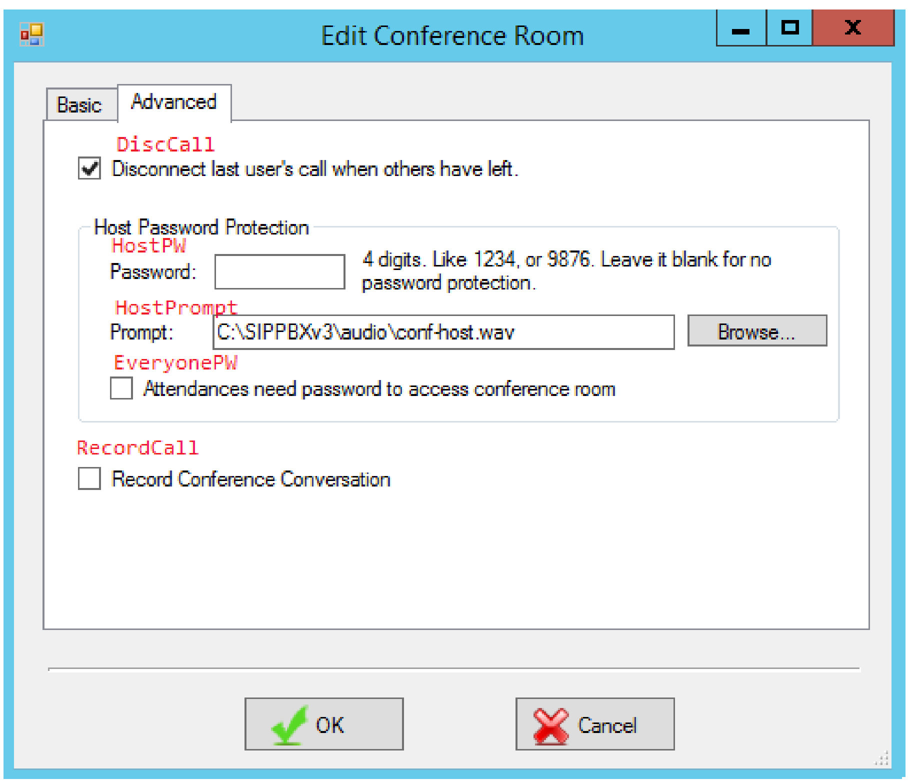
HostPW: Conference host's password

HostPrompt: Conference host's prompt

RecordCall: 1 = record conference call, 0 = not record

EveryonePW: if everyone needs a password, 1 = yes, 0 = no


Edit Conference Room
Basic
Advanced
Conference Room Name:
Name
Must be unique. Sample: Conf 1, Tech Conf Room,...
Max number of users(concurrent calls) allowed in this conference room:
MaxCallNum
0
0 means no limit. Maximum 16.
User join-in prompt:
JoinPrompt
C:\SIPPBXv3\audio\conf-join.wav
Browse...
User leaving prompt:
LeavePrompt
C:\SIPPBXv3\audio\conf-leave.wav
Browse...
When there is only one user in conference room, play music in folder:
C:\SIPPBXv3\moh
MOHDir
Browse...
OK
Cancel


Edit Conference Room
Basic
Advanced
DiscCall
Disconnect last user's call when others have left.
Host Password Protection
HostPW
Password:
4 digits. Like 1234, or 9876. Leave it blank for no password protection.
HostPrompt
Prompt:
C:\SIPPBXv3\audio\conf-host.wav
Browse...
EveryonePW
Attendances need password to access conference room
RecordCall
Record Conference Conversation
OK
Cancel

9.16 cfg_calllimit

DialPlan: Can be dialplan name, or extension name, or sip account name.

Seconds: How many seconds to allow

RoundupSeconds: Roundup seconds