PC Best Networks SIP PBX Reference
Setup and Development Guide
(For V2 and V3)

support@pcbest.net

Index:

1 Introduction

PC Best Networks provides NO.1 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.

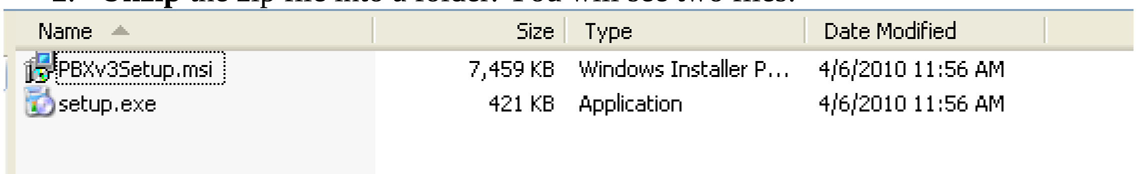
REQUIREMENT:

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

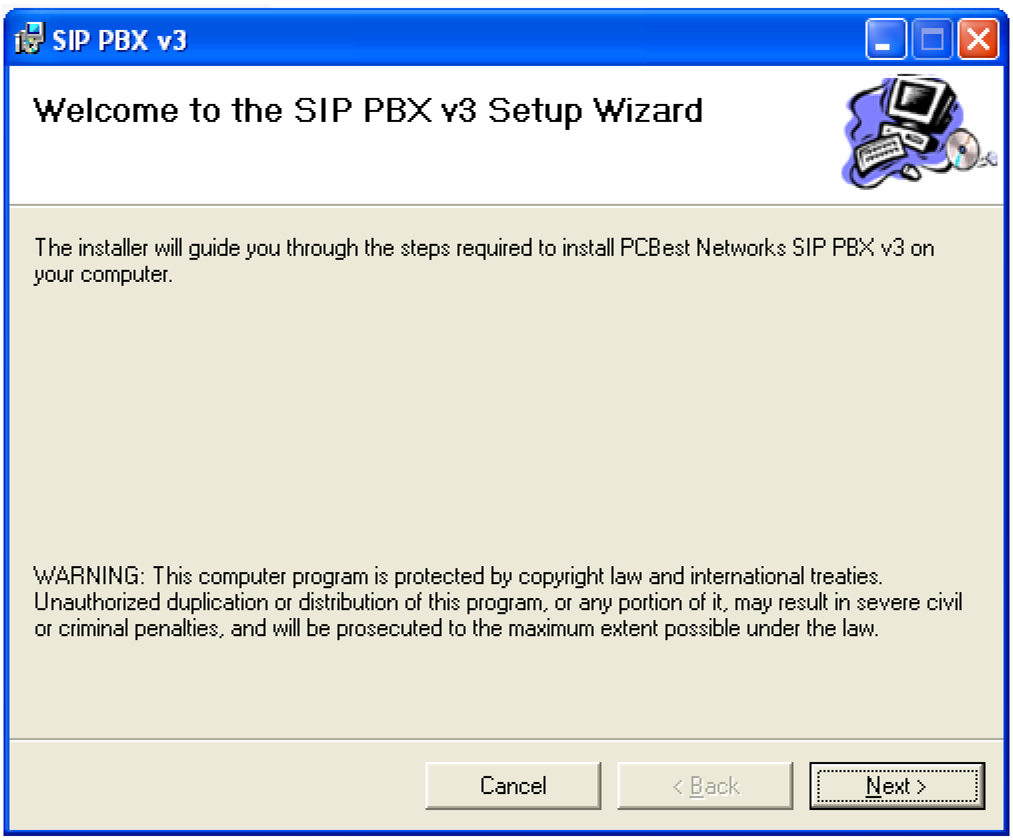
Our contact information for support:
Email: support@pcbest.net
Toll Free(USA & Canada): 1-888-733-6620
Local and International: 1-613-800-2202

2 Installing PBX

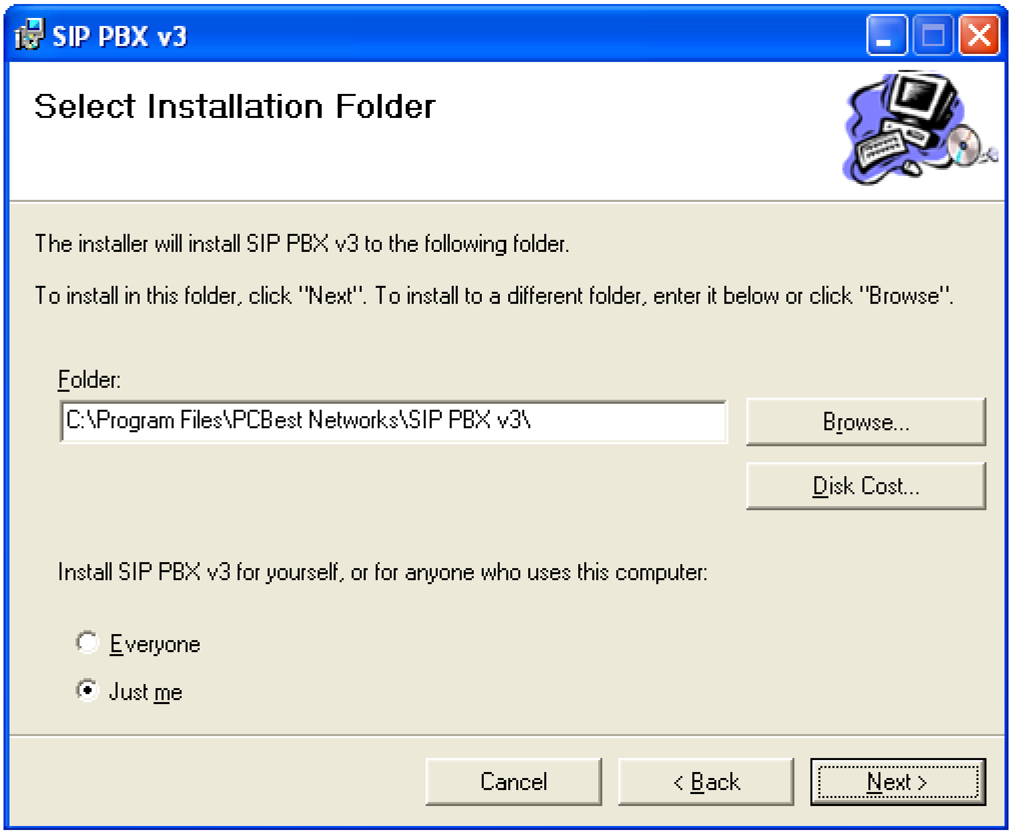
1. **Download** PC Best PBX v3 from this page: http://www.pcbest.net/sip_pbx.php
2. **Unzip** the zip file into a folder. You will see two files:



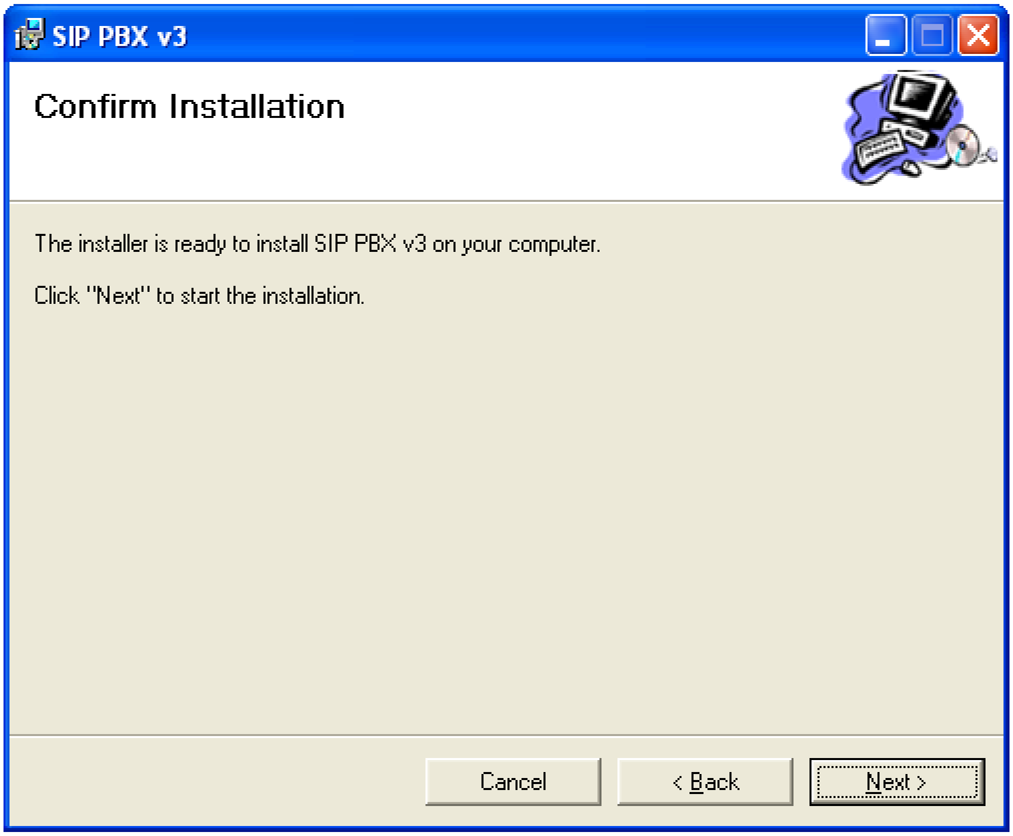
3. *Run* **setup**.exe. *Under Window7 or 2008, please right click setup.exe, and* ***run as administrator***.



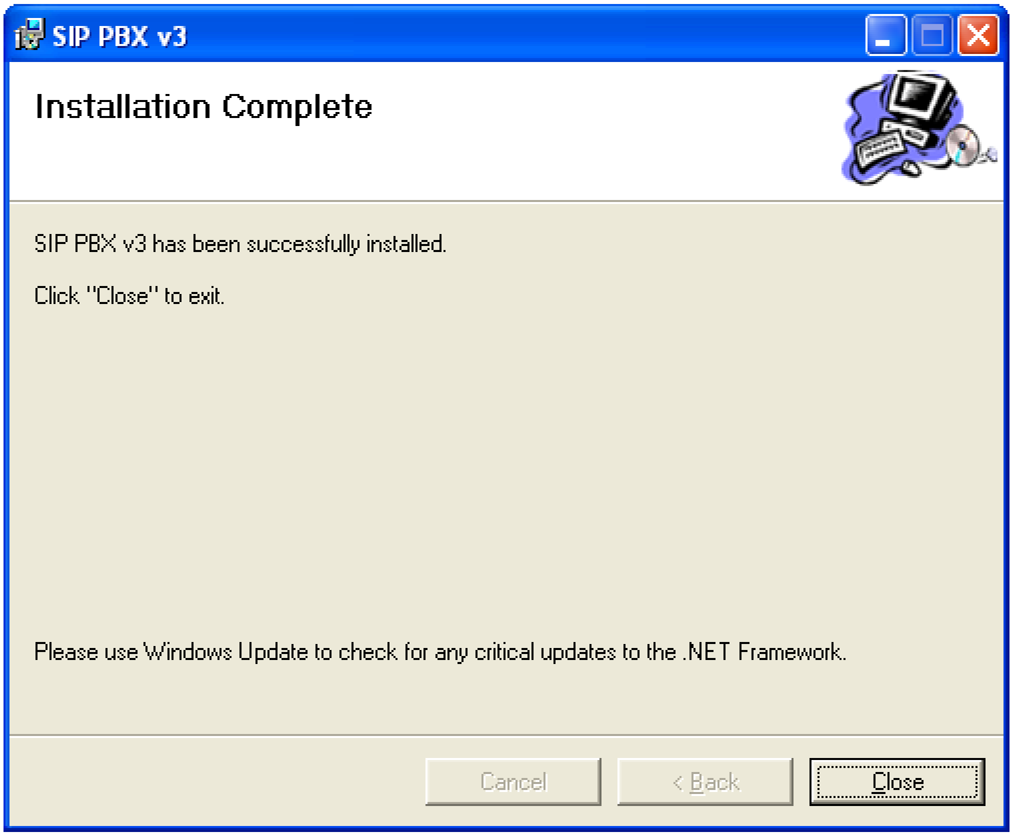
Click next.



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



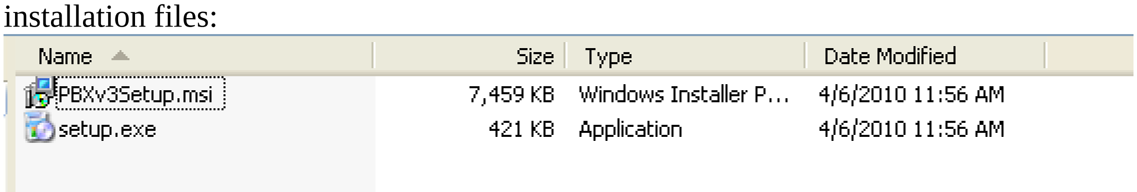
Then confirm the installation.



It is done.

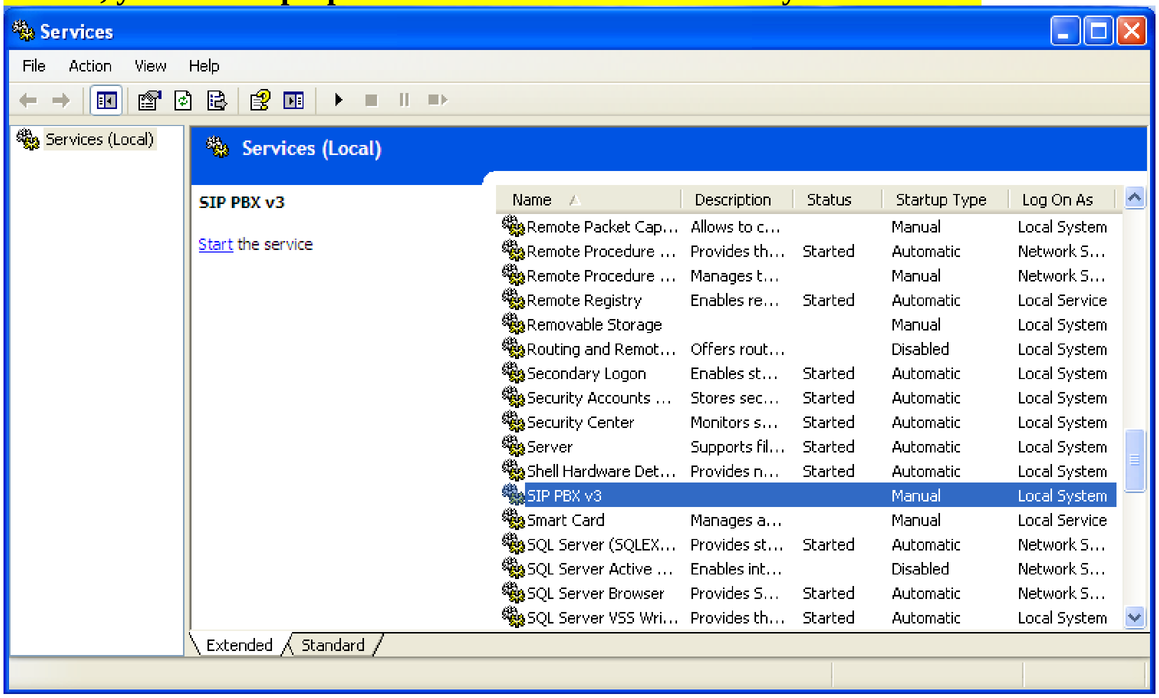
Some customers reported they encountered error 2869, and installation cannot be completed. For this error, it is because of the errors in your Windows registry. There are several ways to solve it(choose one of them):

a. The 2869 error is a common error regarding MSI files when they are executed in Windows due to the user access control. A workaround for this error is as follows:
 1. Run a command line as an administrator. This can be done by clicking on the start menu and typing CMD . When the option appears, right click and select Run as Administrator.
 2. Run “msiexec /i PBXv3Setup.msi” in the command line, in the directory of two installation files:



b. Under Windows 7, 2008, or Vista, please right click setup.exe, and **run as administrator**.

For V3, you need to prepare database for PBX v3 before you can run it.



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

For V2, you don't have to setup database in order to run.
V2 is NOT a service application, so you won't see it in Service list like above picture.

4. Setup **Database**.

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

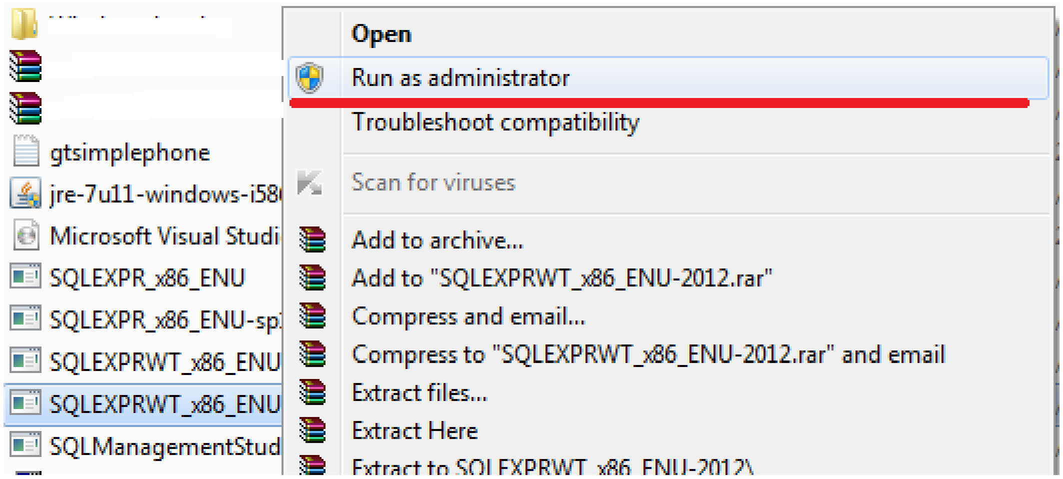
Microsoft® SQL Server® 2012 Express:
http://www.microsoft.com/en-ca/download/details.aspx?id=29062

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

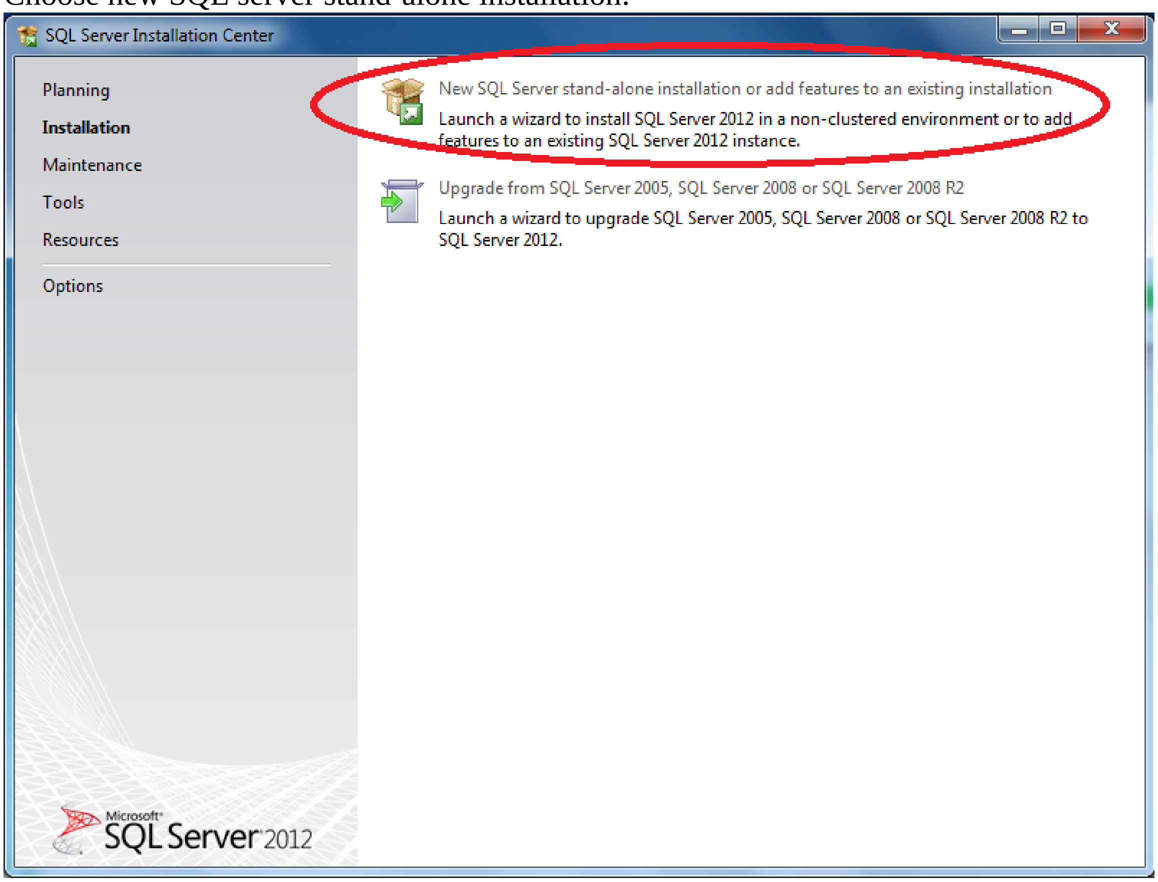
64bit OS download one of the following:

ENU\x64\SQLEXPRADV_x64_ENU.exe 1.3 GB Download
ENU\x64\SQLEXPRWT_x64_ENU.exe 669.9 MB

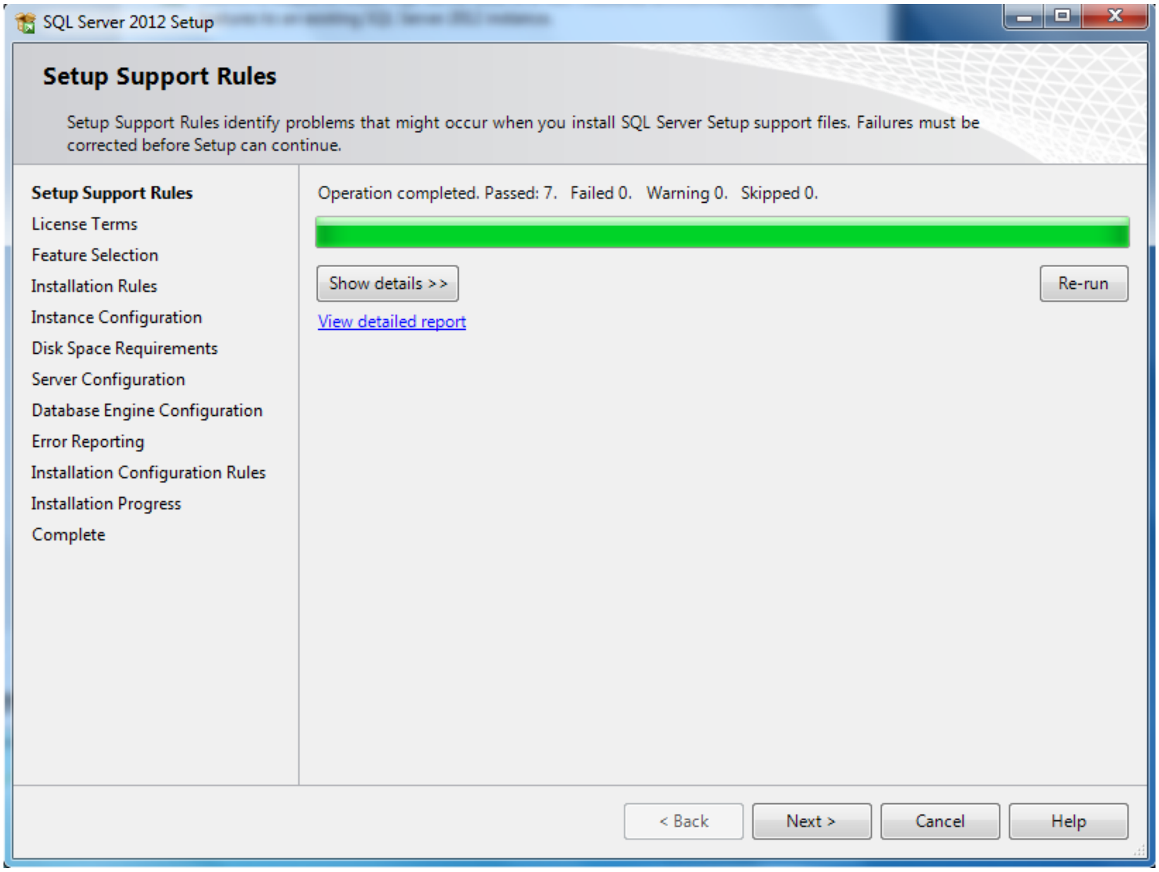
Assume we use SQL Server 2012 Express here. It is free to download from website. We download SQL Server 2012 with tool, which has management studio. Right click on SQLEXPRWT_x86_ENU.exe for 32bit Windows or SQLEXPRWT_x64_ENU.exe for 64bit Windows, and "Run as administrator":



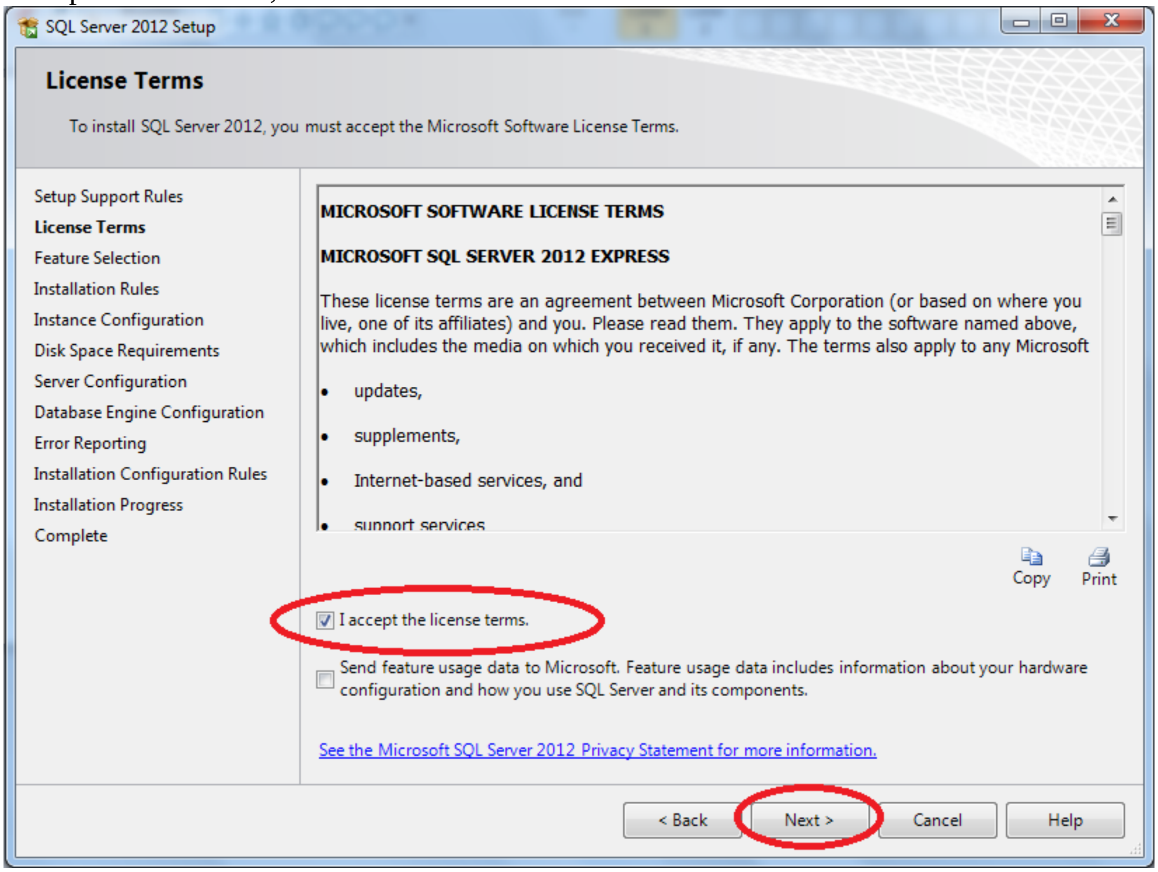
Choose new SQL server stand-alone installation:

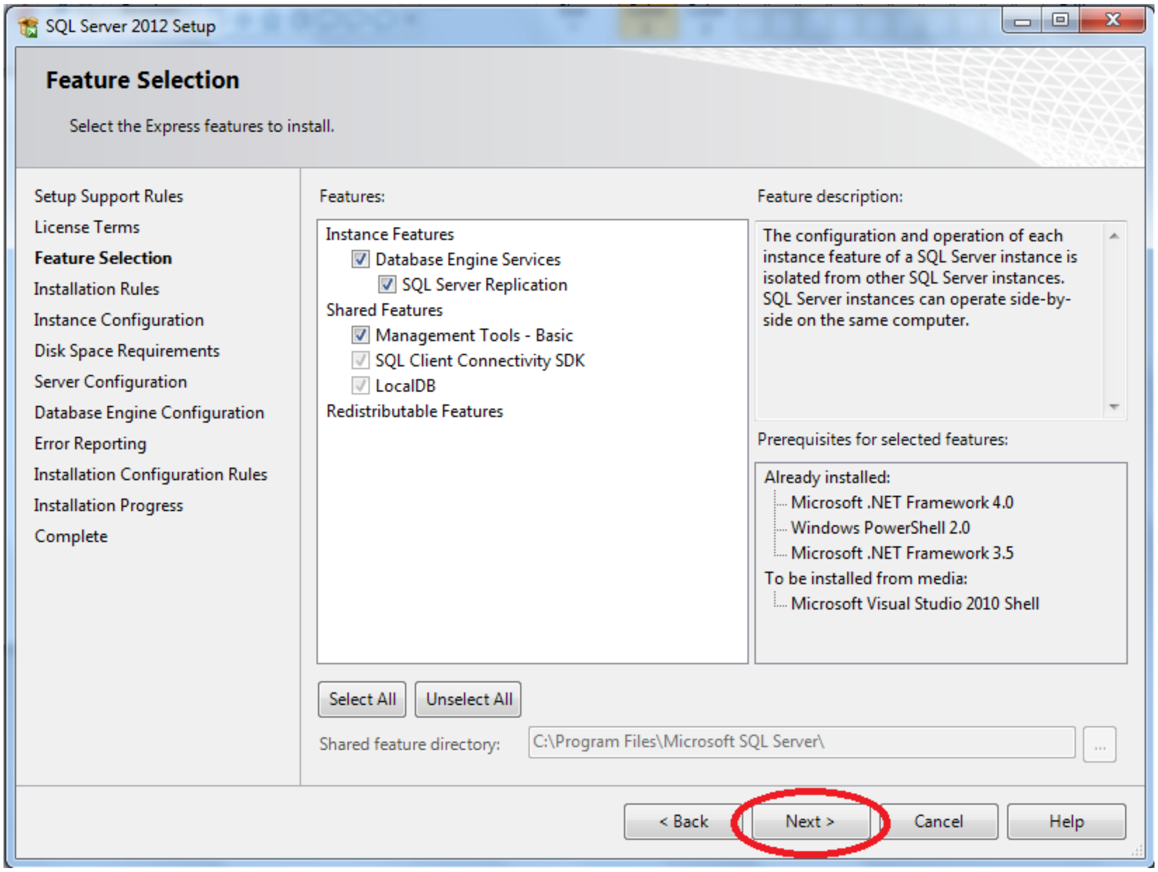


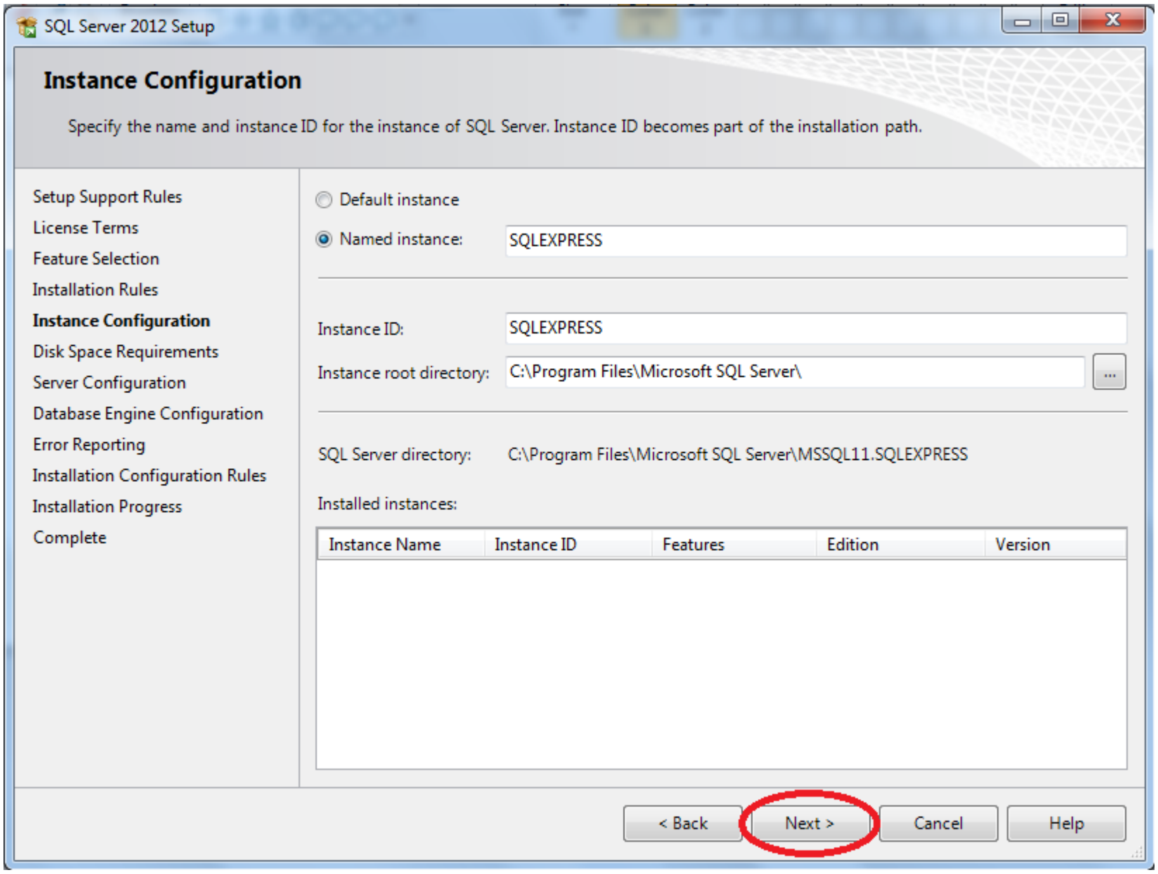
Of course, if you already have 2005, 2008, or 2008R2, you can upgrade it to 2012.
Click next:

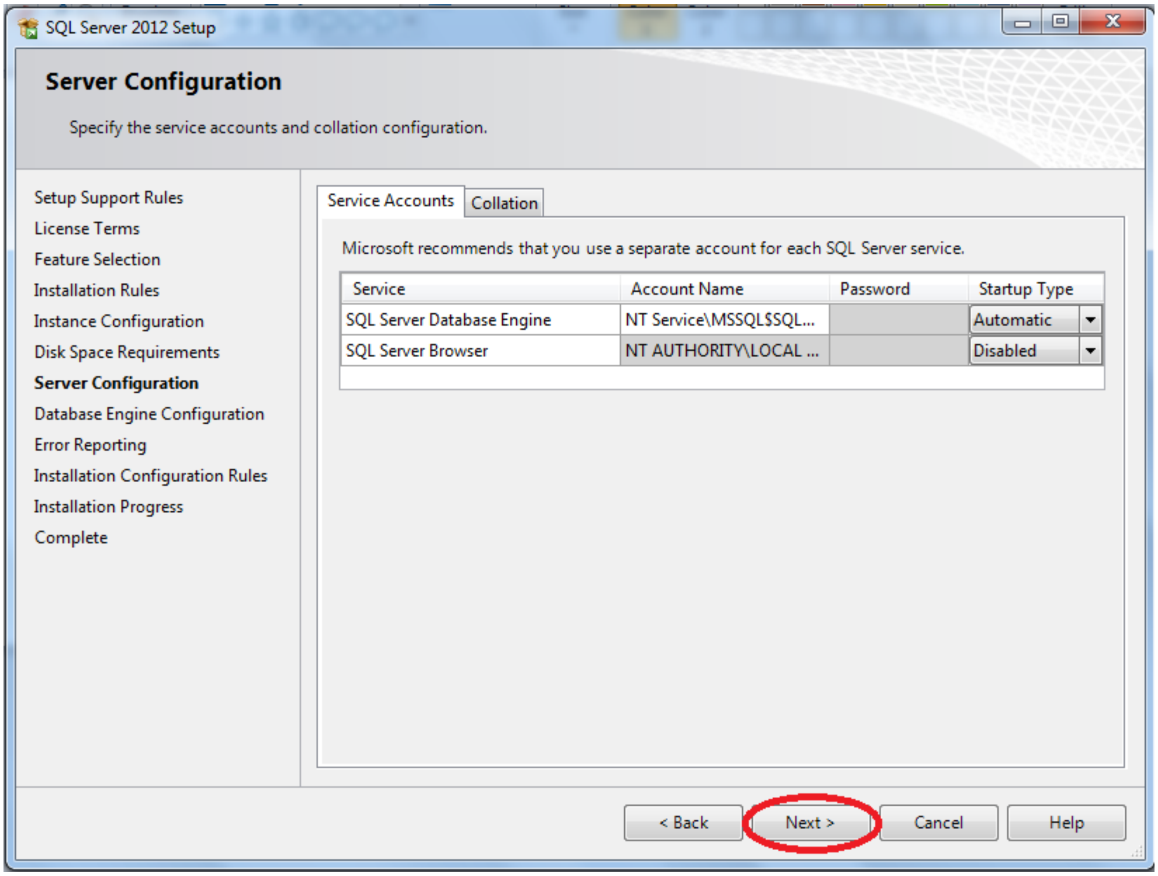


Accept license terms, and clieck Next:

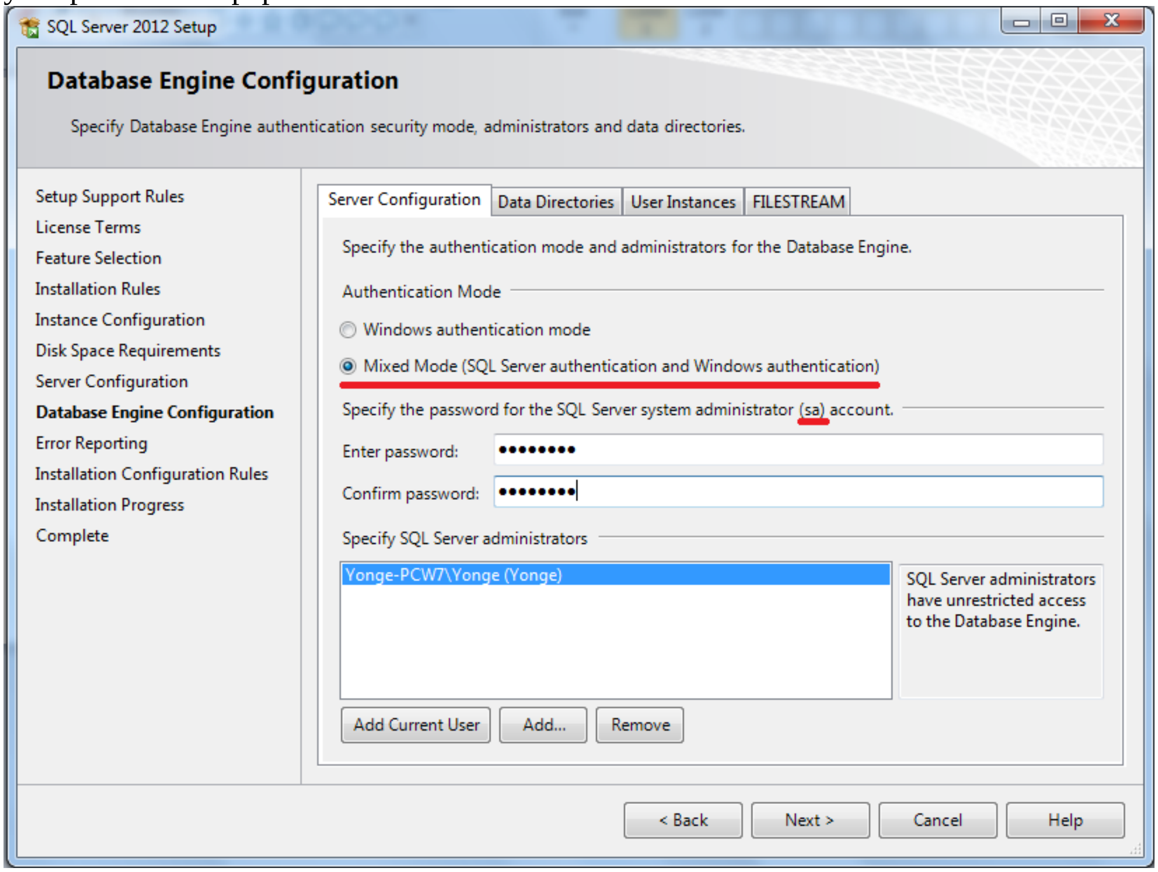


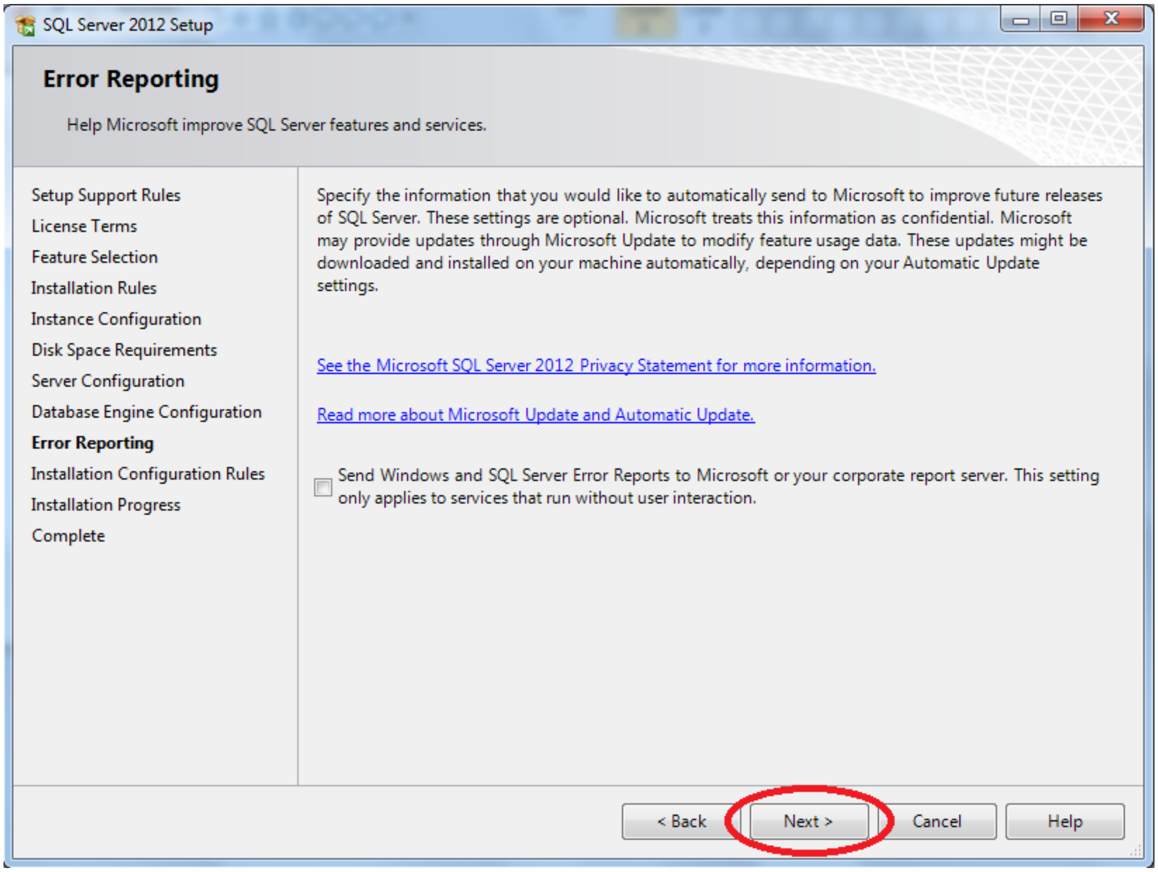

SQL Server 2012 Setup
Feature Selection
Select the Express features to install.
Setup Support Rules
License Terms
Feature Selection
Installation Rules
Instance Configuration
Disk Space Requirements
Server Configuration
Database Engine Configuration
Error Reporting
Installation Configuration Rules
Installation 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

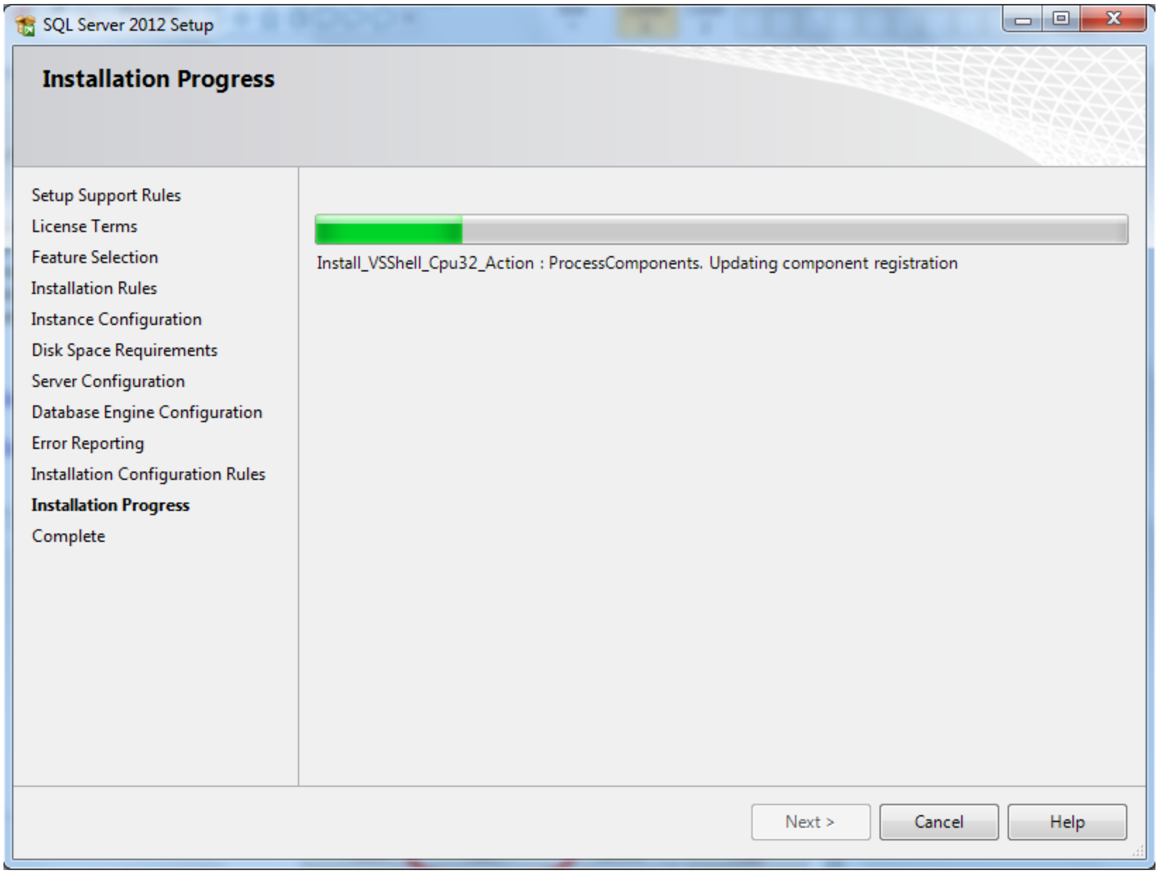

SQL Server 2012 Setup
Instance Configuration
Specify the name and instance ID for the instance of SQL Server. Instance ID becomes part of the installation path.
Setup Support Rules
License Terms
Feature Selection
Installation Rules
Instance Configuration
Disk Space Requirements
Server Configuration
Database 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

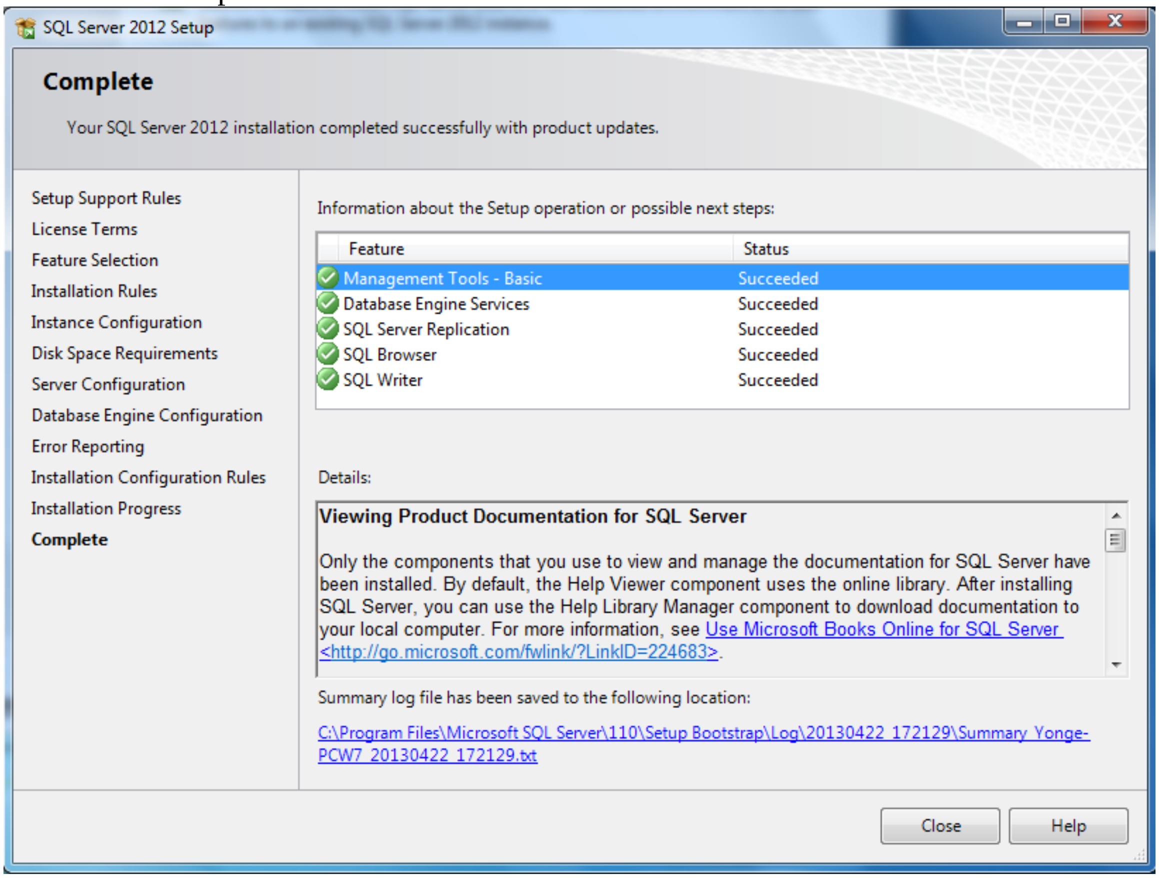
Choose Mixed Mode, and set password for account sa. NOTE: please write down your password in paper for later use.



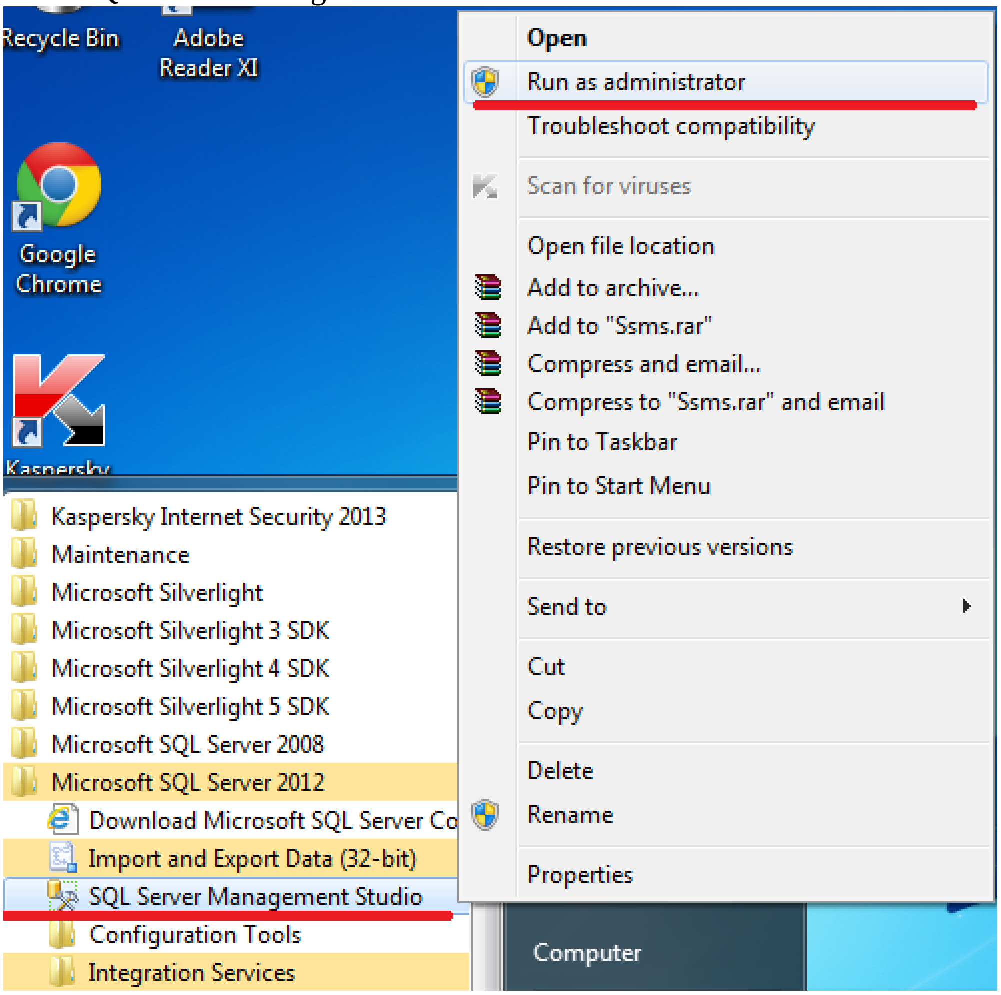

SQL Server 2012 Setup
Error Reporting
Help Microsoft improve SQL Server features and services.
Setup Support Rules
License Terms
Feature Selection
Installation Rules
Instance Configuration
Disk Space Requirements
Server Configuration
Database Engine Configuration
Error Reporting
Installation Configuration 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

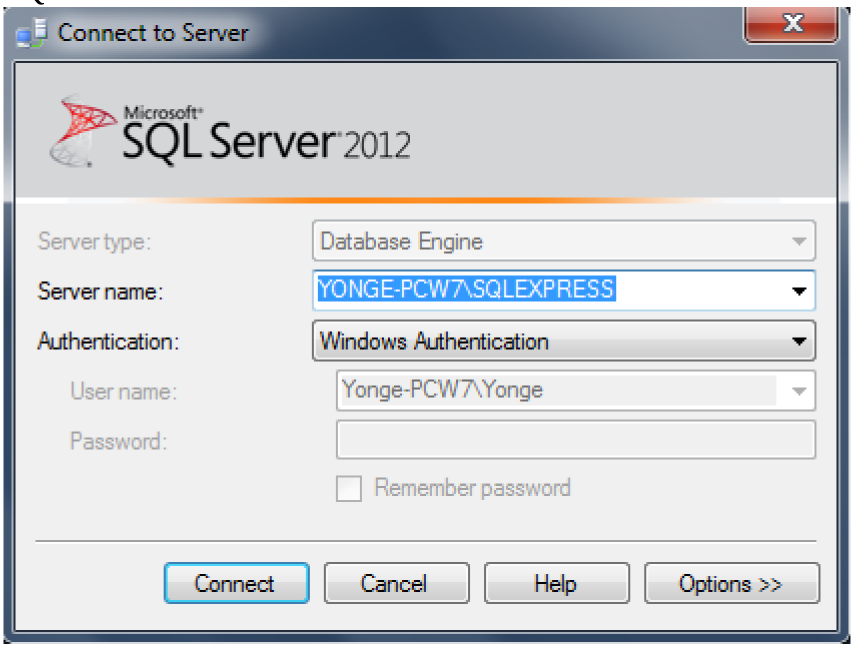
Installation Complete:



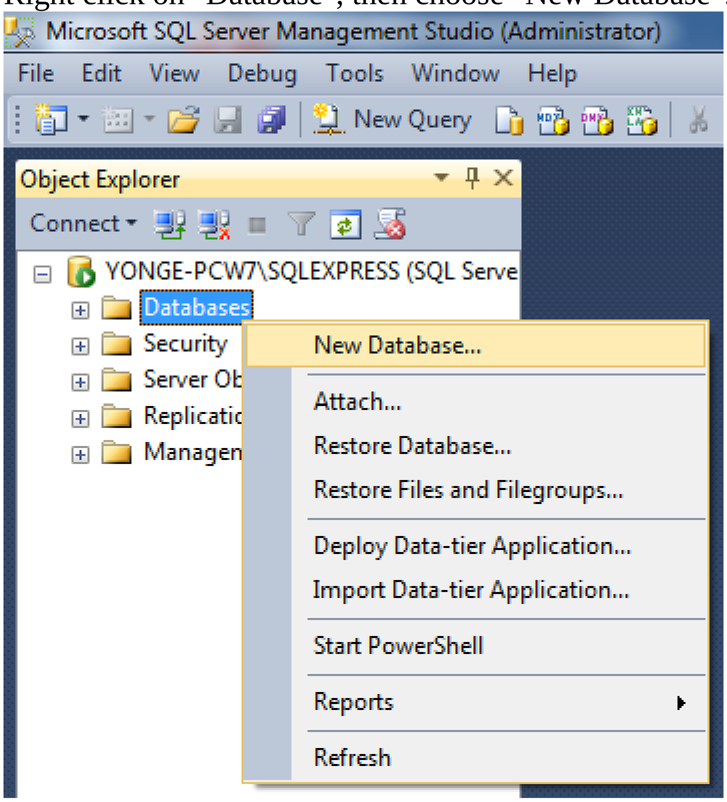
Run "SQL Server Management Studio":



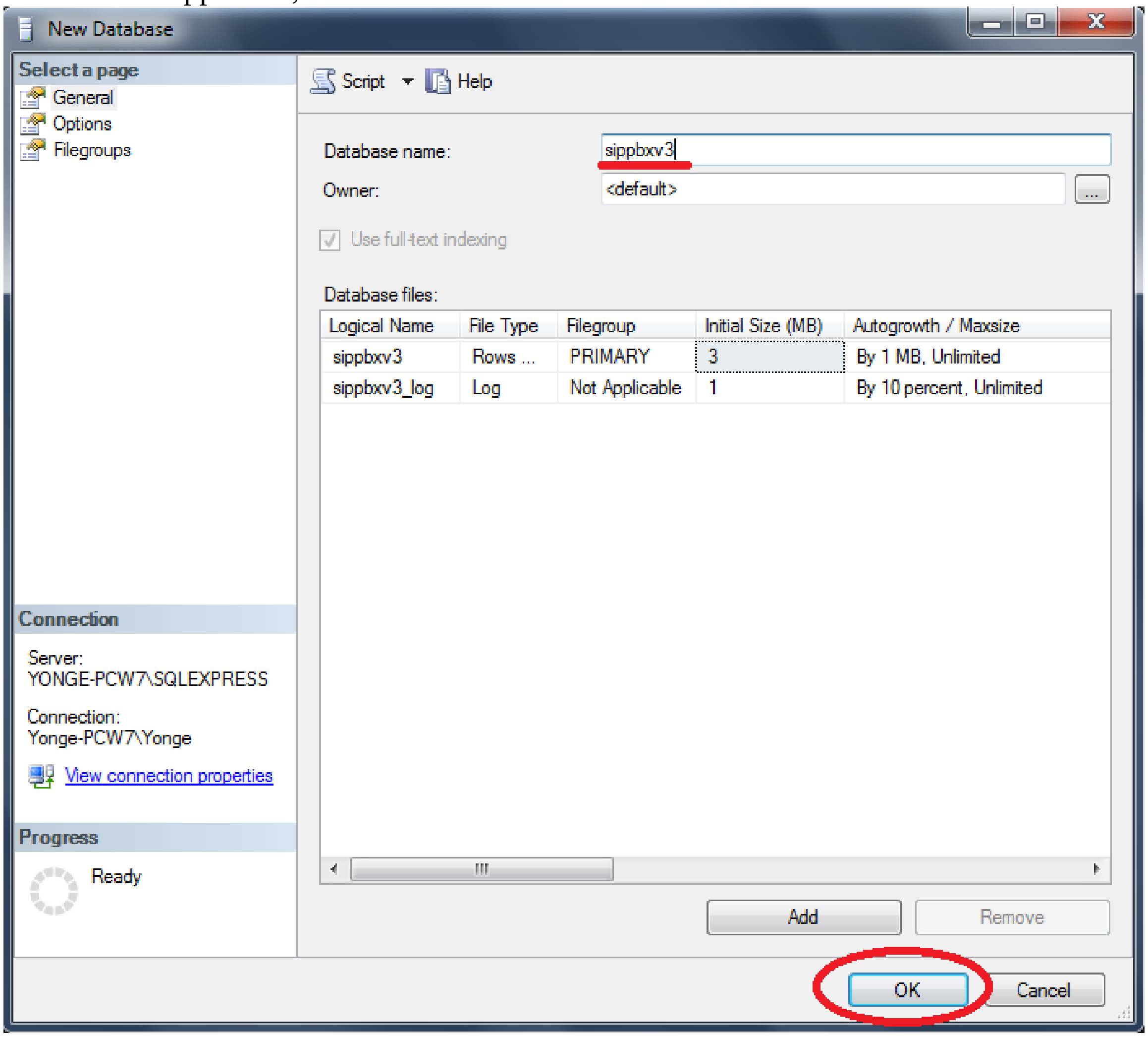
You can use "Windows Authentication" here, and click Next, or use SQL Authentication, then give username sa, password whatever you set when installing SQL server.



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



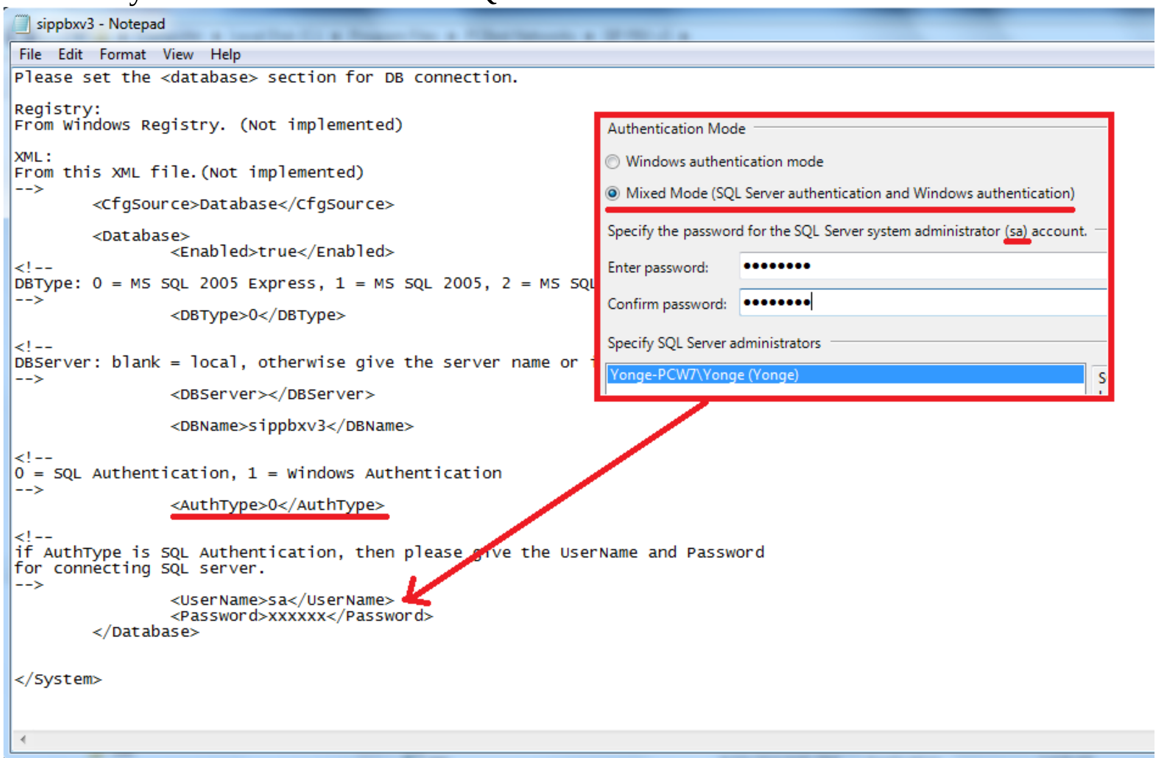
Give name "sippbxv3", and click "OK":



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

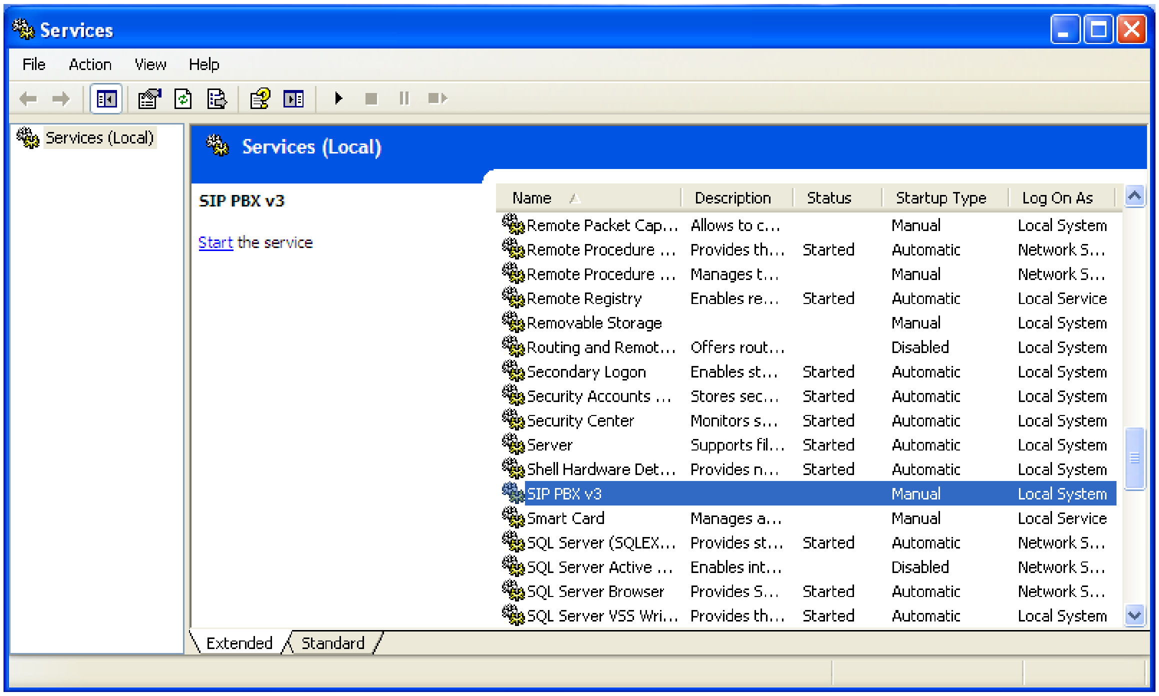
5. After the database is created, change the configuration file **sippbxv3.xml** for DB connection. The file can be found in PBX installation folder. Use Notepad or any text editor to open it. Under Windows7 or 2008, in order to change this file, you may need to run Notepad as Administrator first, then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

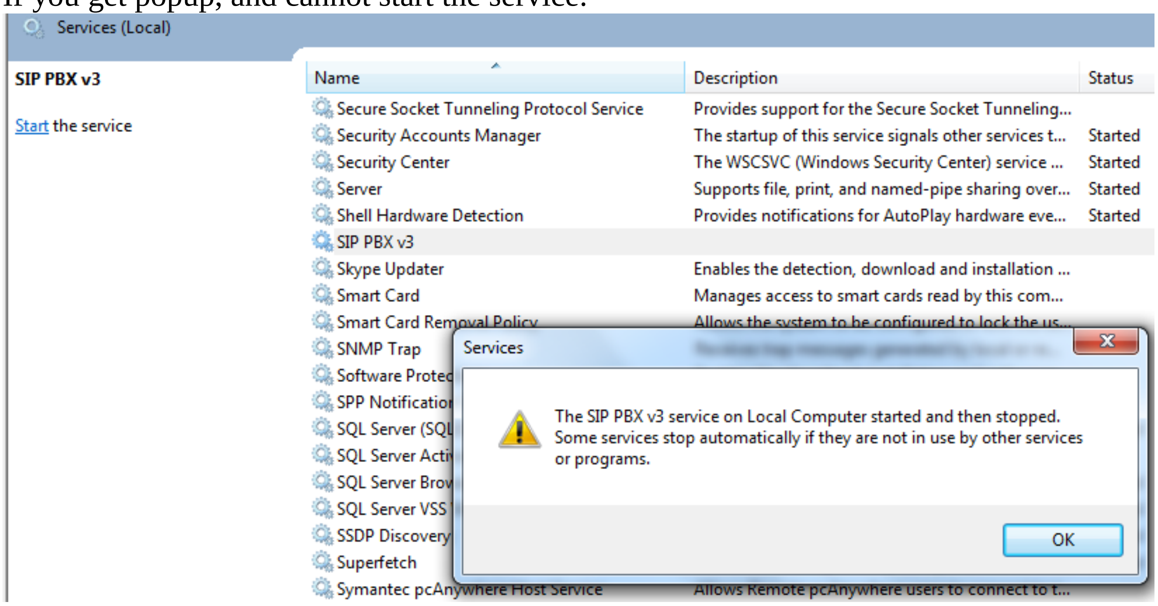


6. Start SIP PBX v3 service

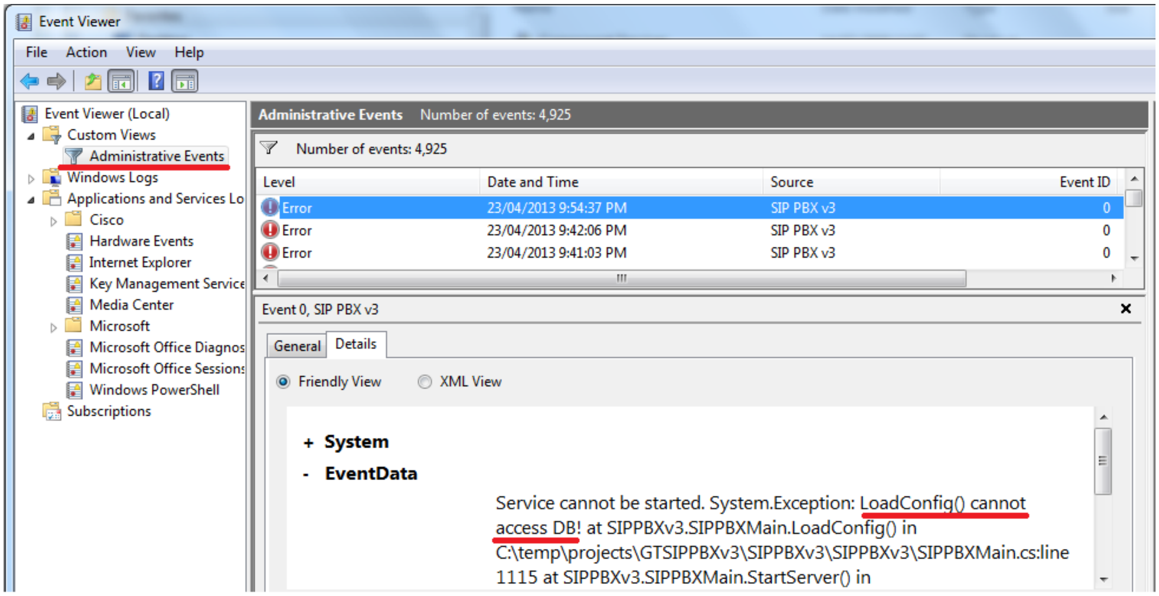
From Control Panel -> Administrator Tool -> Open Windows Services, then find SIP PBX v3 service, then click start(the triangle button):



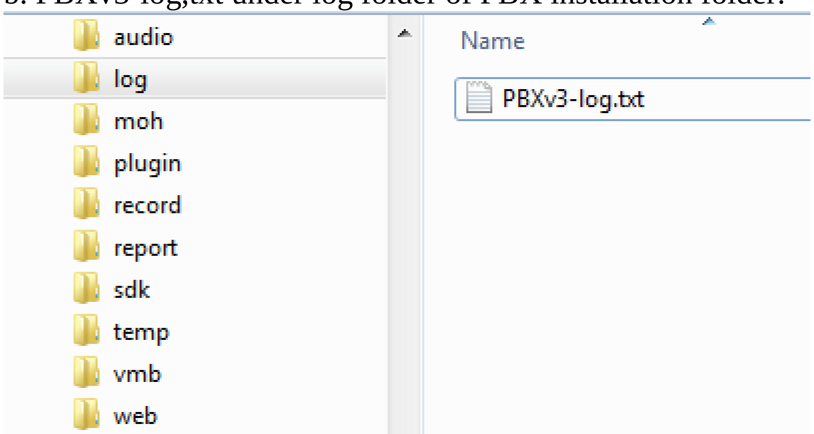
If you get popup, and cannot start the service:

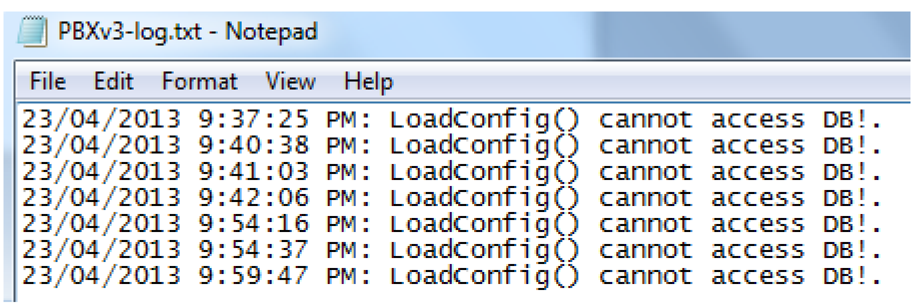


Please check:
a. Event Viewer:

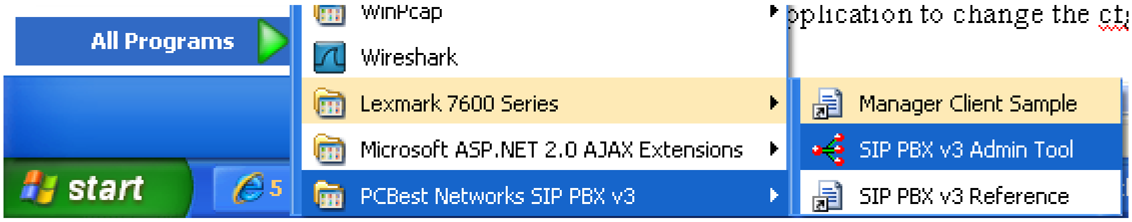


b. PBXv3-log,txt under log folder of PBX installation folder:

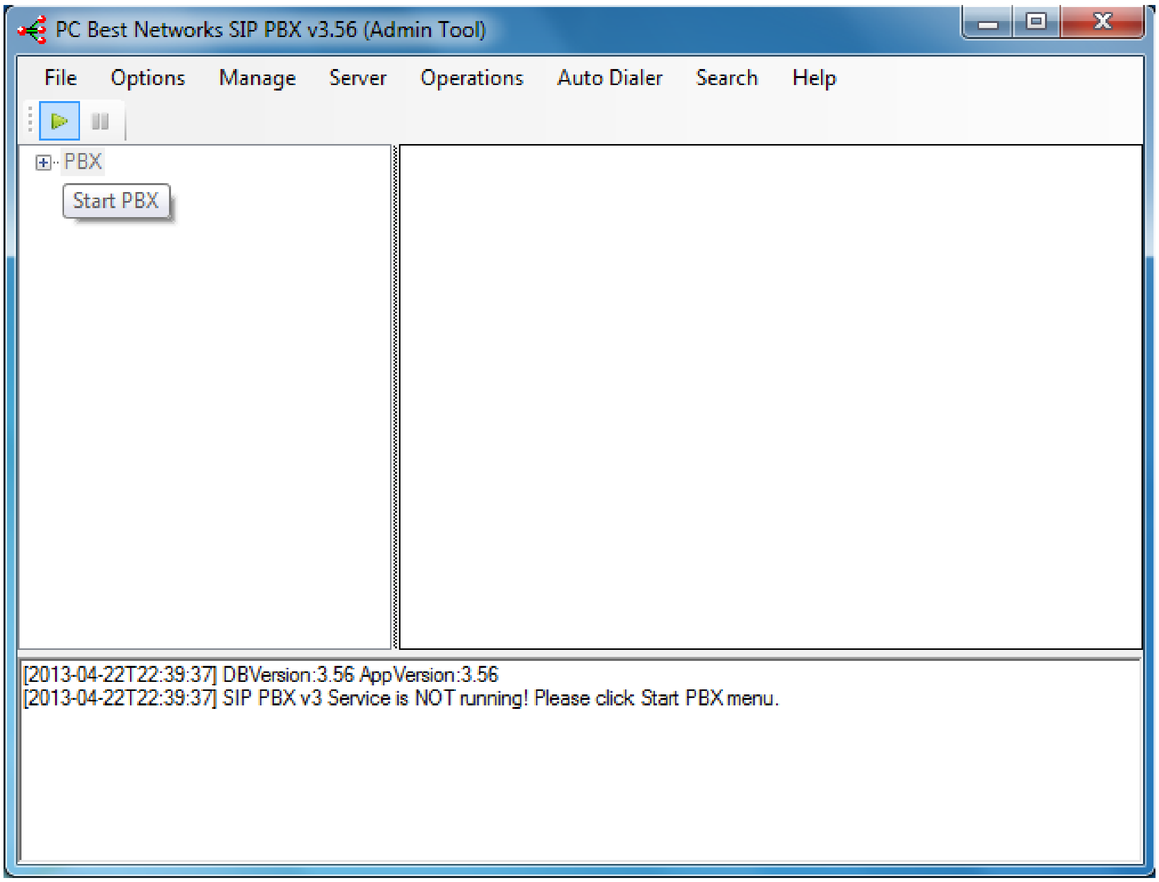




7. Run PBX v3 admin tool. **NOTE: if you are using Vista or Windows 7, you need to "Run as administrator" because admin tool needs administrator right to start or stop PBX v3 service.**



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



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

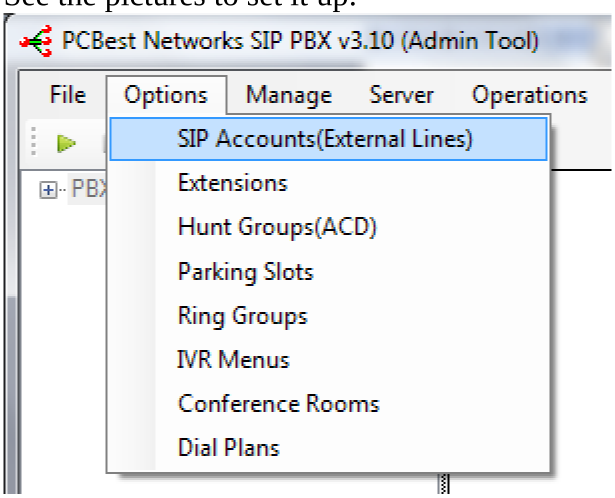
Before you start, you need to setup the following common settings for all tests.

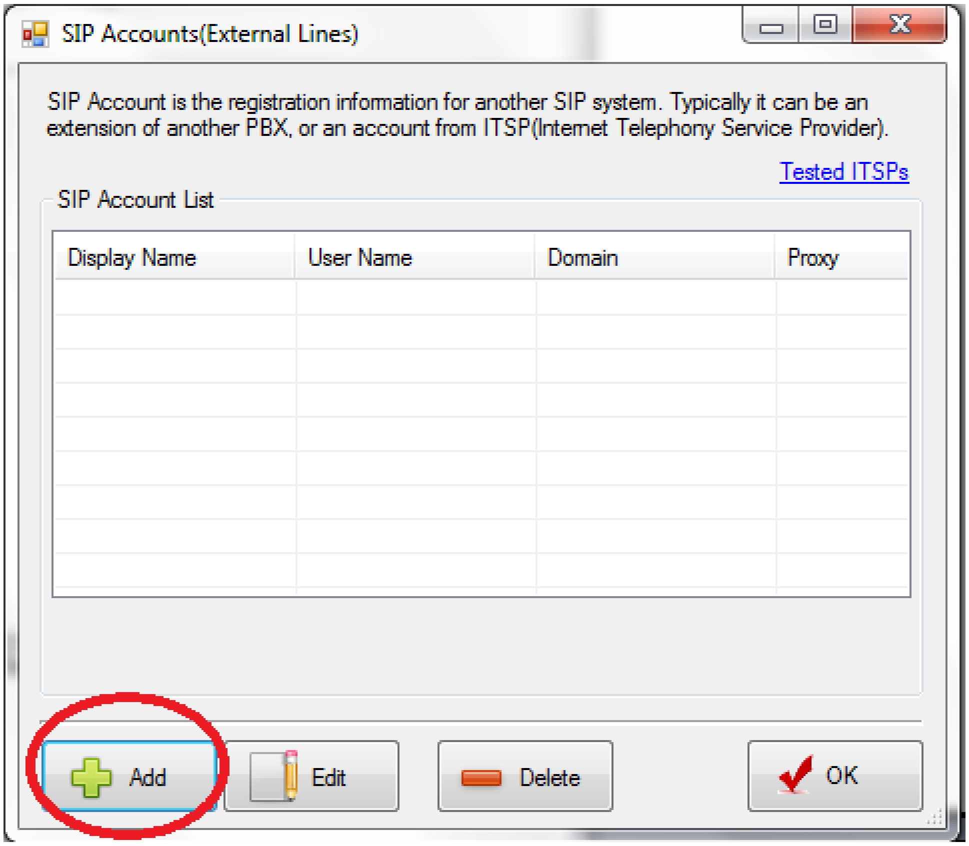
SIP Accounts(External Lines)

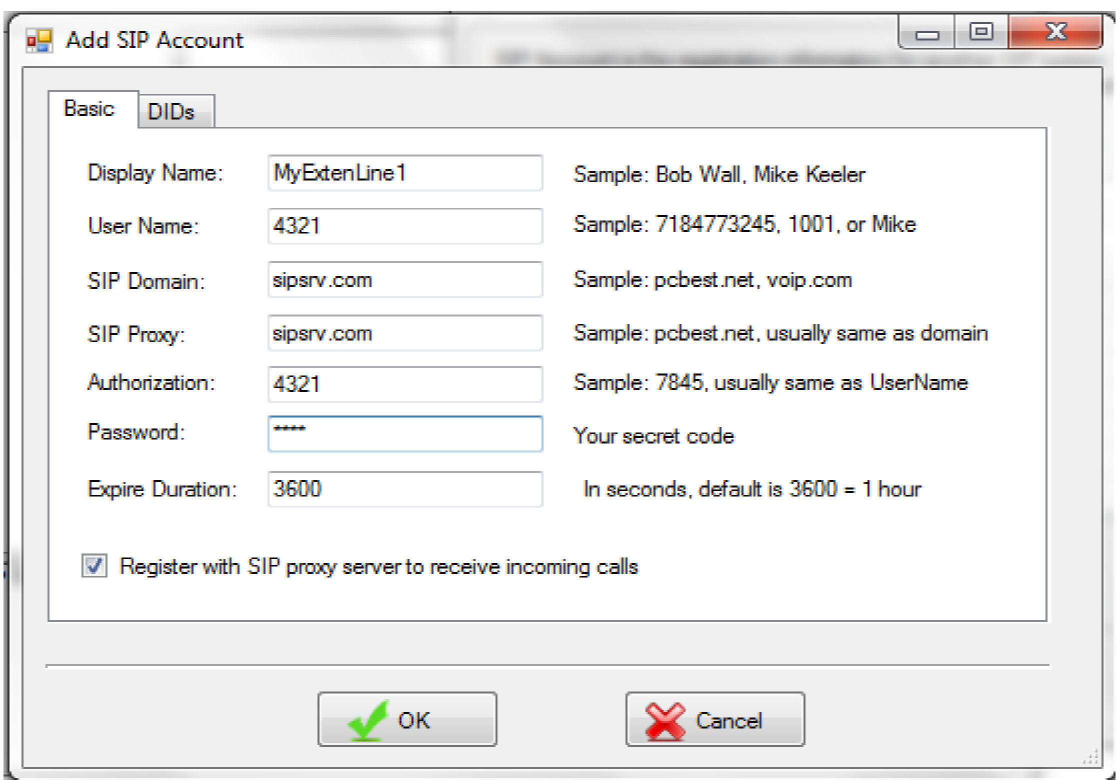
SIP Accounts are the credit info that you can use it to dial out external lines, or receive calls from out lines. For example, you can get a SIP account from ITSP(Internet Telephony Service Provider), then you can make calls to regular phone numbers, or receives calls to your DID.

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

See the pictures to set it up:

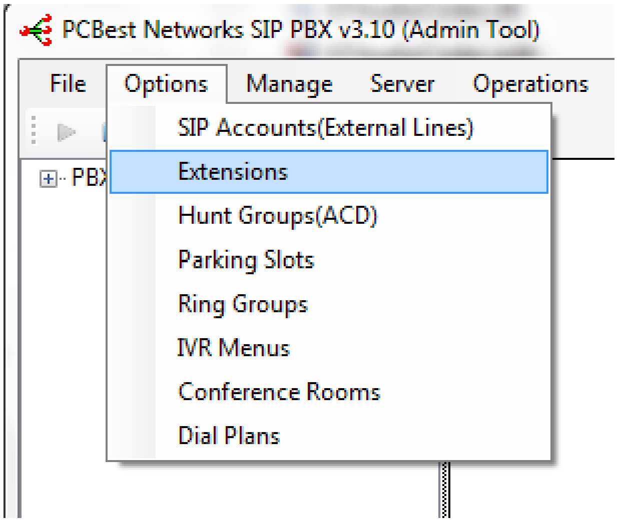


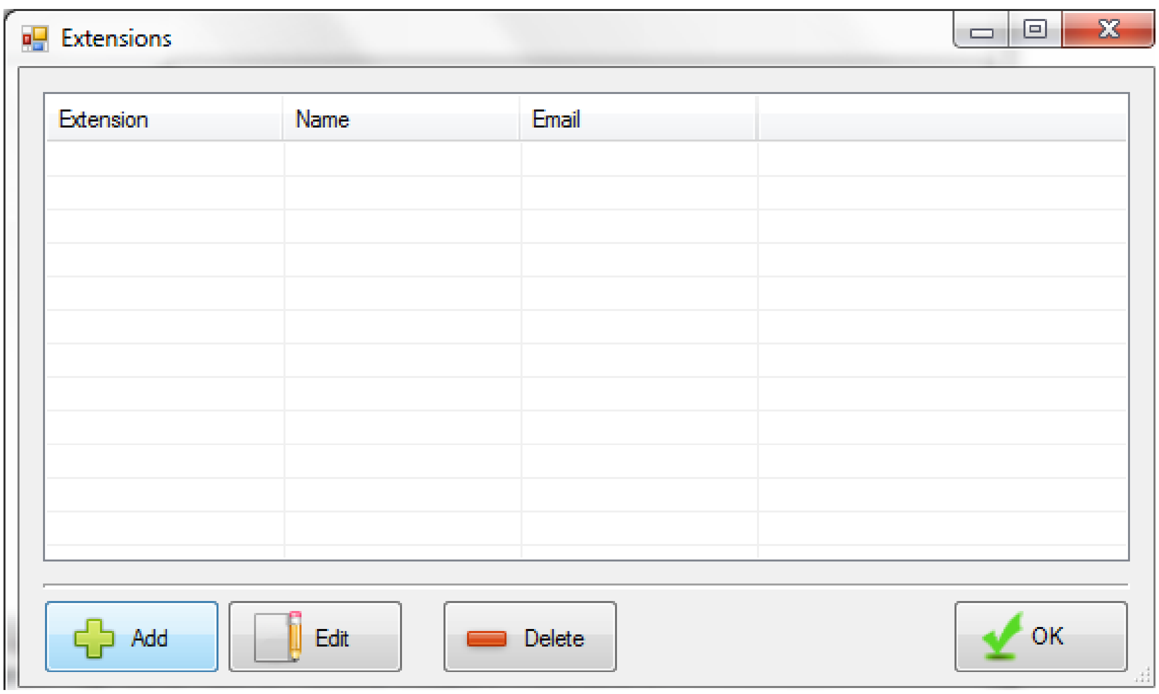

SIP Accounts(External Lines)
SIP Account is the registration information for another SIP system. Typically it can be an extension of another PBX, or an account from ITSP(Internet Telephony Service Provider).
Tested ITSPs
SIP Account List
Display Name
User Name
Domain
Proxy
Add
Edit
Delete
OK

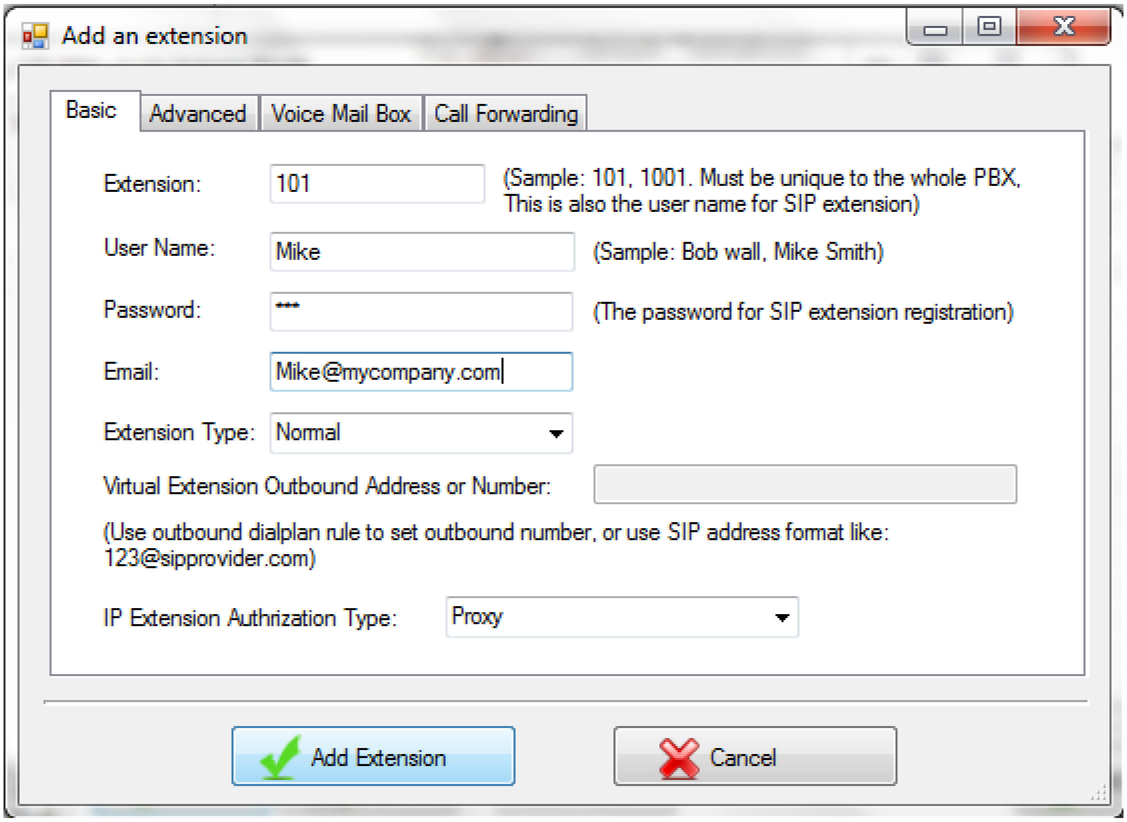

Add SIP Account
Basic
DIDs
Display Name:
MyExtenLine1
Sample: Bob Wall, Mike Keeler
User Name:
4321
Sample: 7184773245, 1001, or Mike
SIP Domain:
sipsrv.com
Sample: pcbest.net, voip.com
SIP Proxy:
sipsrv.com
Sample: pcbest.net, usually same as domain
Authorization:
4321
Sample: 7845, usually same as UserName
Password:
****
Your secret code
Expire Duration:
3600
In seconds, default is 3600 = 1 hour
Register with SIP proxy server to receive incoming calls
OK
Cancel

Extensions

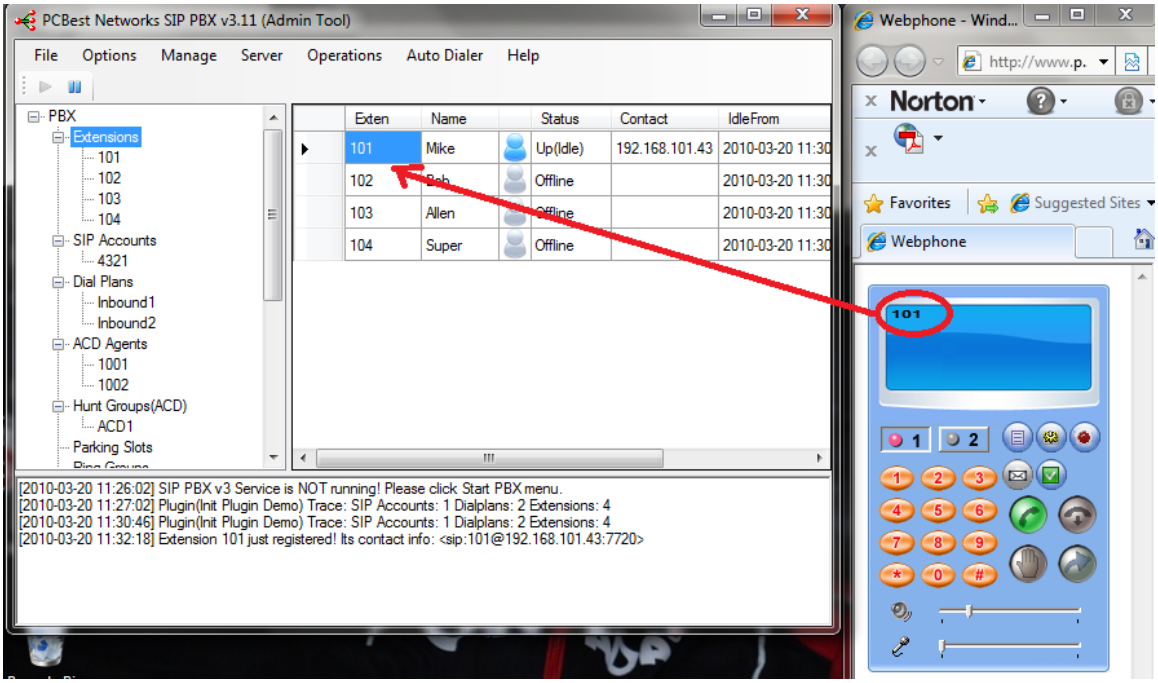
Extensions are internal phones to handle the calls. Usually extension name are three or four digits length, Like 101, 2010. One extension can also be considered as one SIP account for IP phone, or an outline for another PBX. Assume we setup three extensions here.





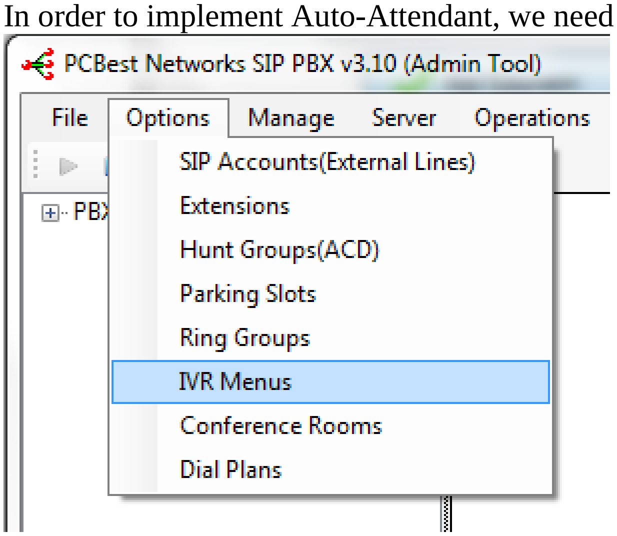


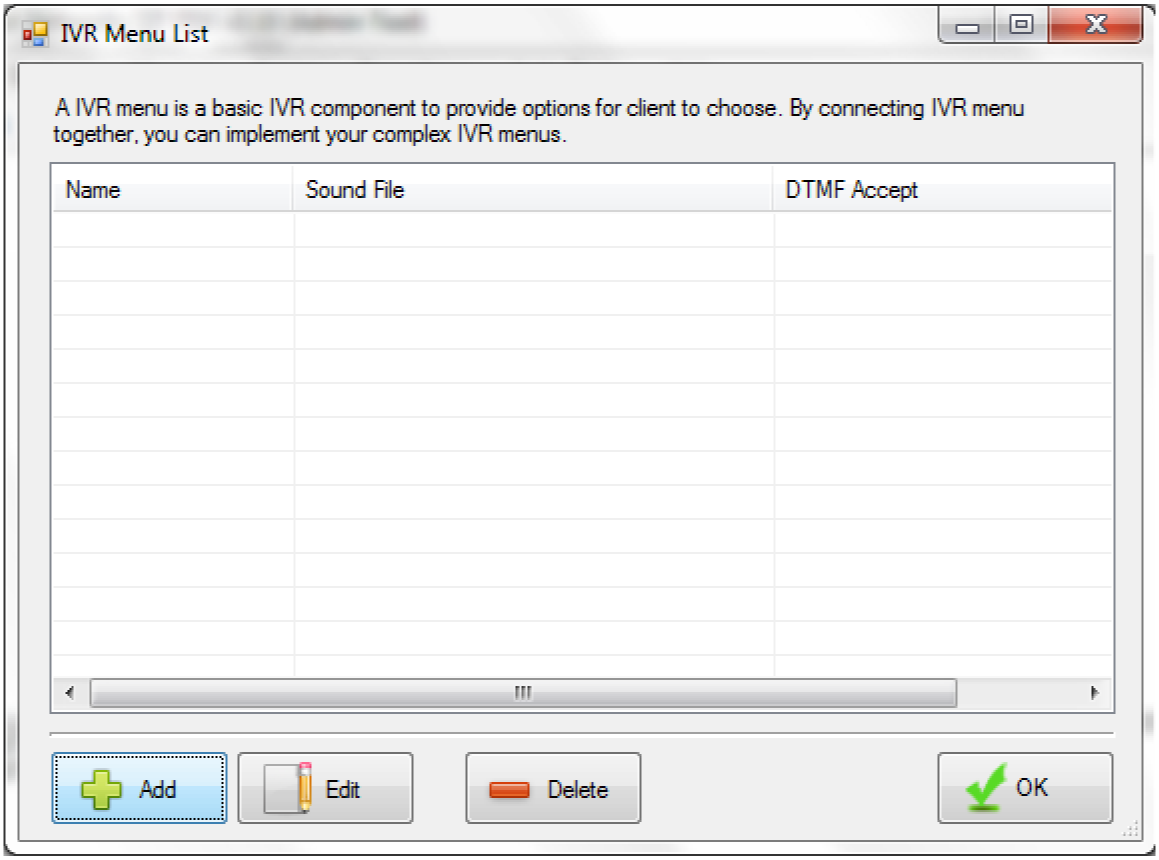
After you have setup three extensions 101, 102, and 103, you need to have 3 ipphones or computers to register on PBX to work as extensions. You can use any SIP hardware phones or softphones, like PCBest SIP ActiveX phone here: http://www.pcbest.net/activex.php

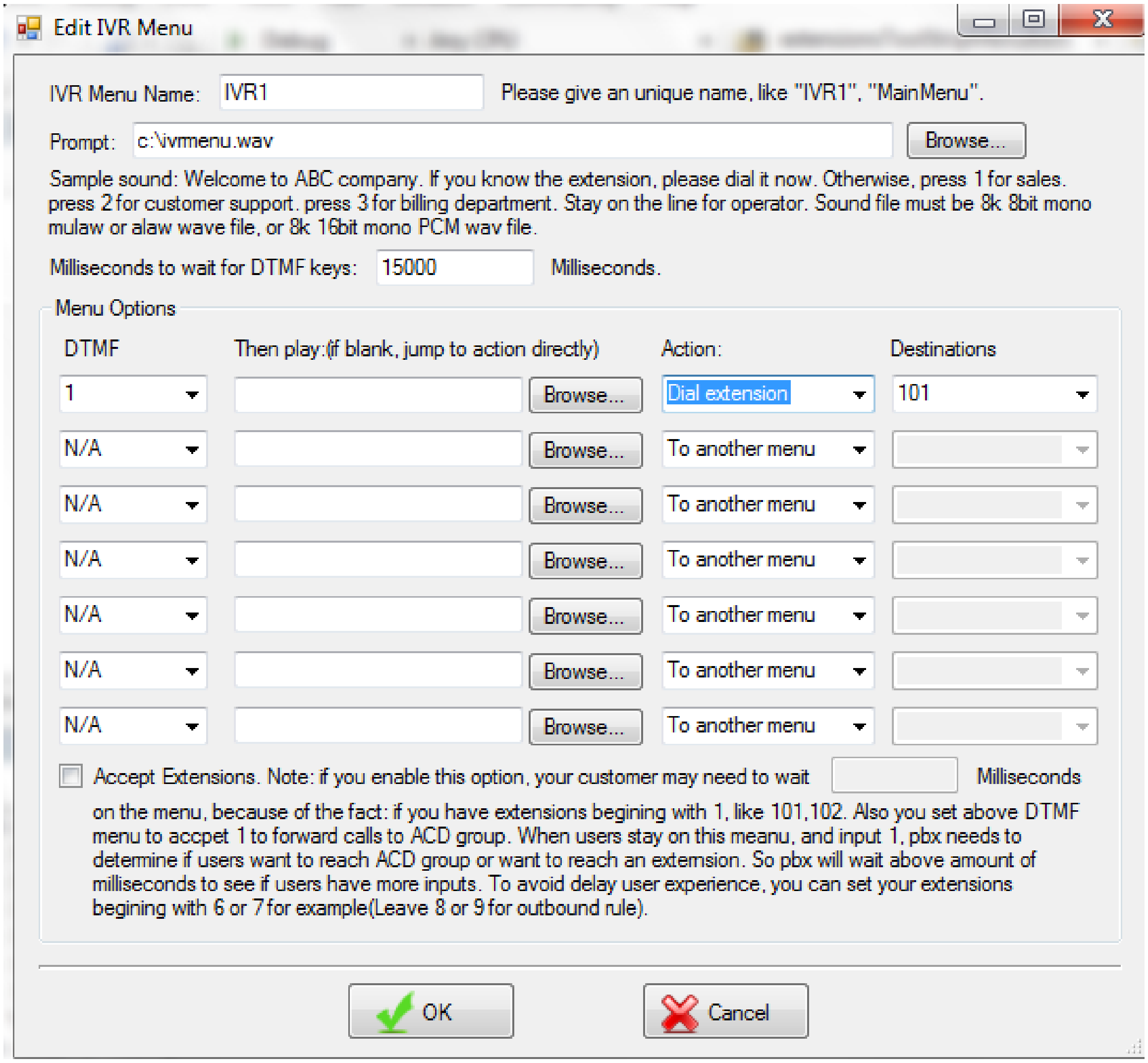


3.2 Auto Attendant

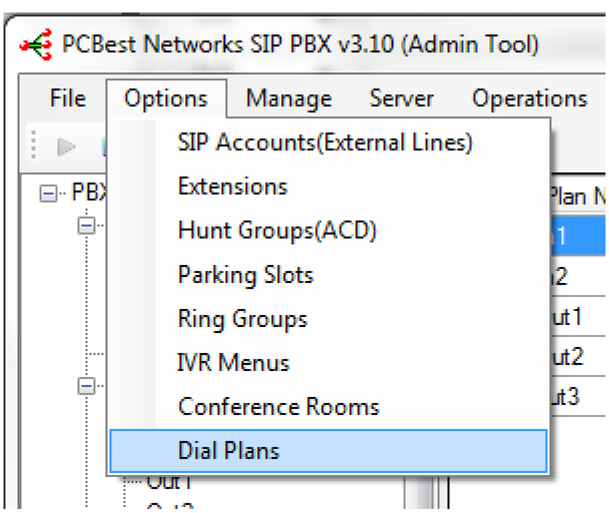
In order to implement Auto-Attendant, we need to set an IVR Menu first to play prompts.

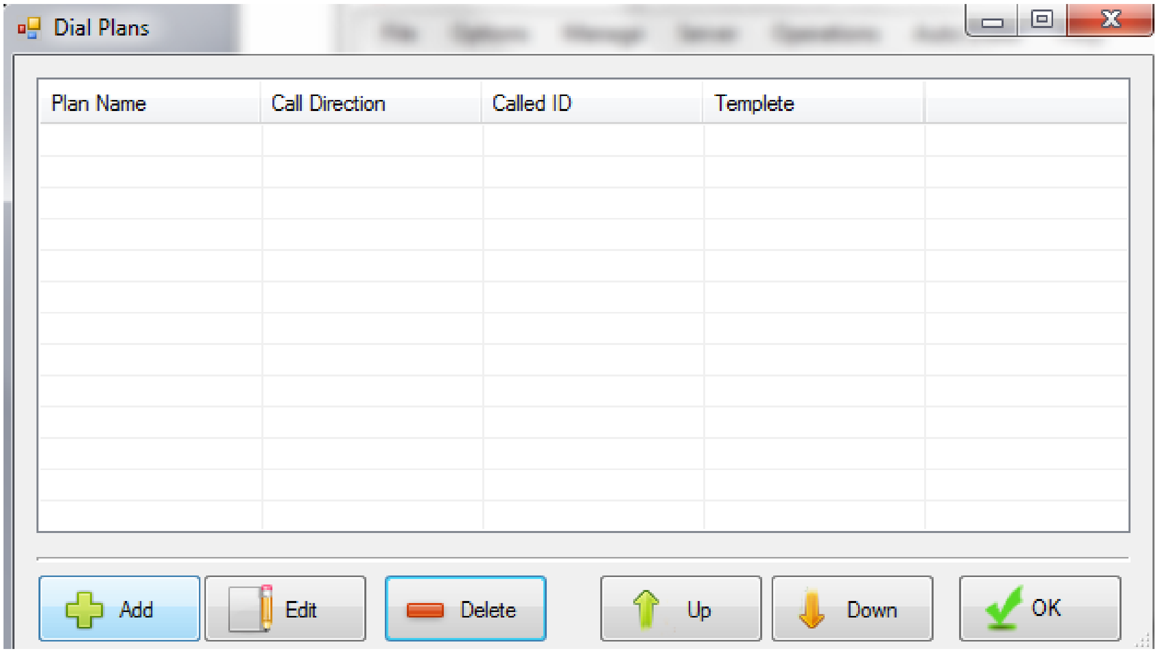



IVR Menu List
A IVR menu is a basic IVR component to provide options for client to choose. By connecting IVR menu together, you can implement your complex IVR menus.
Name
Sound File
DTMF Accept
Add
Edit
Delete
OK

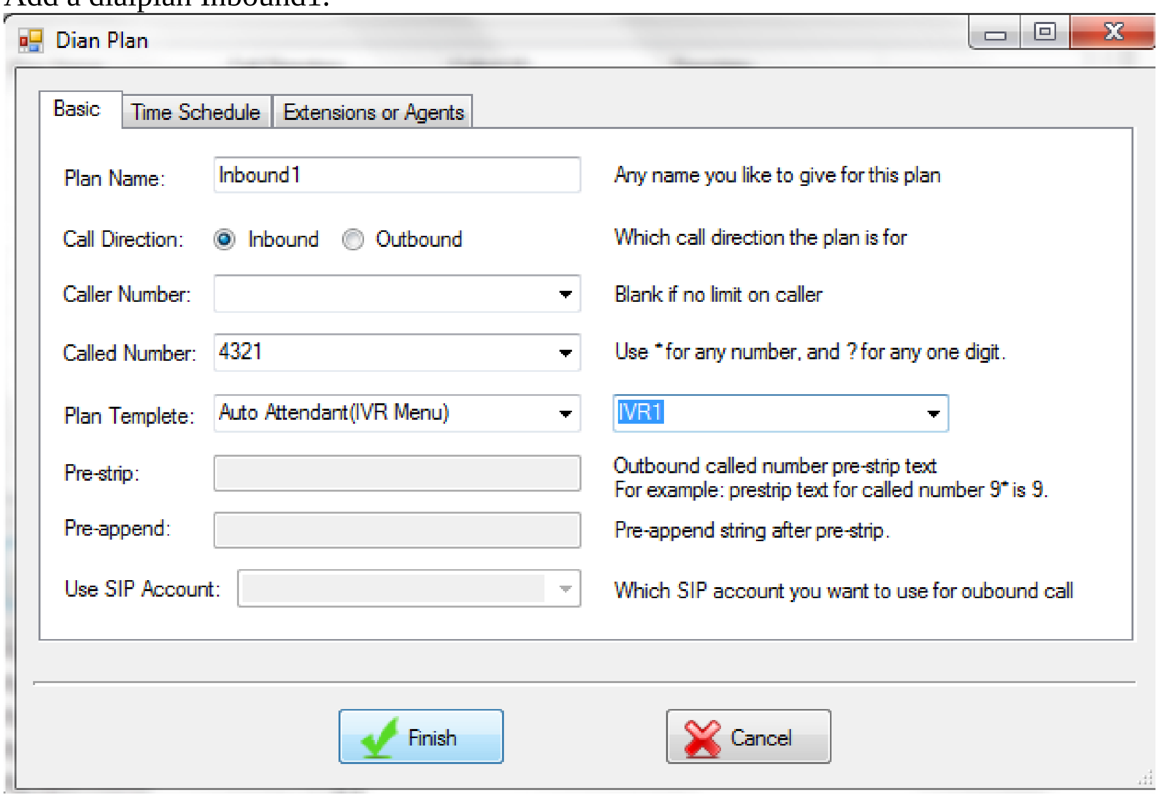


Then we need to setup an inbound dialplan to connect incoming calls into this IVR menu.





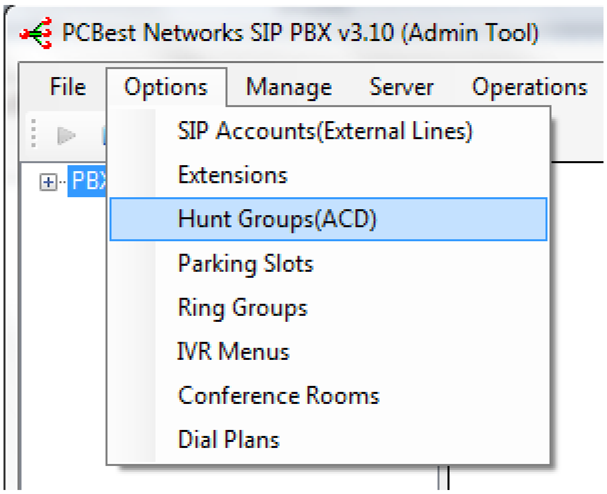
Add a dialplan Inbound1.

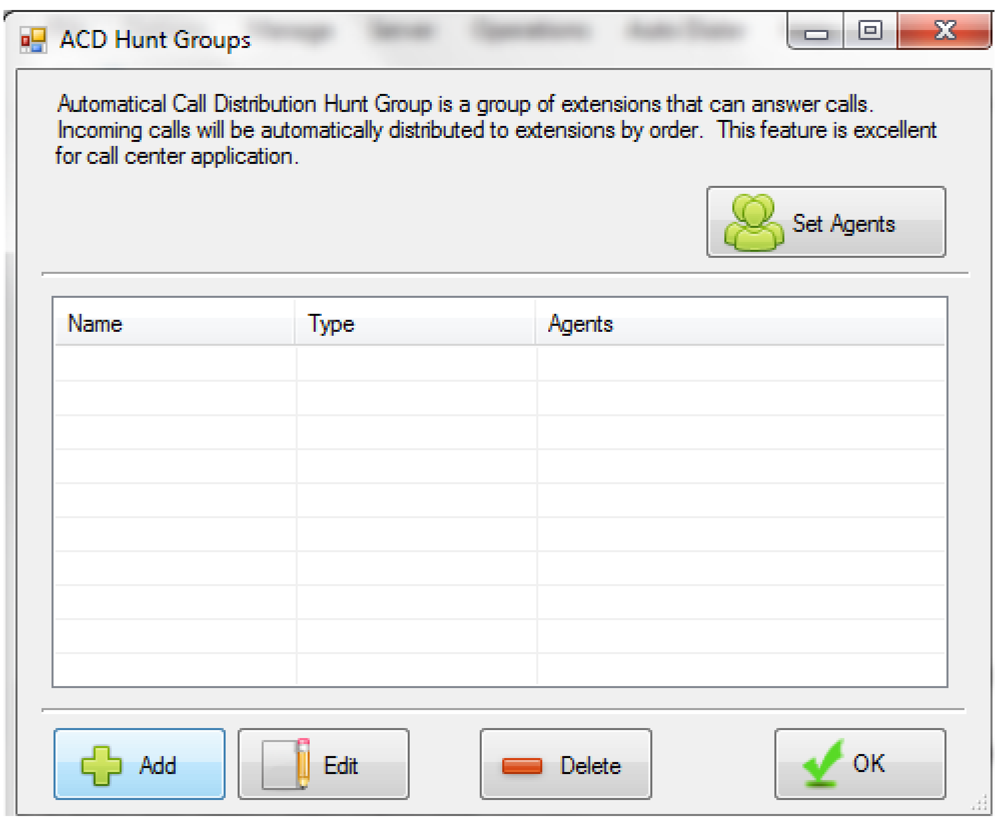


Then when you dial the DID that SIP account 4321 is linked, it will use Dialplan "Inbound1" to handle the call, and call goes to IVR menu "IVR1".

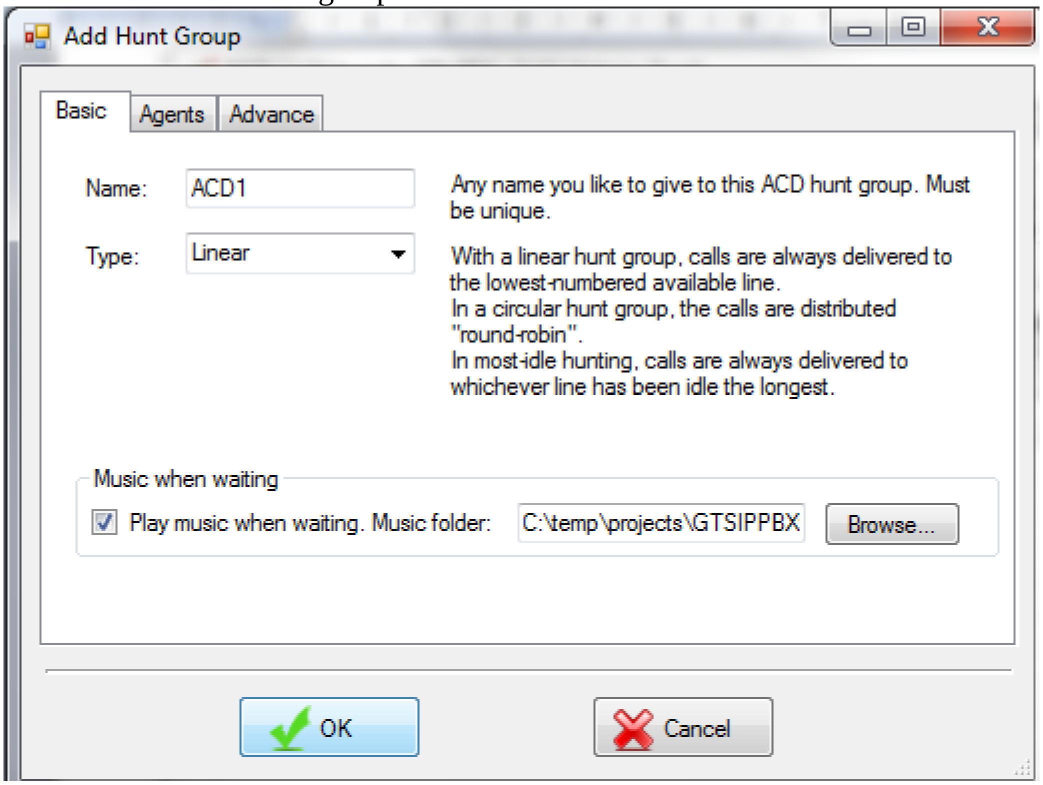
3.3 ACD(Automatical Call Distribution)

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.

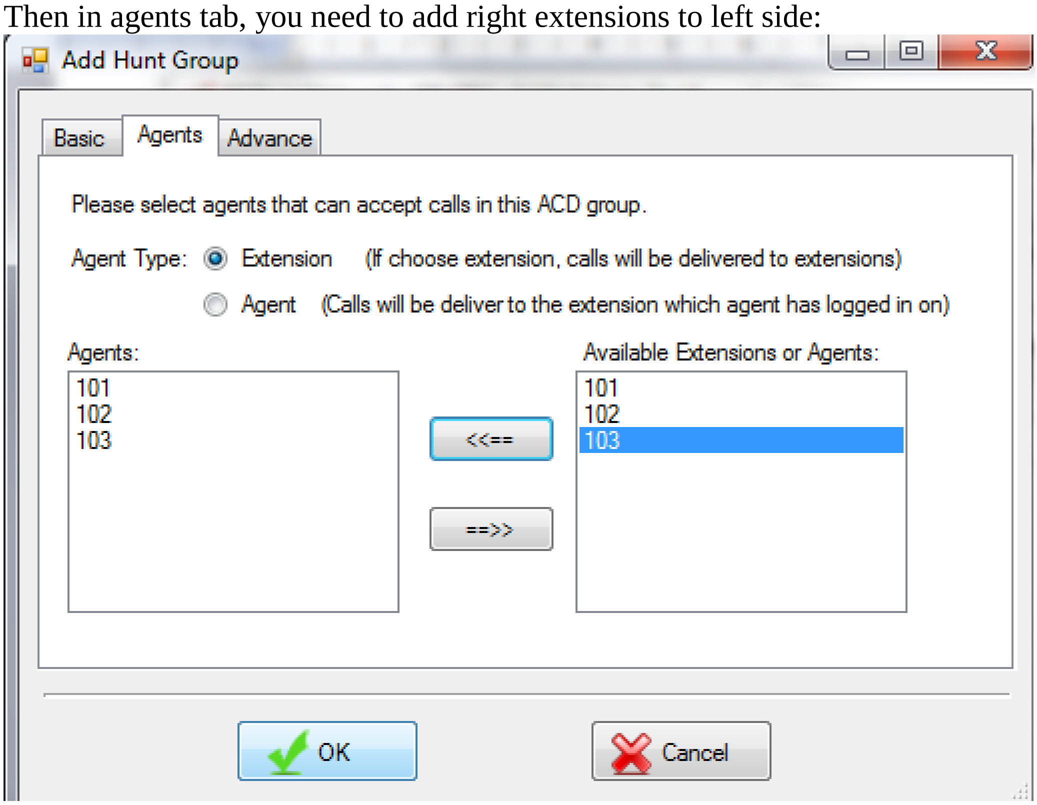




Then add one ACD huntgroup:

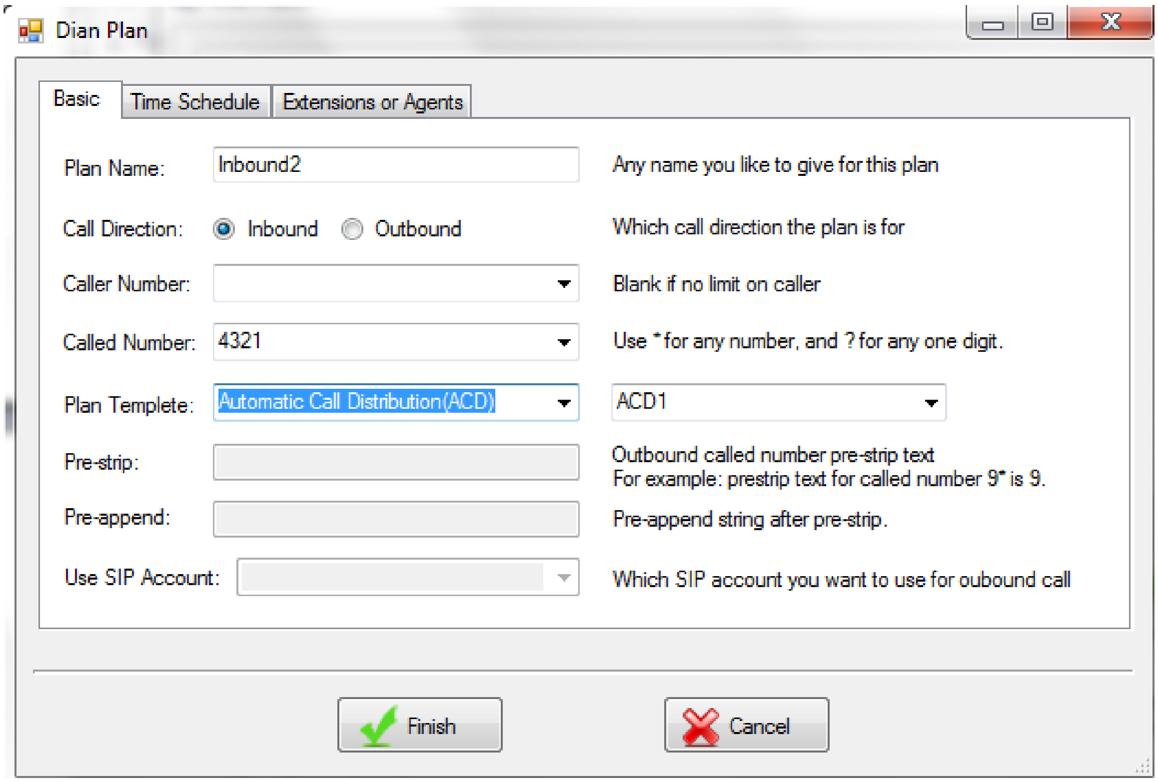


Then in agents tab, you need to add right extensions to left side:



Then click OK.

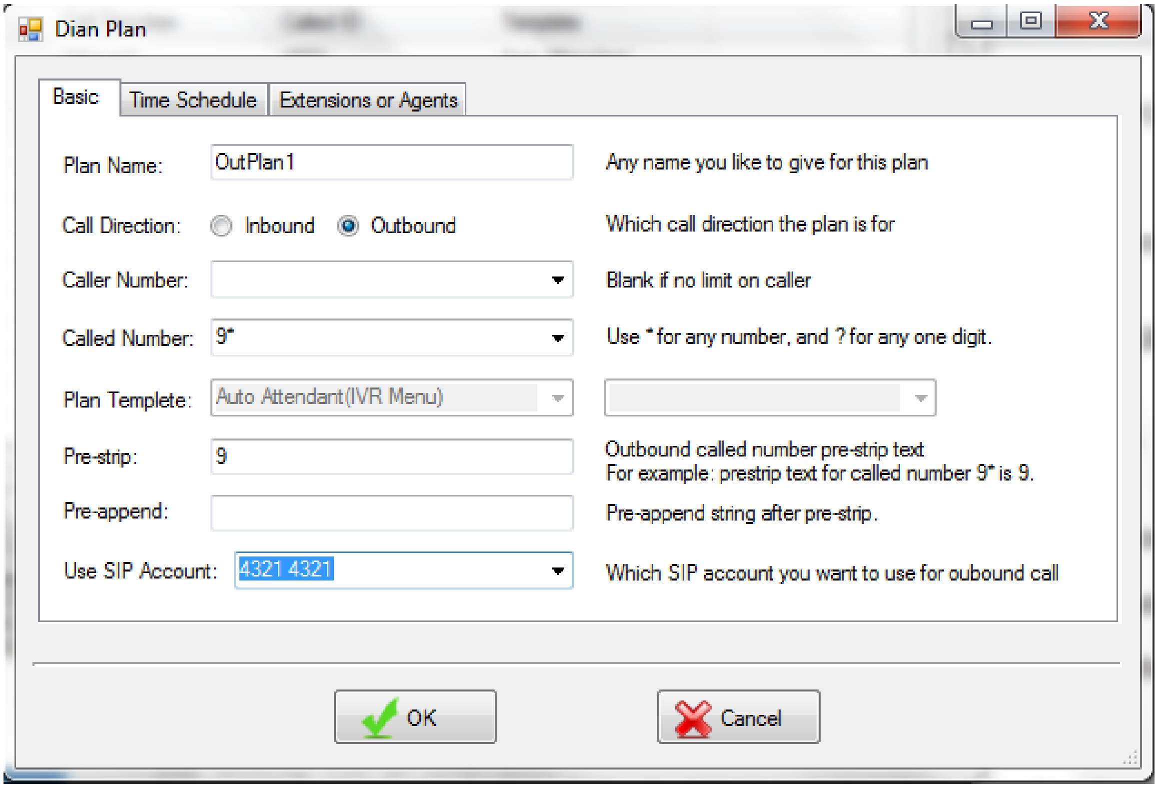
Again, we need to setup an inbound dialplan to connect inbound calls to this ACD huntgroup. Assume we add an inbound dialplan Inbound2 to handle this situation.



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

3.4 Outbound Calls

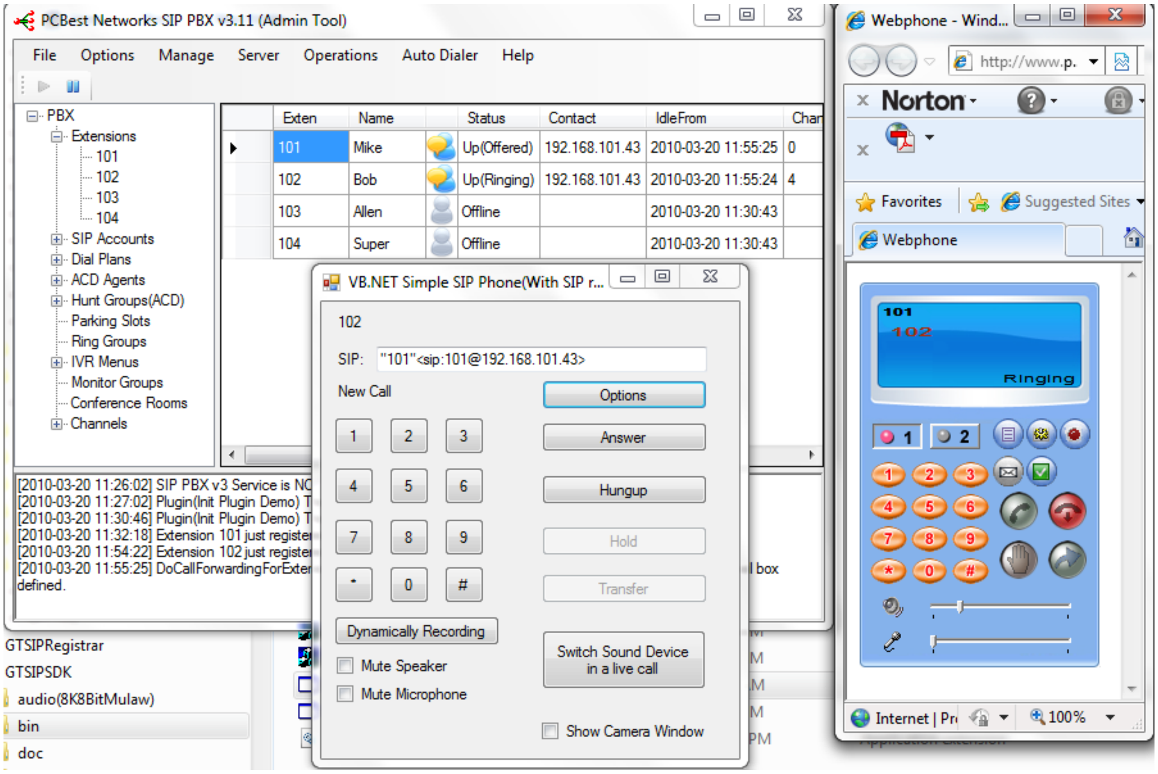
Add a dialplan. Give an plan name like OutPlan1. Set it to outbound type.



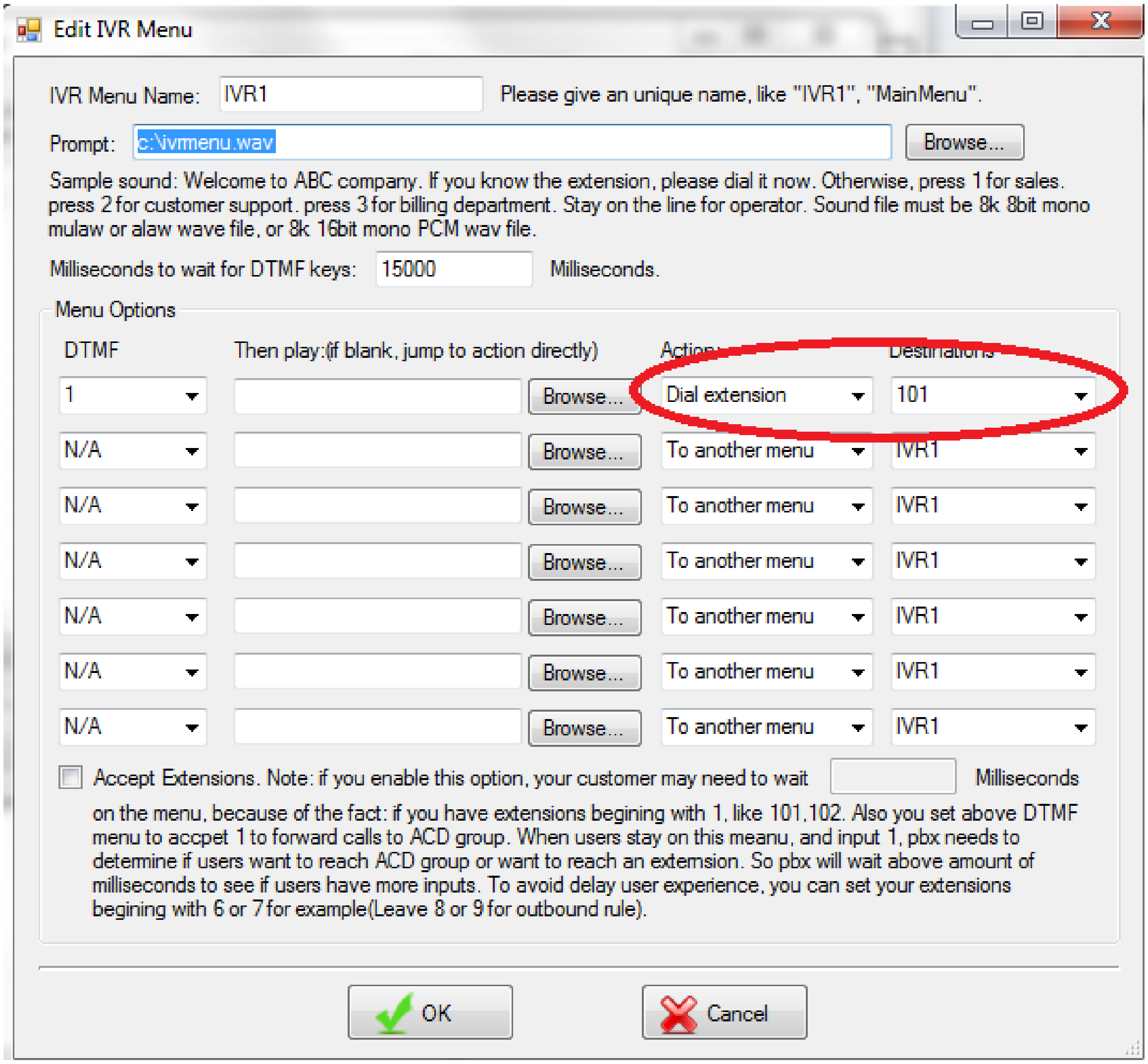
Set called number as 9*, and set the SIP account you want to use for dialing out.
Set pre-strip as 9.
It means that any calls go into PBX, which called id starts with 9, the PBX will regard it as an outbound call. PBX will take 9 in the front of called number, and use SIP account 4321 we created to dial 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

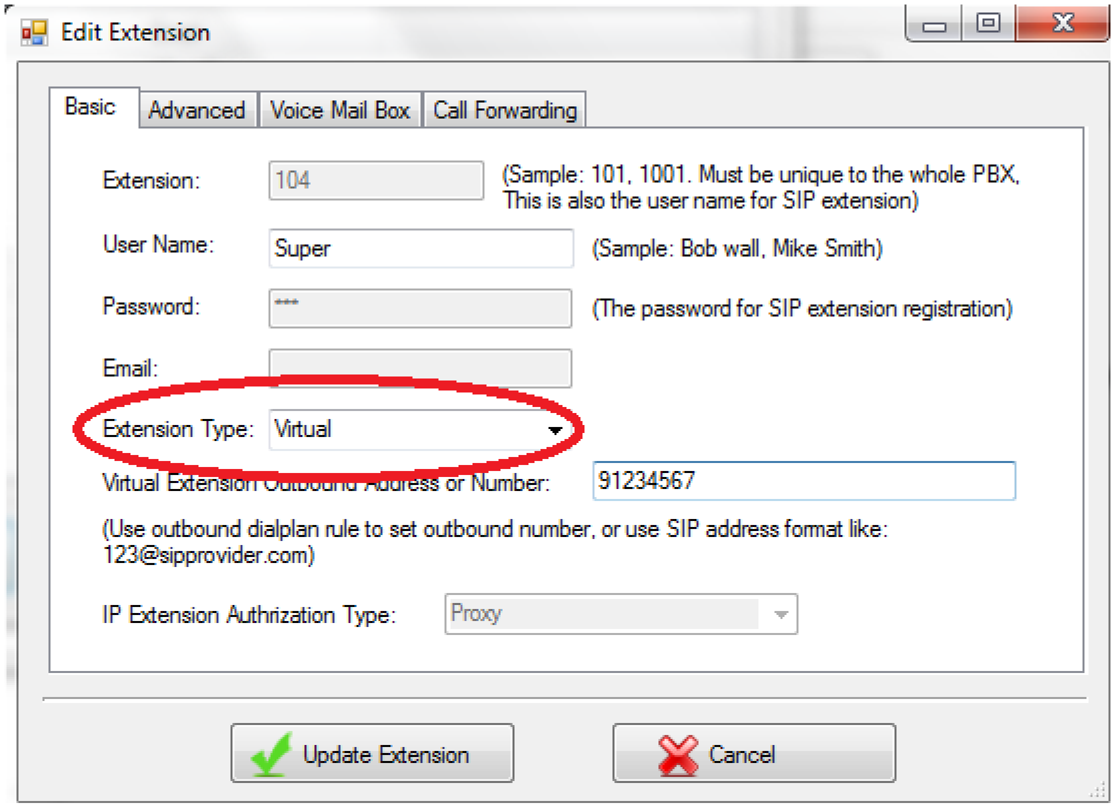


Dial to extension from other options(ACD, IVR menu, …)



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.

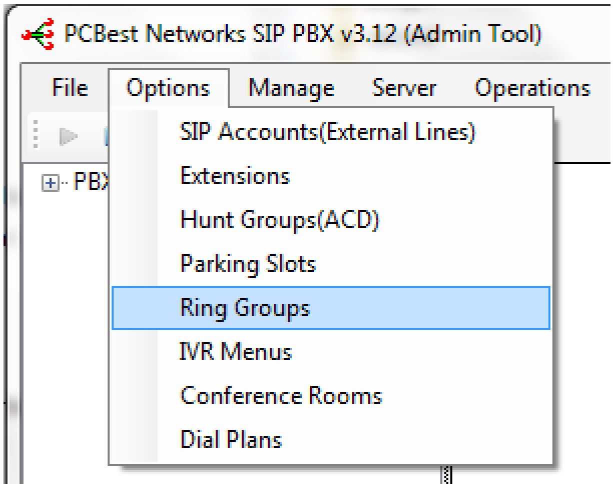


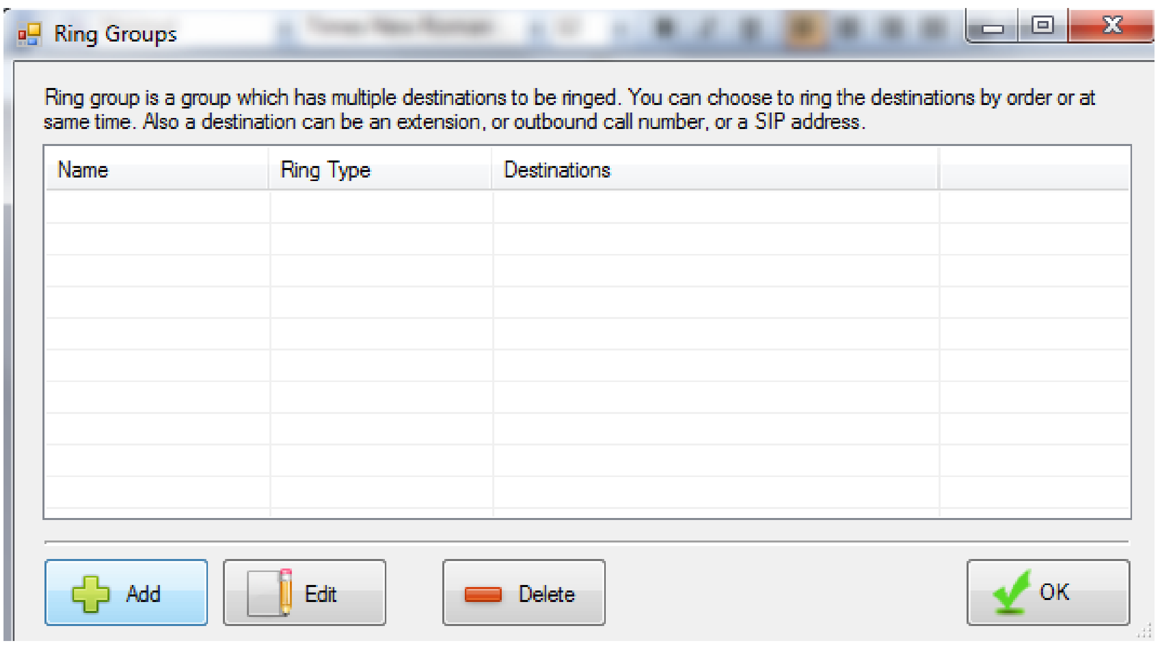
We set 91234567 here, which means using outbound plan 9*. When calls go to this extension, PBX will try to reach outside number 1234567.

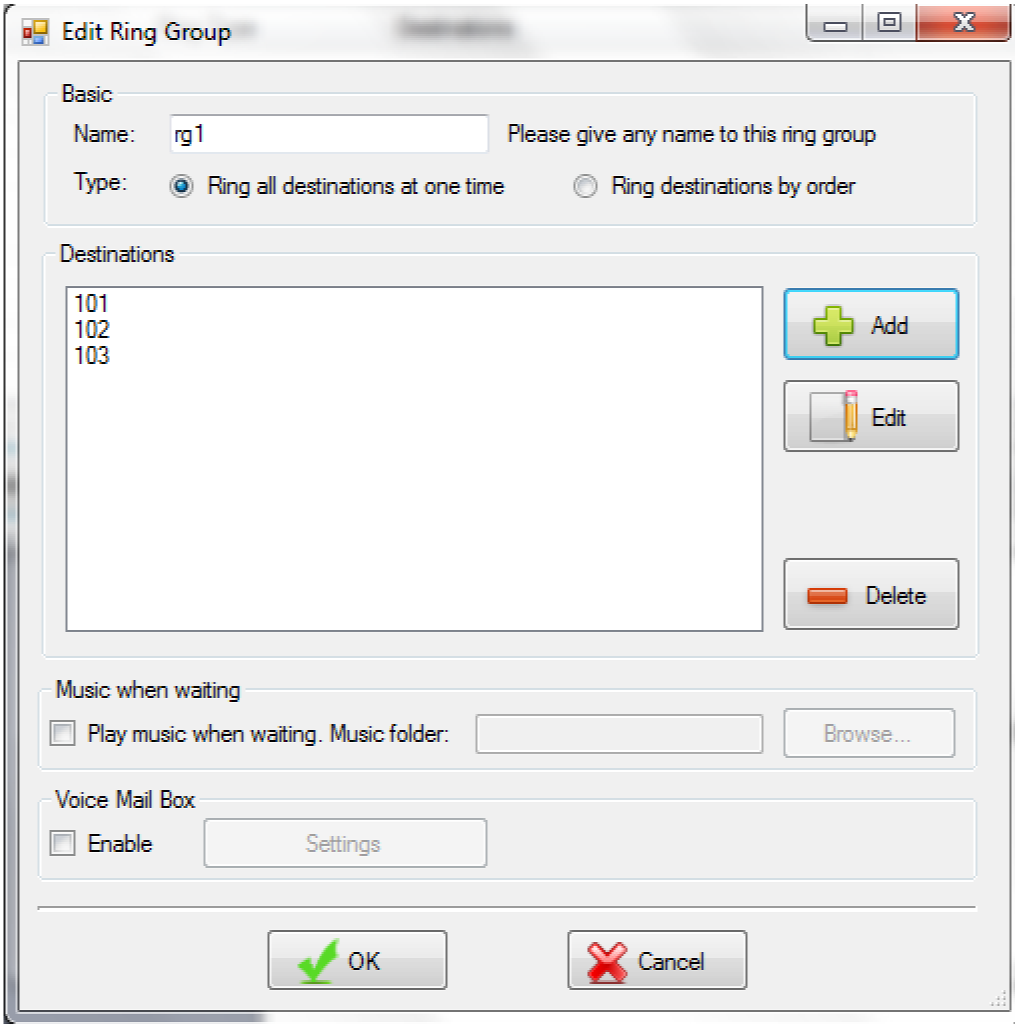
3.7 Ring group

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

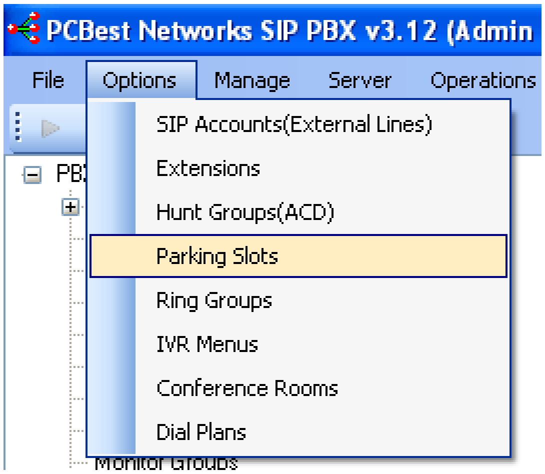

Ring Groups
Ring group is a group which has multiple destinations to be ringed. You can choose to ring the destinations by order or at same time. Also a destination can be an extension, or outbound call number, or a SIP address.
Name
Ring Type
Destinations
Add
Edit
Delete
OK



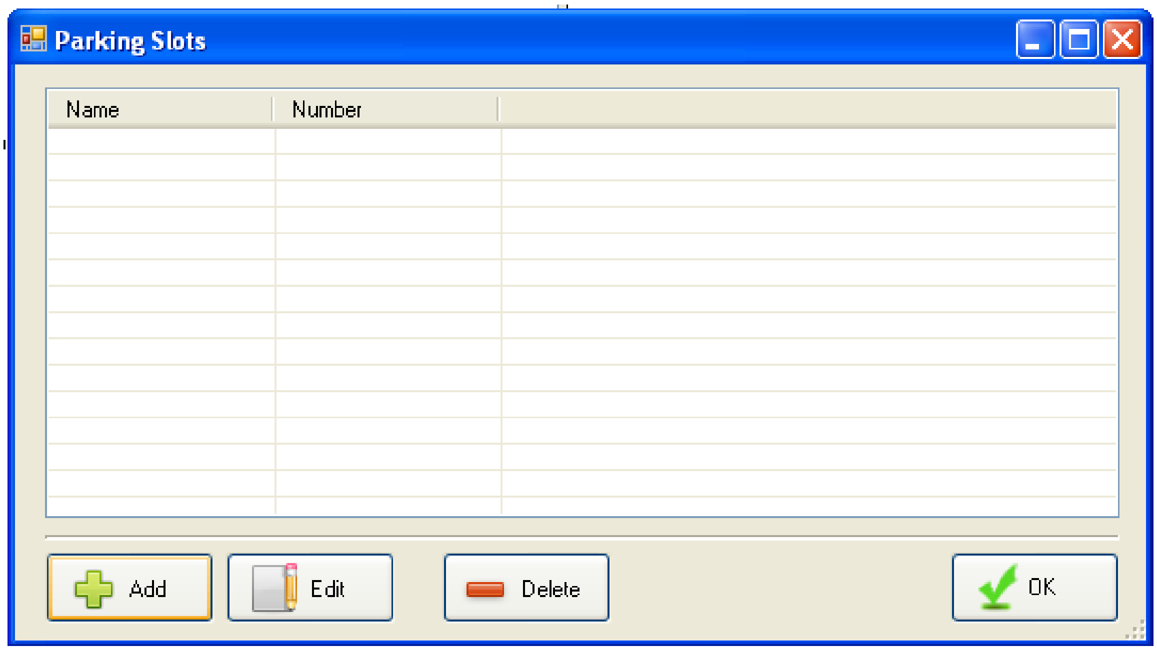
Three extensions 101, 102, 103 are added into ring group rg1. Then we can set up an inbound dialplan, to forward calls to this ring group. When a call comes in and reach this ring group, pbx will ring extensions 101, 102, 103 at same time.

3.8 Call Parking

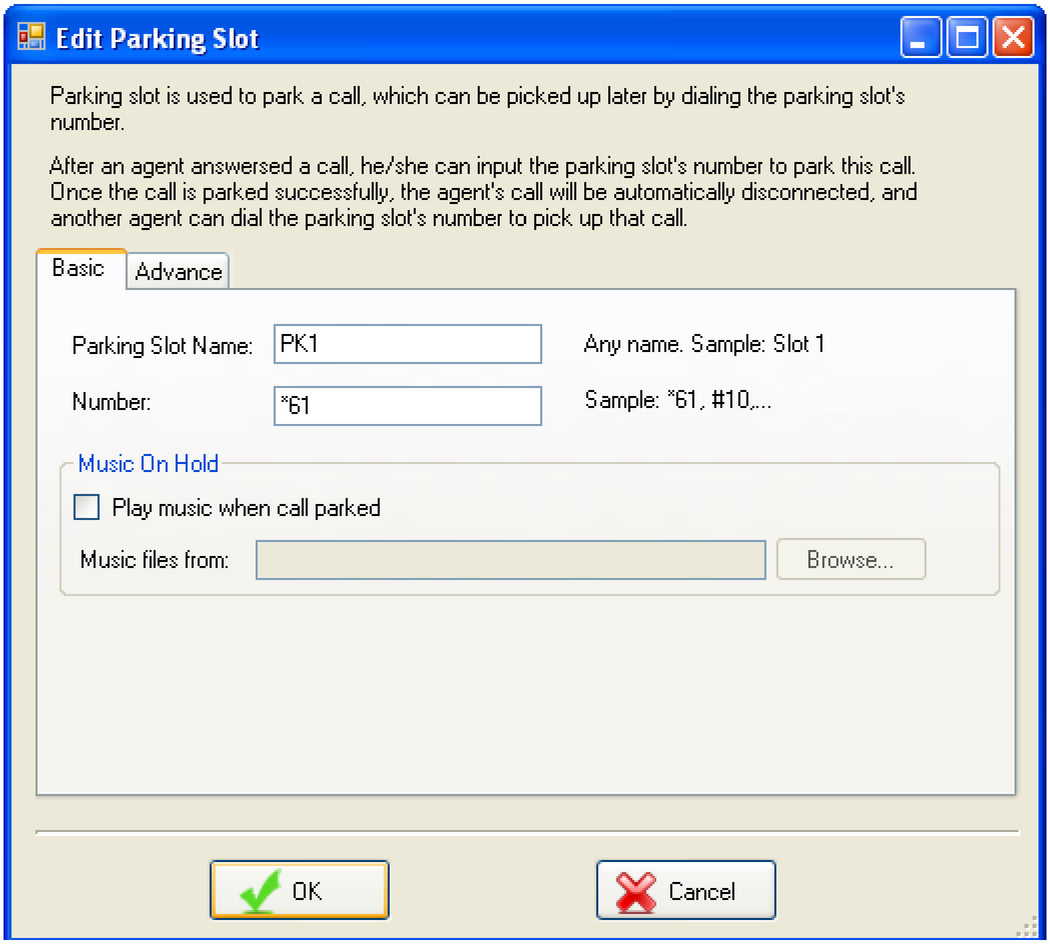
Call Parking is used to park a call. You must define a call parking slot first to allow the call to park, then later the call can be picked up by another extension or agent.



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



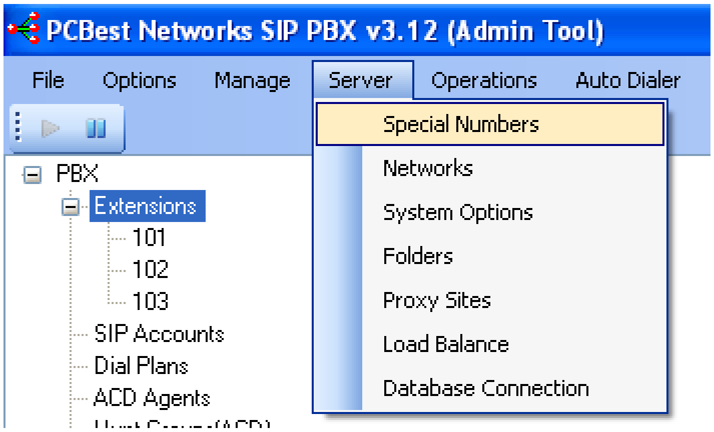
After defined a Parking Slot "PK1", you can try an incoming call which is transferred into an extension or agent. When extension pressed *61, the call should be parked. Another extension should be able to pick up this call by dialing *61 into PBX.

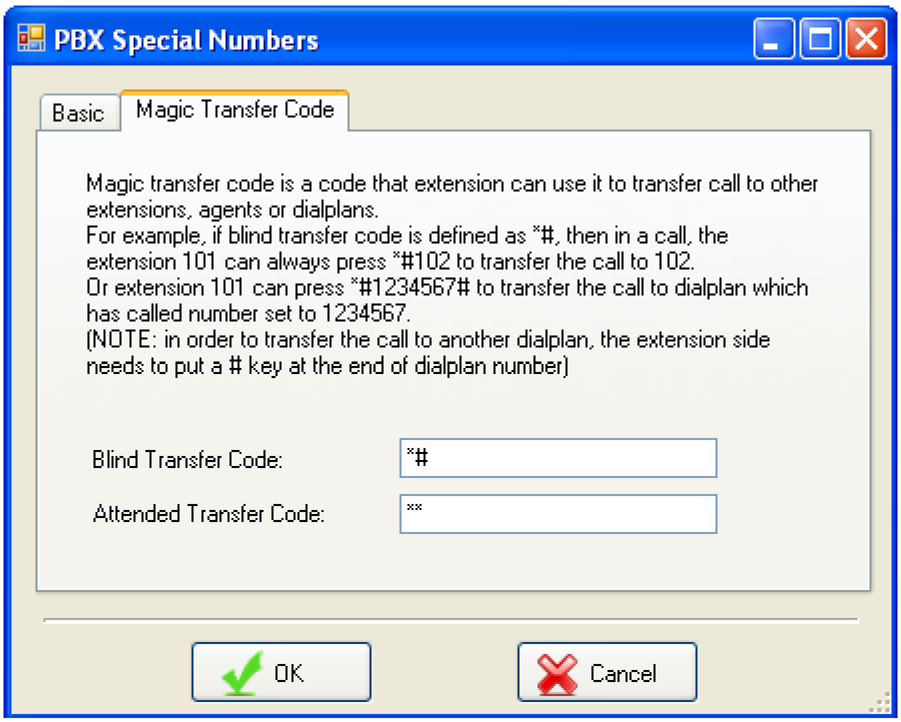
3.9 Magic Transferring Code (ONLY V3)

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 **.

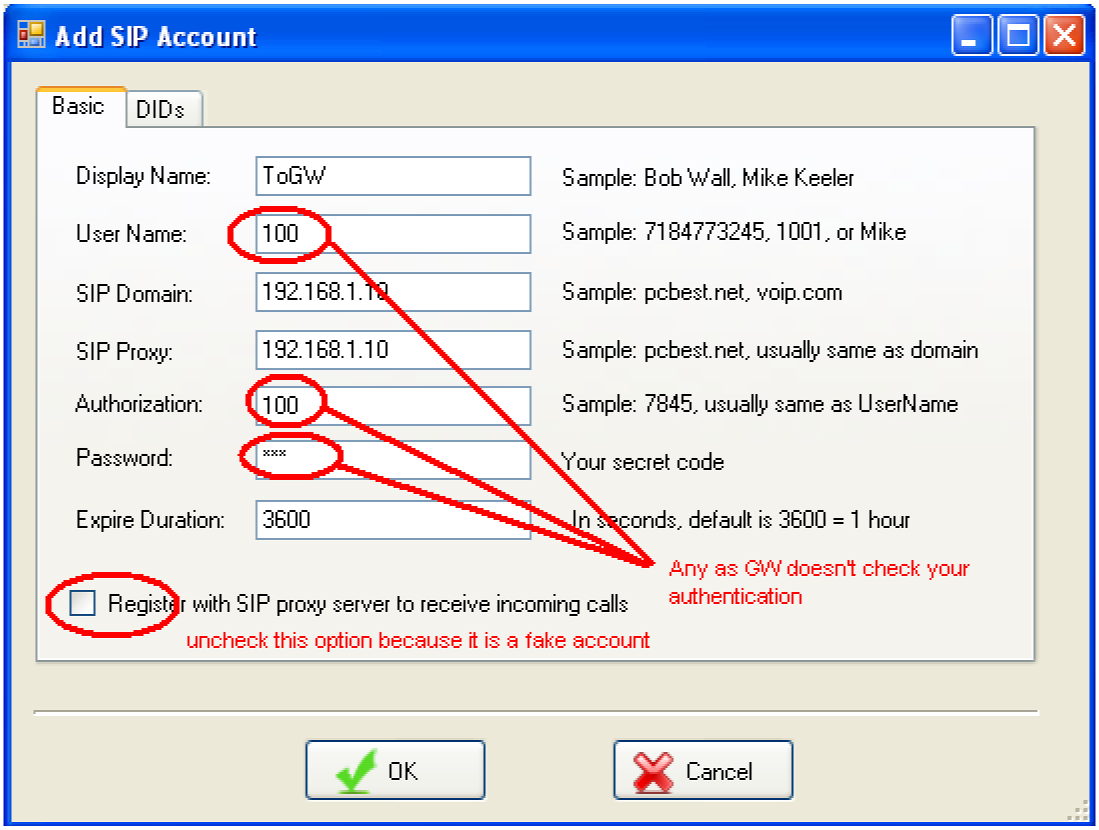




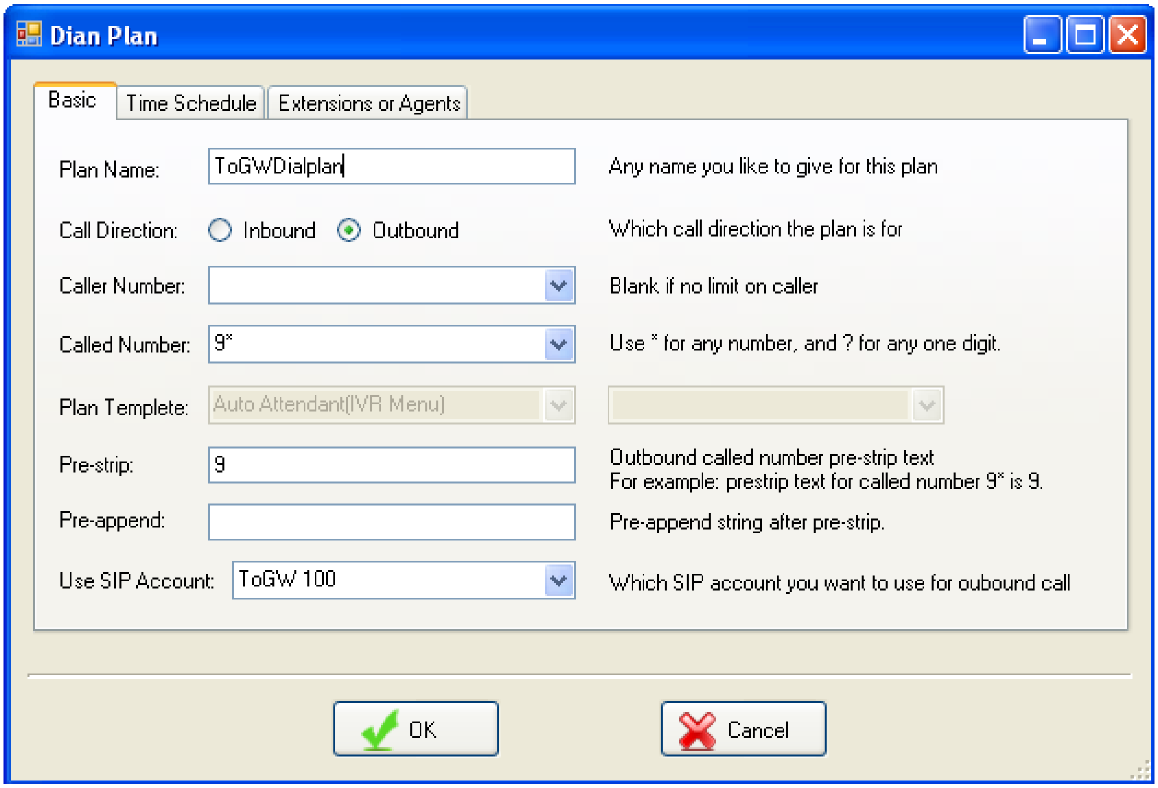
3.10 FXO/FXS or Digital Gateway

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:

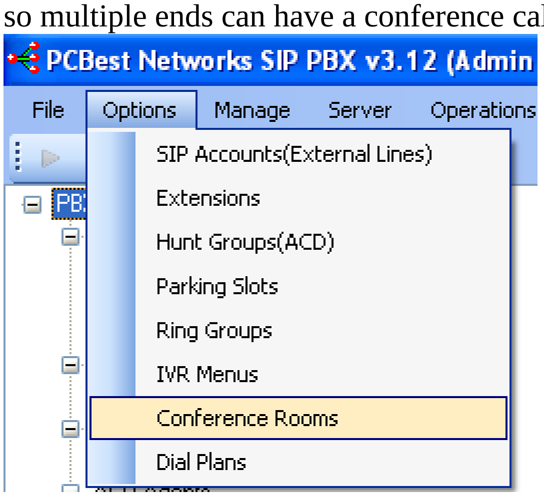


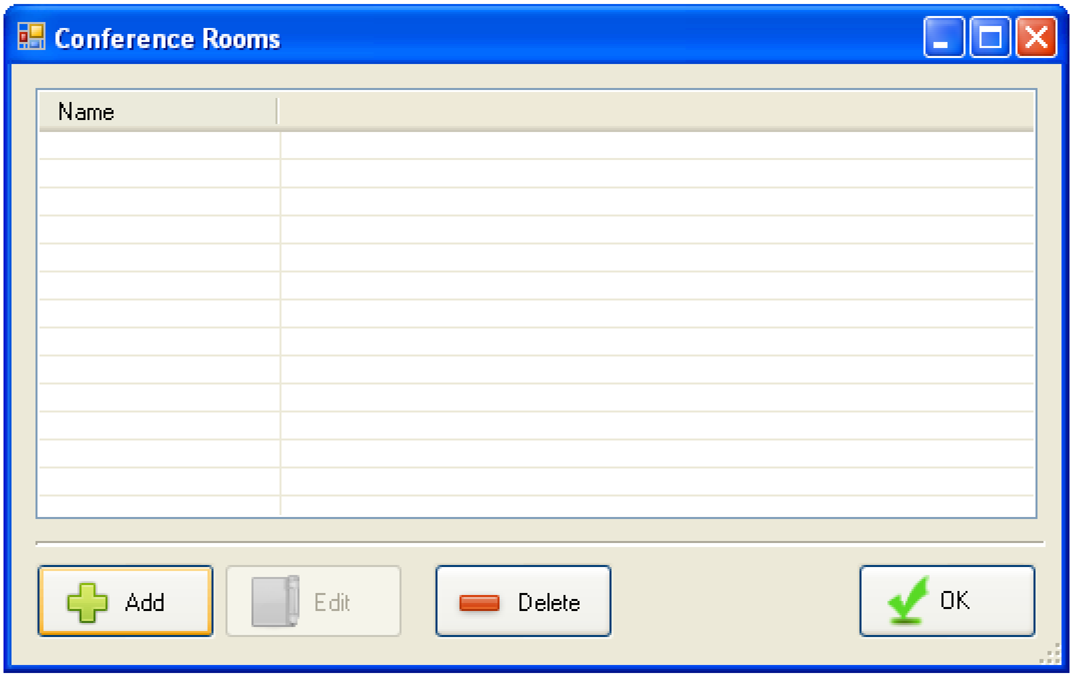
By doing this, you setup a peer which is connected to your gateway. Next step, you need to setup an outbound dialplan to use this sip account to forward extension calls into gateway.

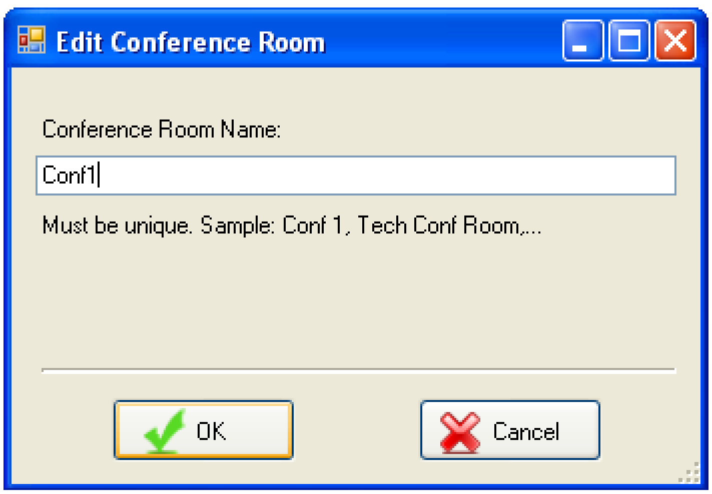


3.11 Conference Room

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





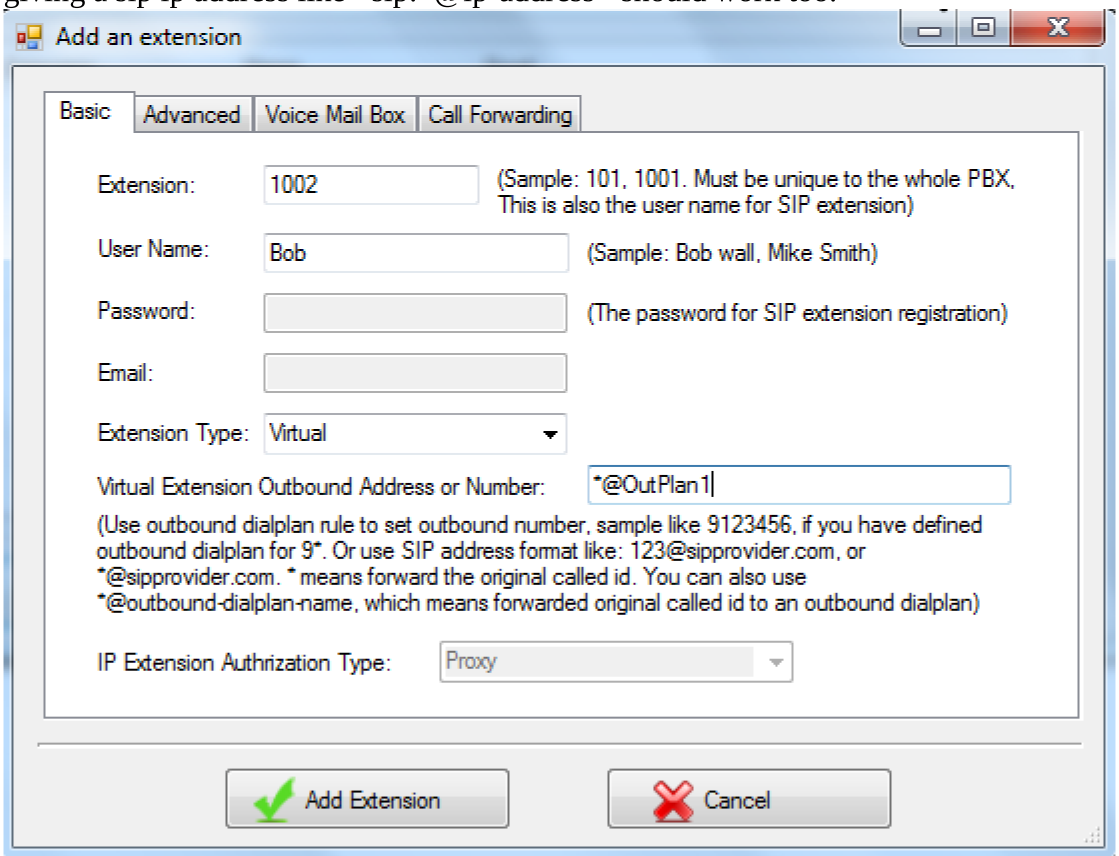


Then you can define a dialplan to forward incoming calls into this conference room.

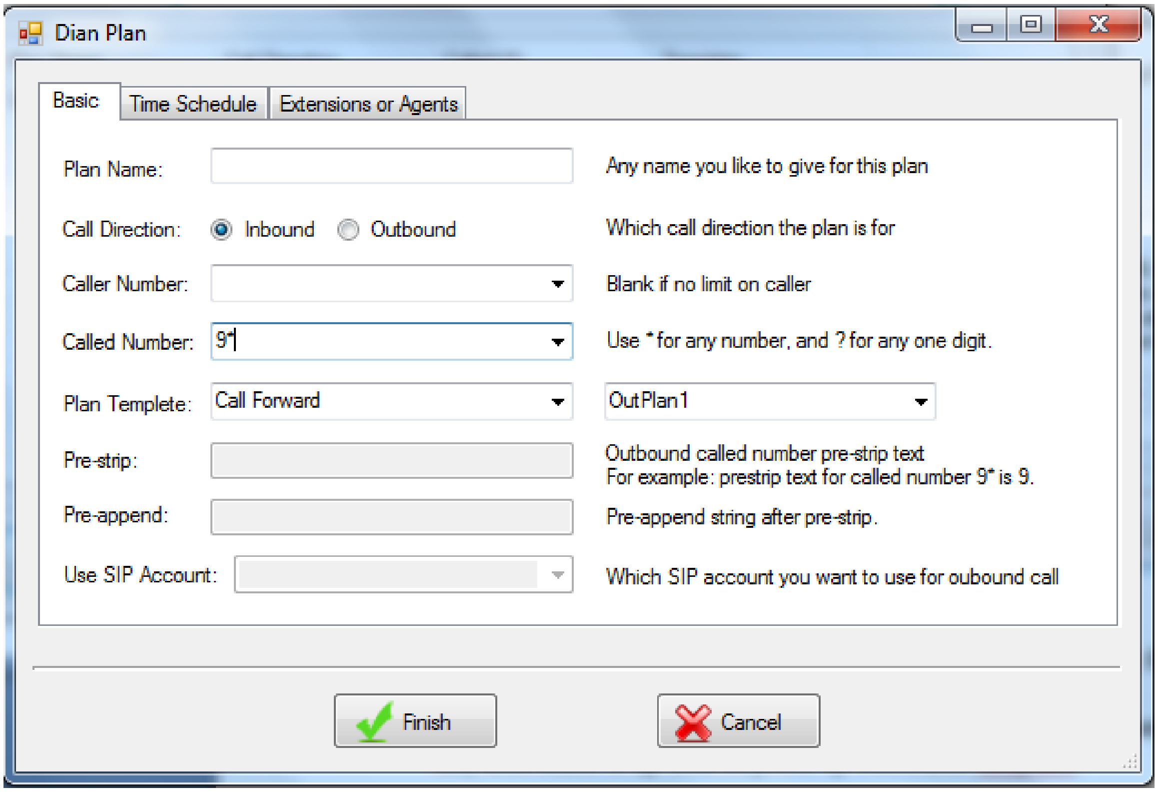
3.12 Inbound 2 Outbound

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.

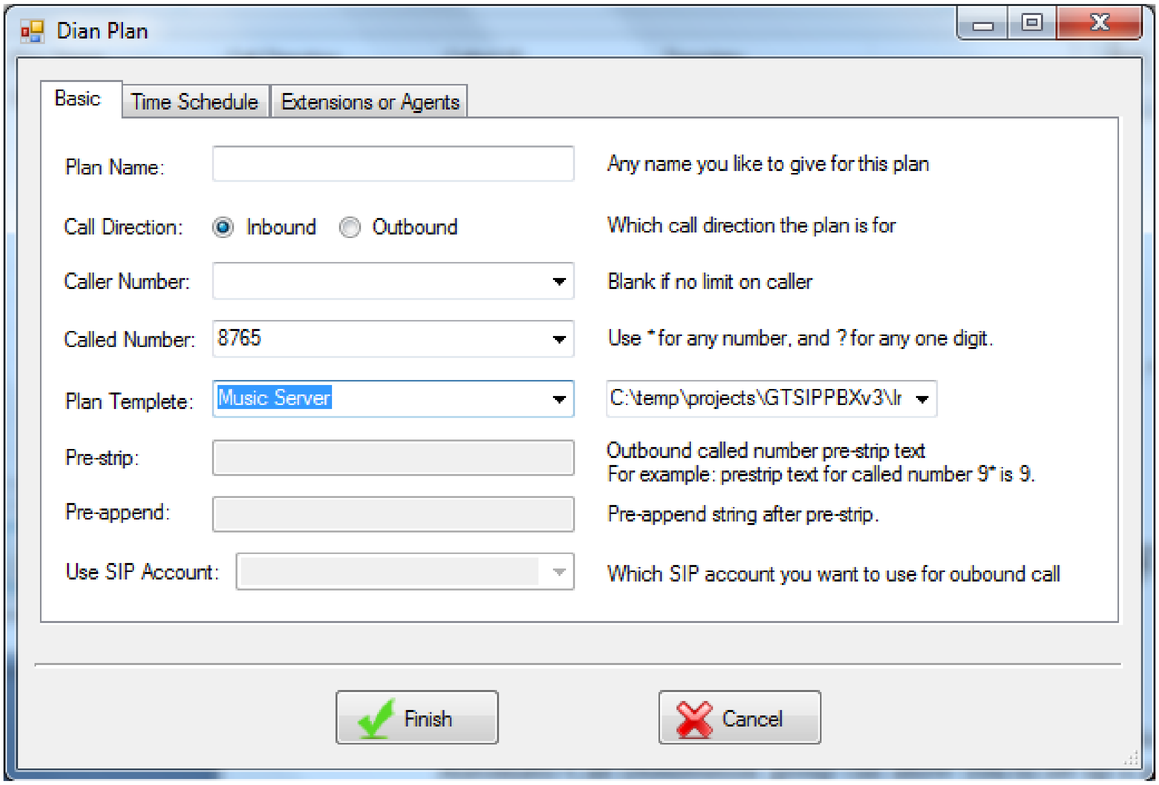


2. Use call forward inbound dialplan
Create an inbound dialplan, set call template to call forward, then choose an outbound dialplan for call forwarding.
Note, for this call forwarding inbound dialplan, please adjust its order in the dialplan list, and make it up and be front of outbound dialplan.



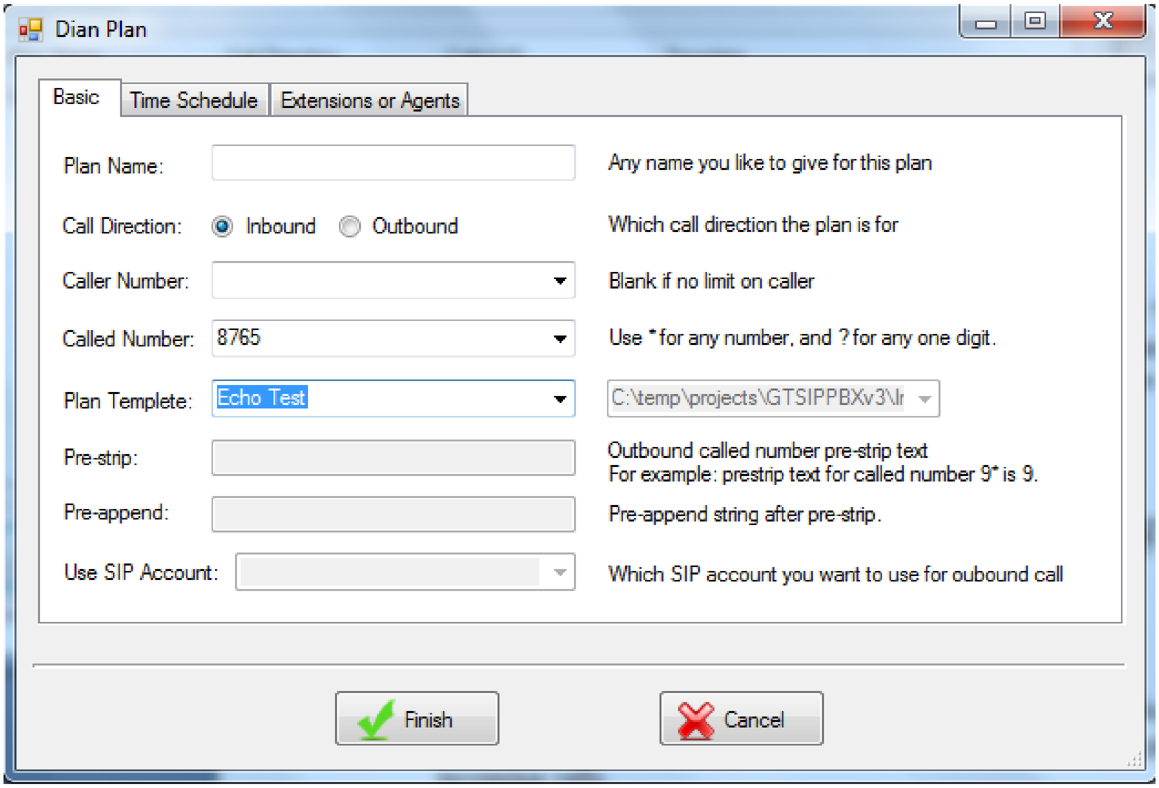
3.13 Setup a music server

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.

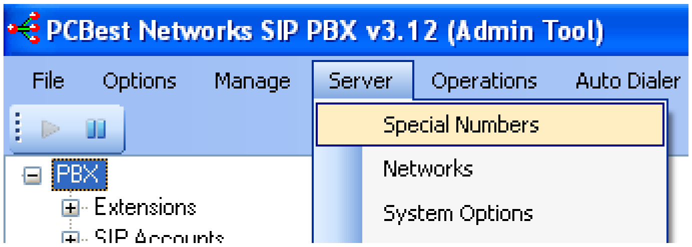


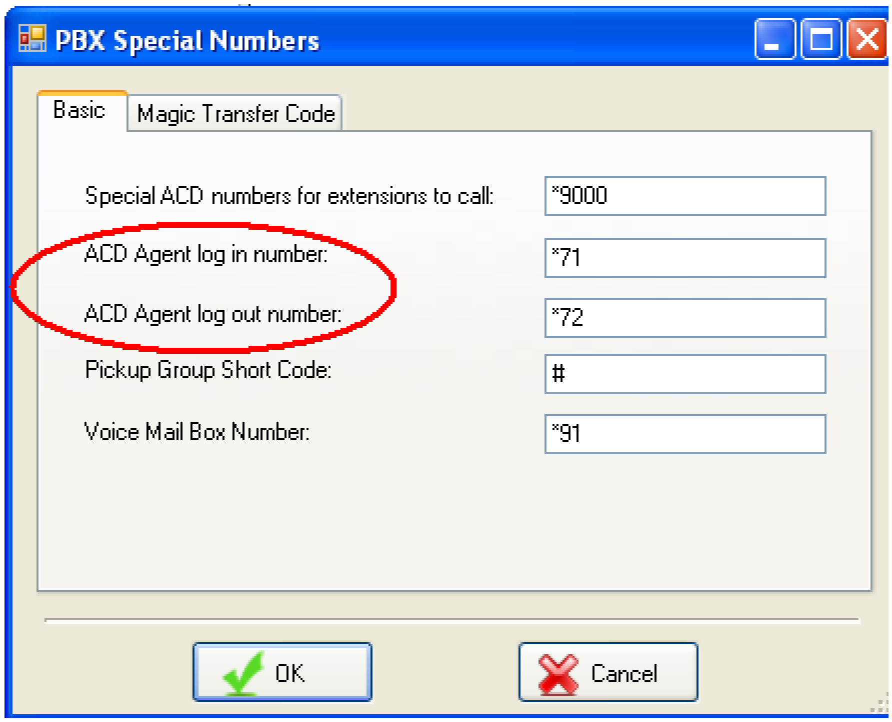
4 PBX Advanced Call Center Features

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

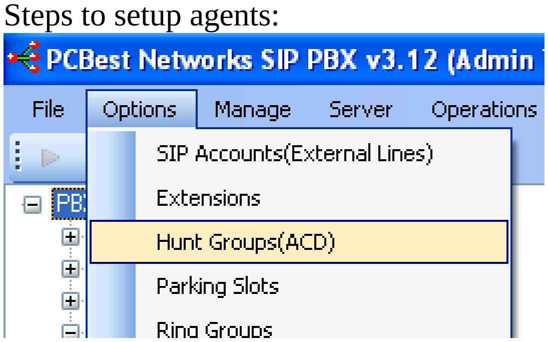
What is an agent? An agent is **NOT** an extension.
An extension is a physical phone, but an agent is a real person to work on an extension. So there may be more than one agent working on the same extension. Usually in a call centre environment, an agent will start to work by login at one of the extension.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

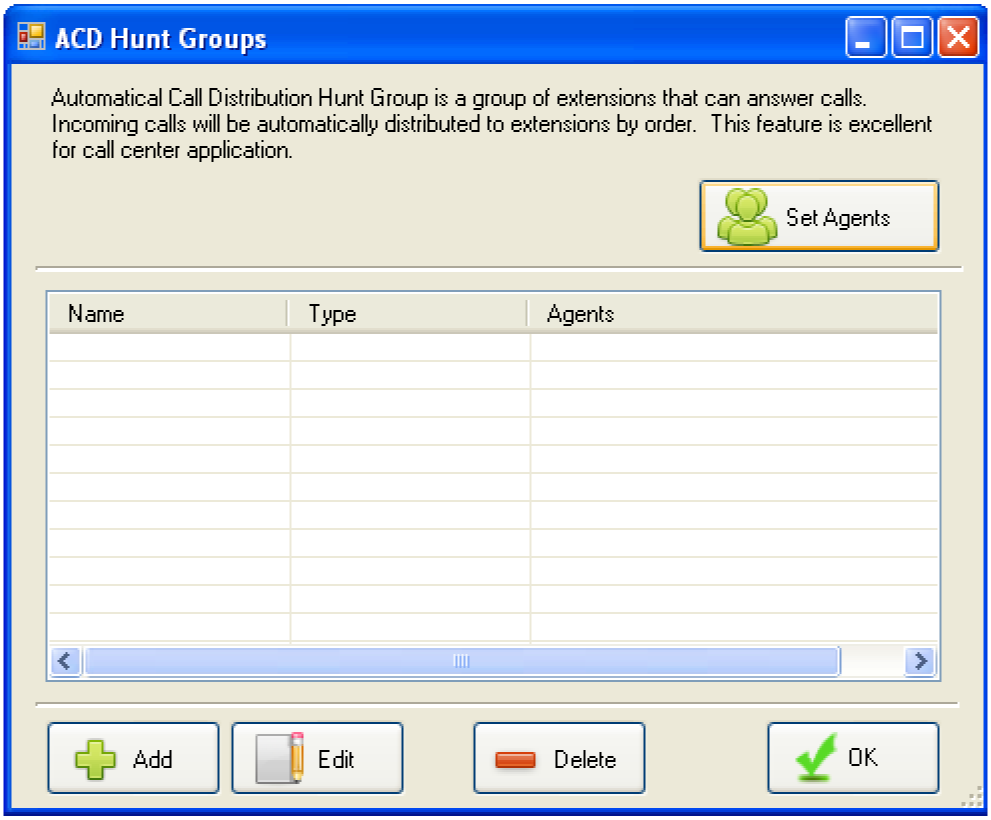


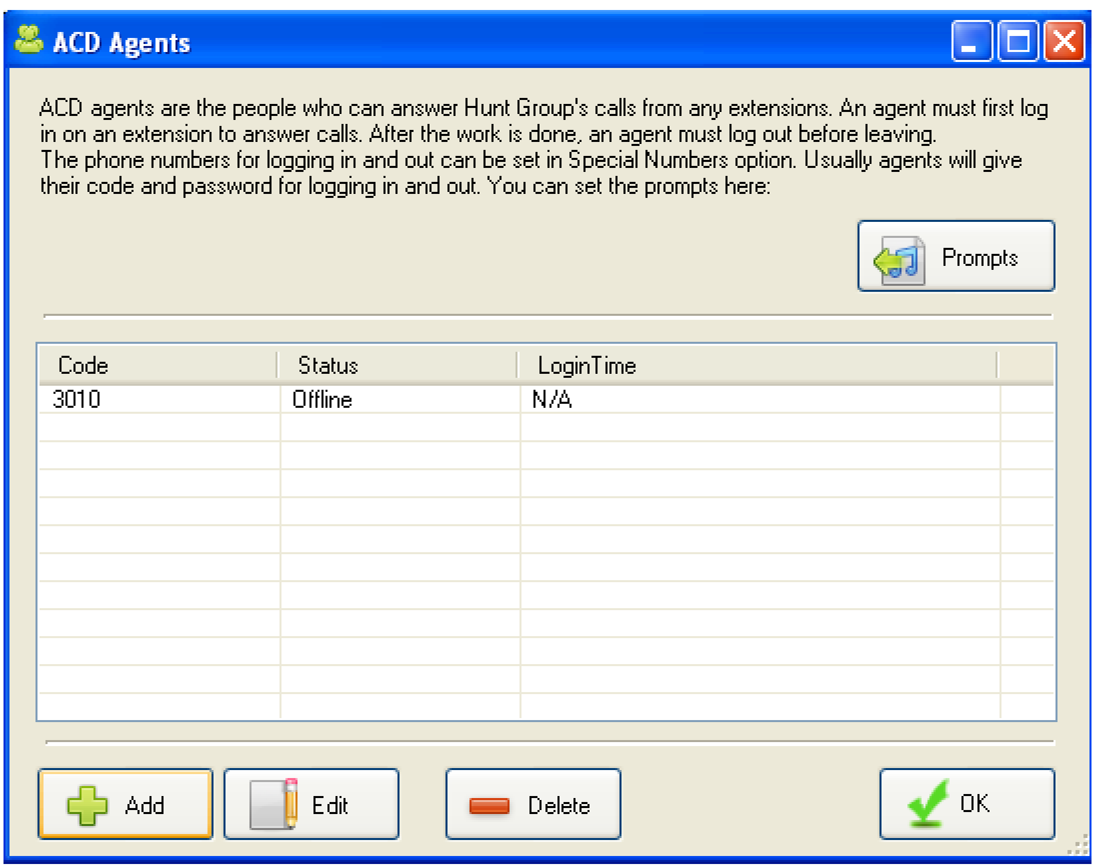


Agents can call above special login and logout numbers from any extension to indicate they are at that extension or not.

Steps to setup agents:



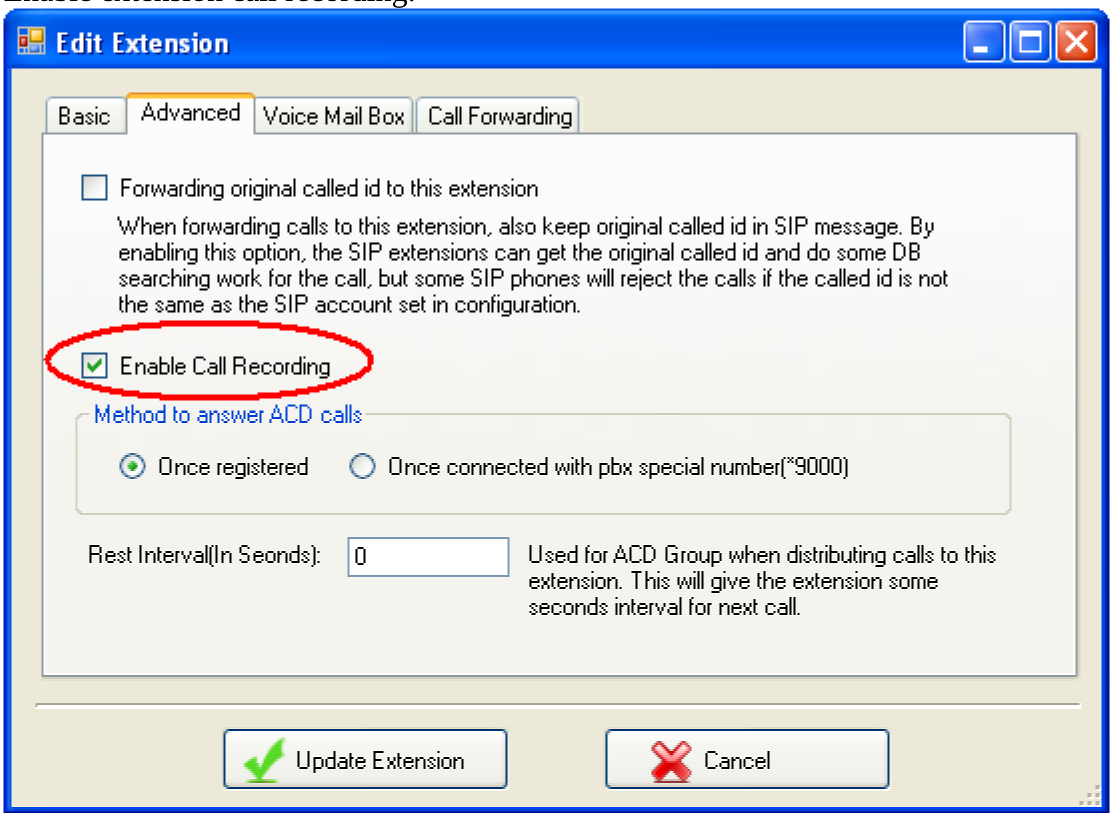

ACD Hunt Groups
Automatical Call Distribution Hunt Group is a group of extensions that can answer calls. Incoming calls will be automatically distributed to extensions by order. This feature is excellent for call center application.
Set Agents
Name
Type
Agents
Add
Edit
Delete
OK


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

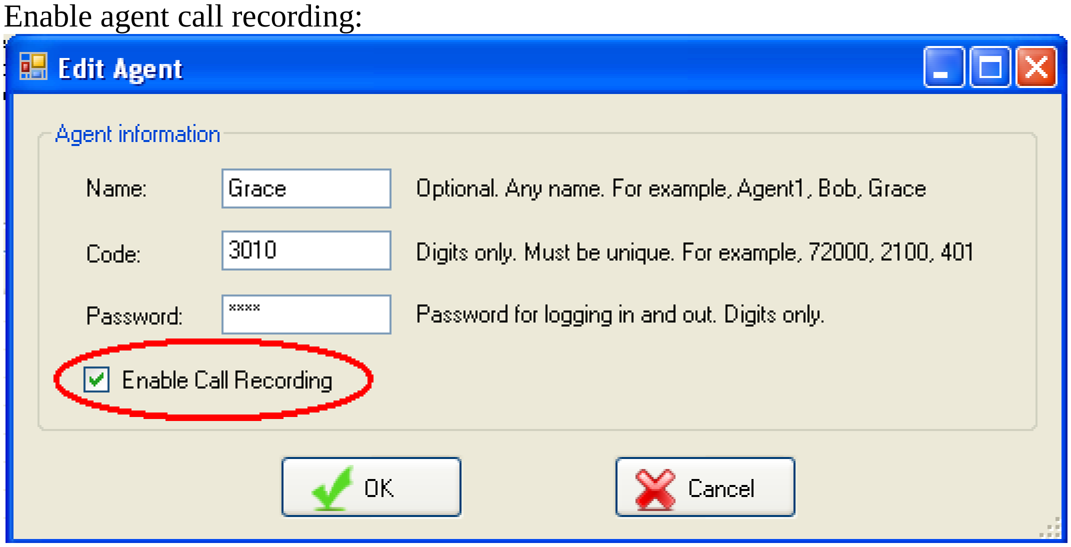
4.2 Enabling Call Recording

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

Enable extension call recording:



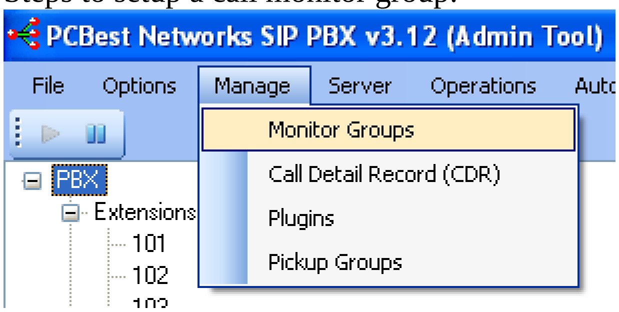
Enable agent call recording:

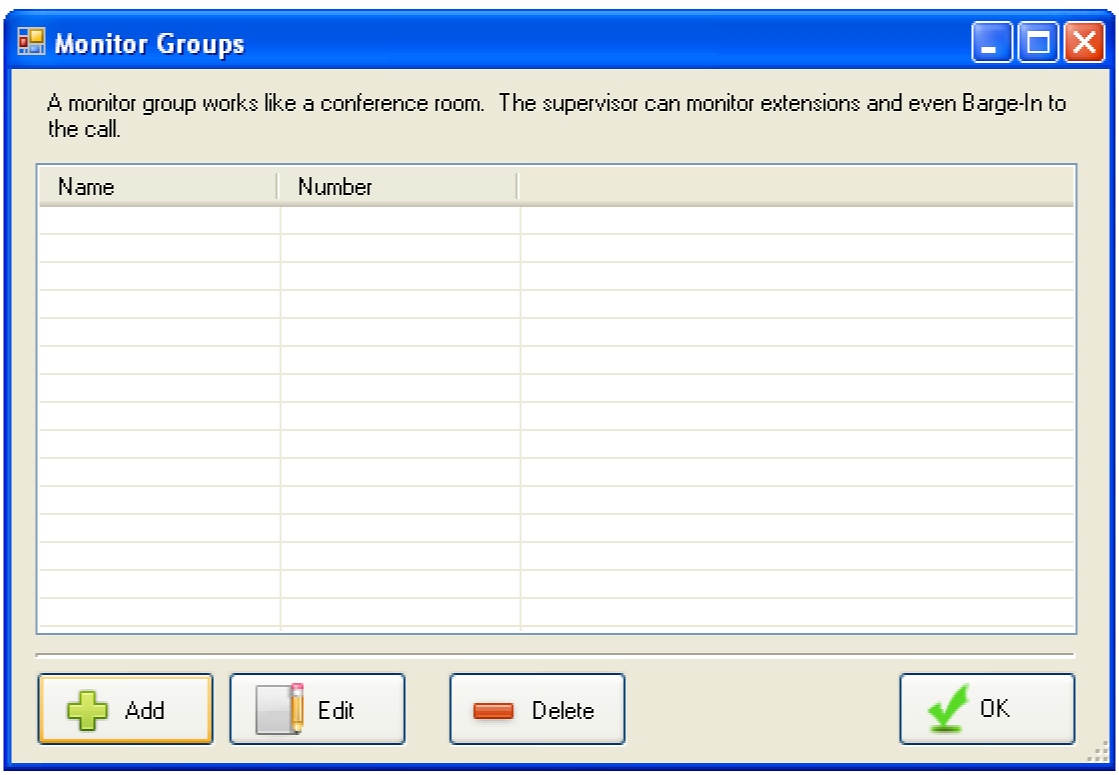


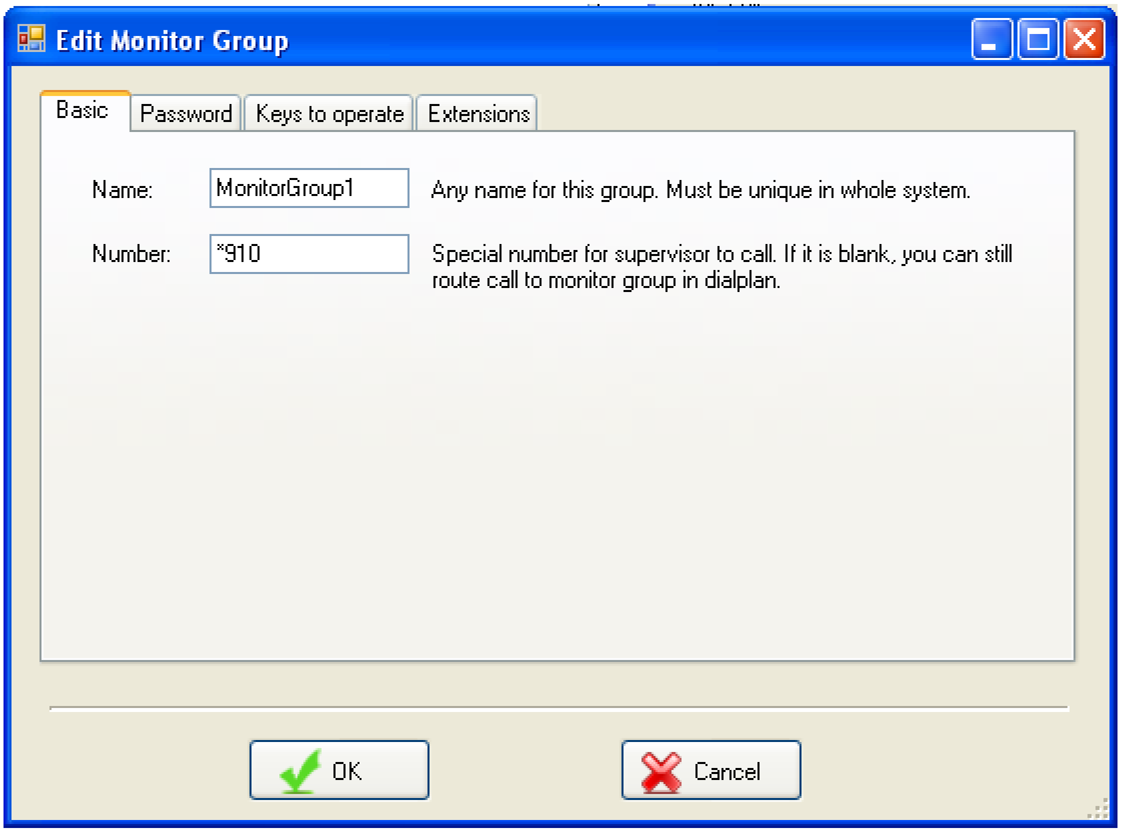
4.3 Supervisor Call Monitoring

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:



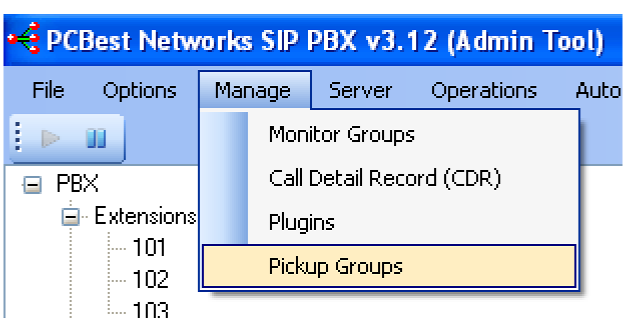


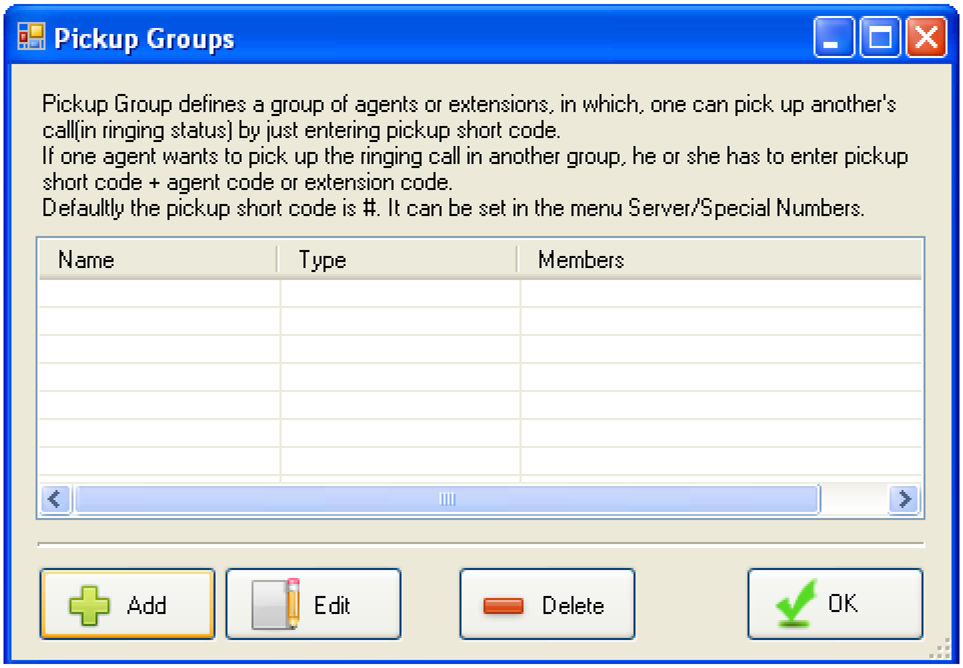


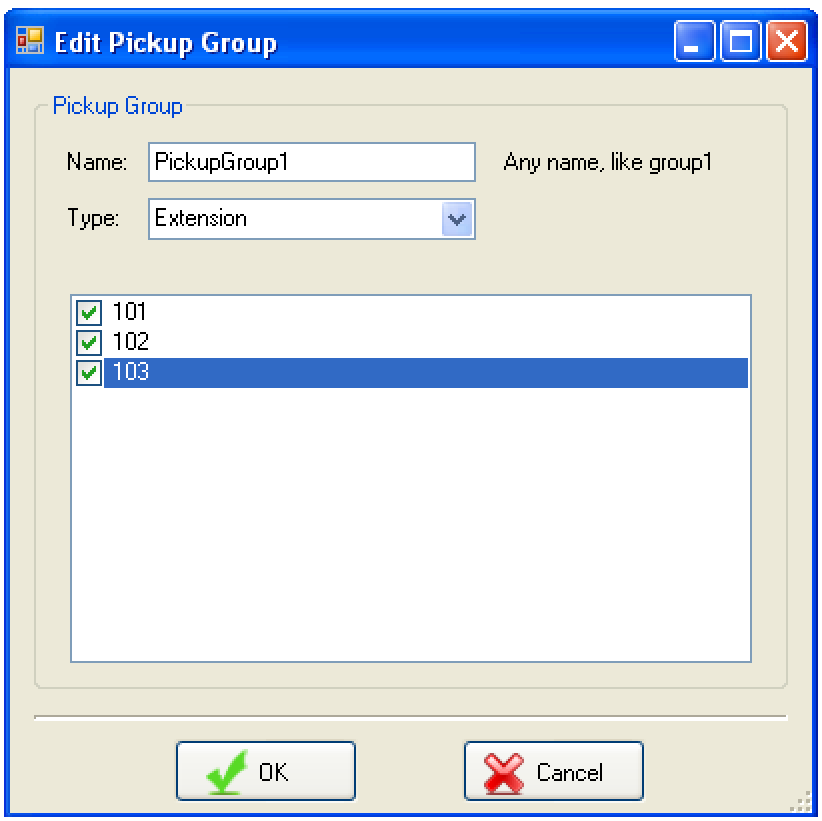
Once you defined a monitor group, please call monitor group number *910 from an supervisor type extension, you will be able to follow the IVR menu to monitor any other extensions.

4.4 Pickup Group

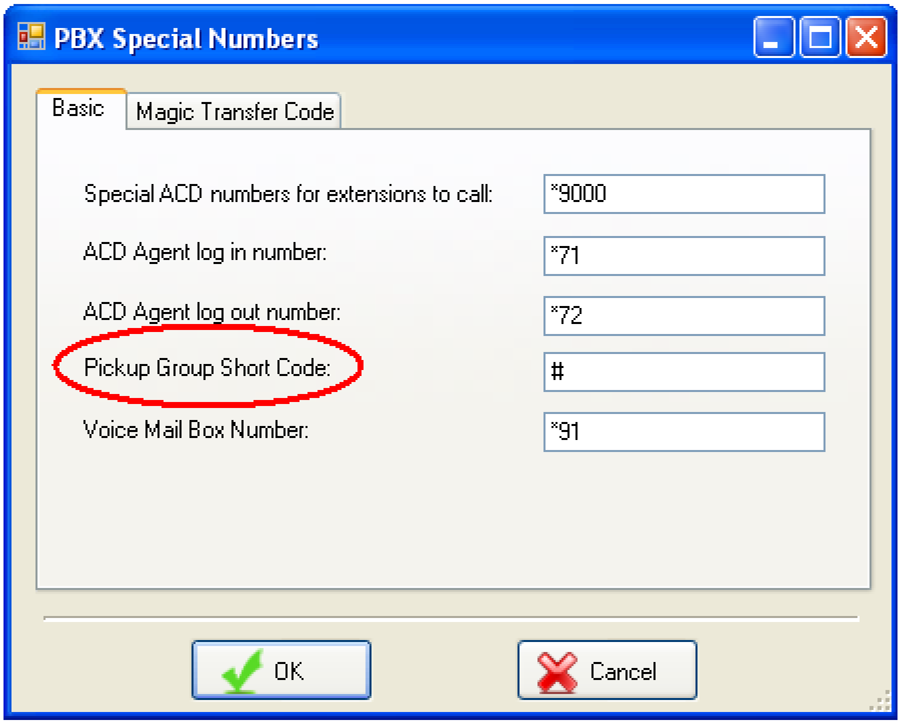
Pickup Group defines a group of agents or extensions, in which, one can pick up another's call(in ringing status) by just entering pickup short code.
If one agent wants to pick up the ringing call in another group, he or she has to enter pickup short code + agent code or extension code.
Defaultly the 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:

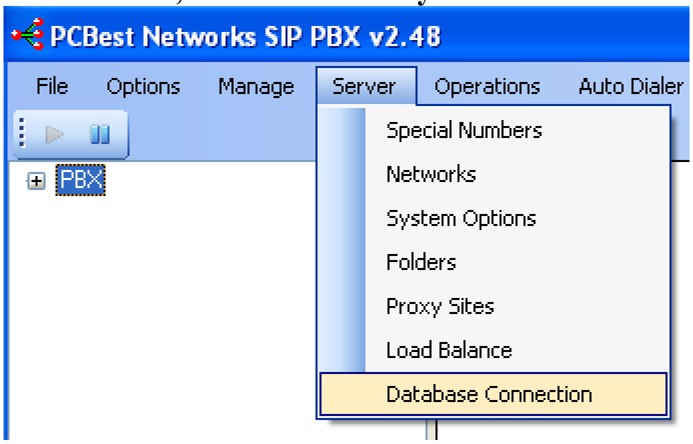
Auto Survey Calls: You can specify an auto dialer task which presents an IVR menu for the connected calls. Once the customer chose an option, then forward the call to another menu, and so on. The customer choices will be record into database like this: IVRMenu1,1;IVRMenu2,2;...

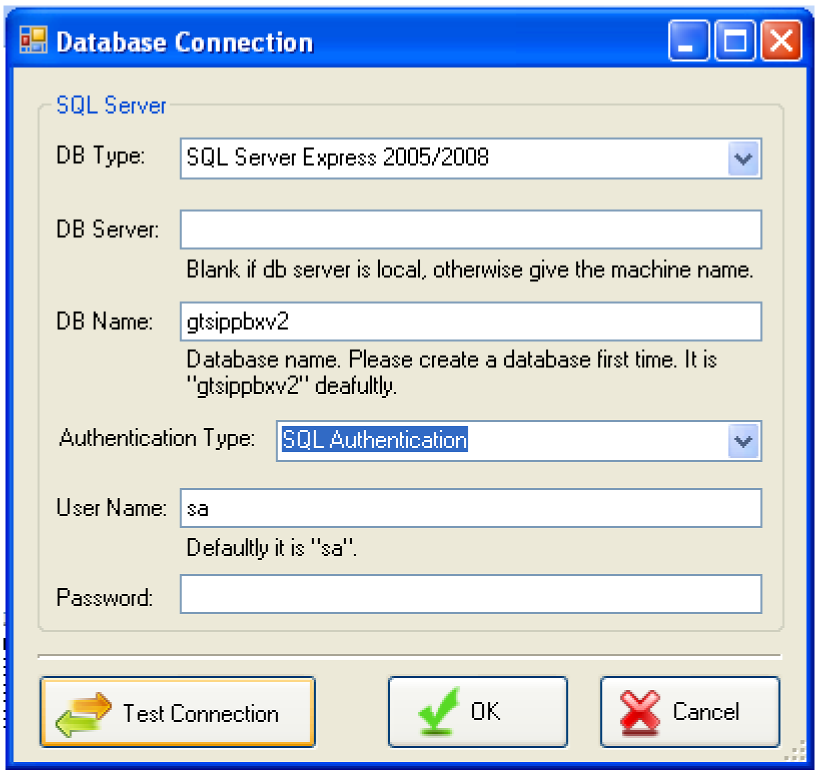
Call Me Back: Your customer can give a phone number to call back on your website. The phone number will be stored into PBX's auto dialer call jobs table. The pbx will call the number, and once the call is connected, then forward the call to an extension(or agent).

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

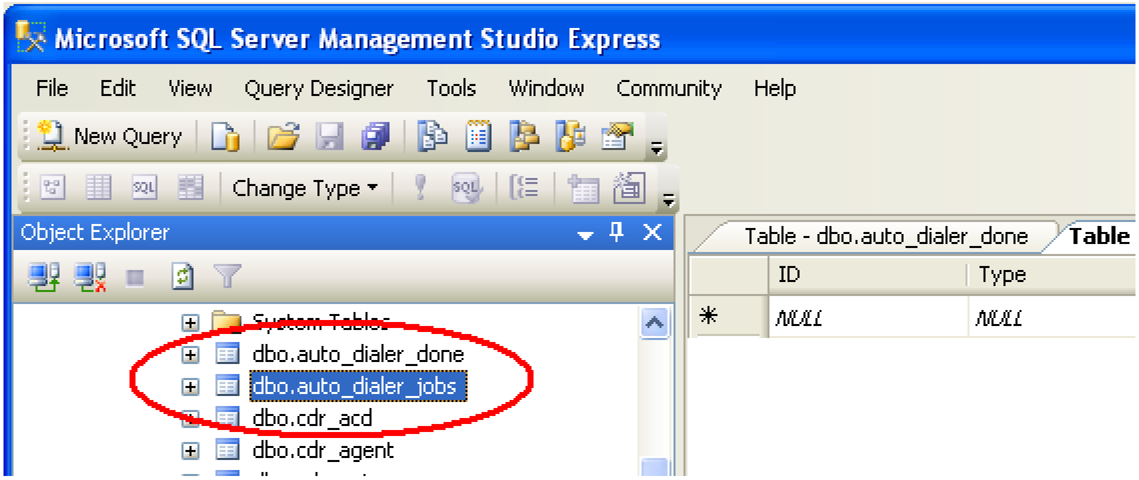
How does it work?

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



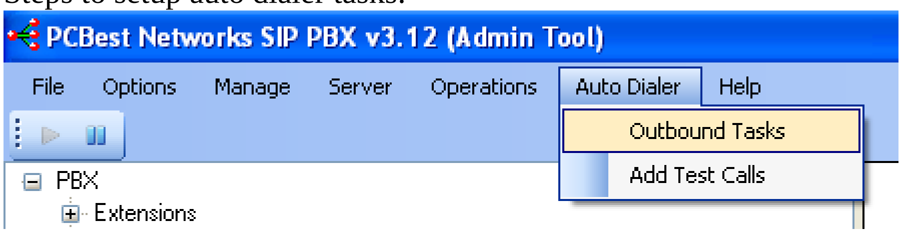


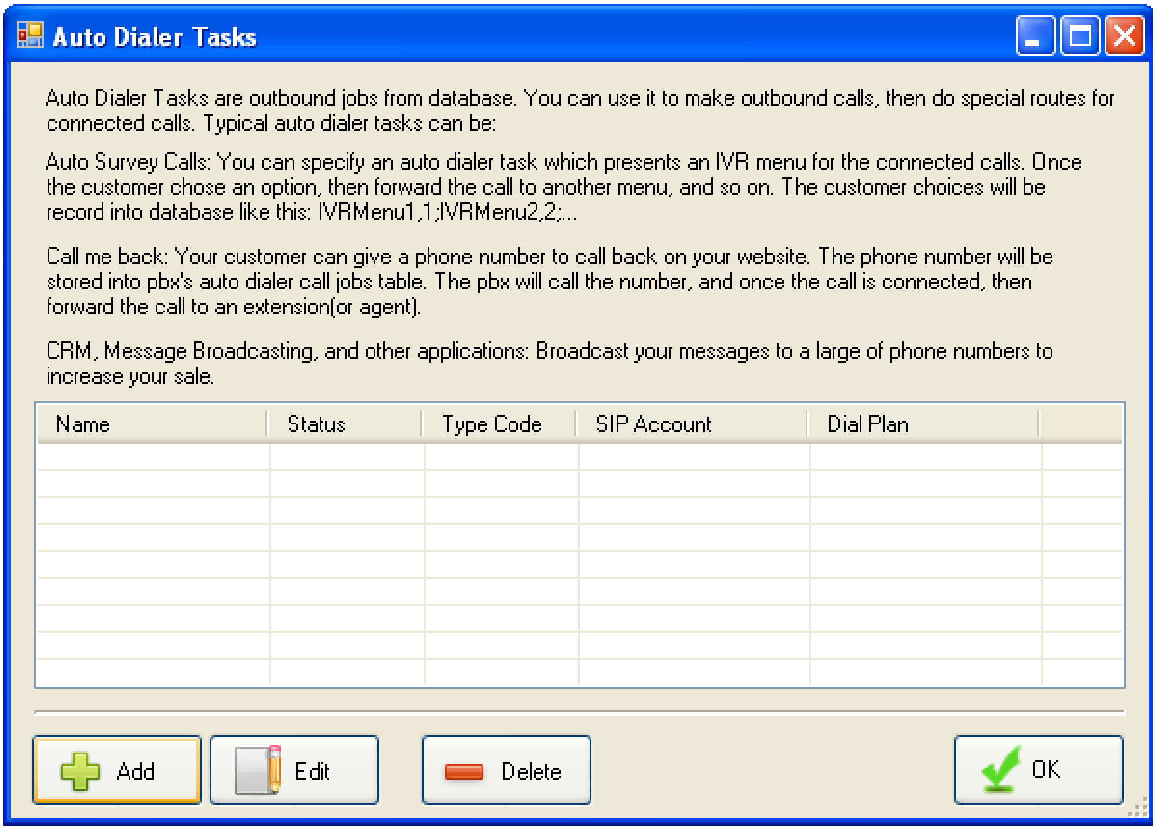
Once the PBX connected with the database, it will create some tables that it needs. Please look at two tables auto_dialer_jobs, and auto_dialer_done.

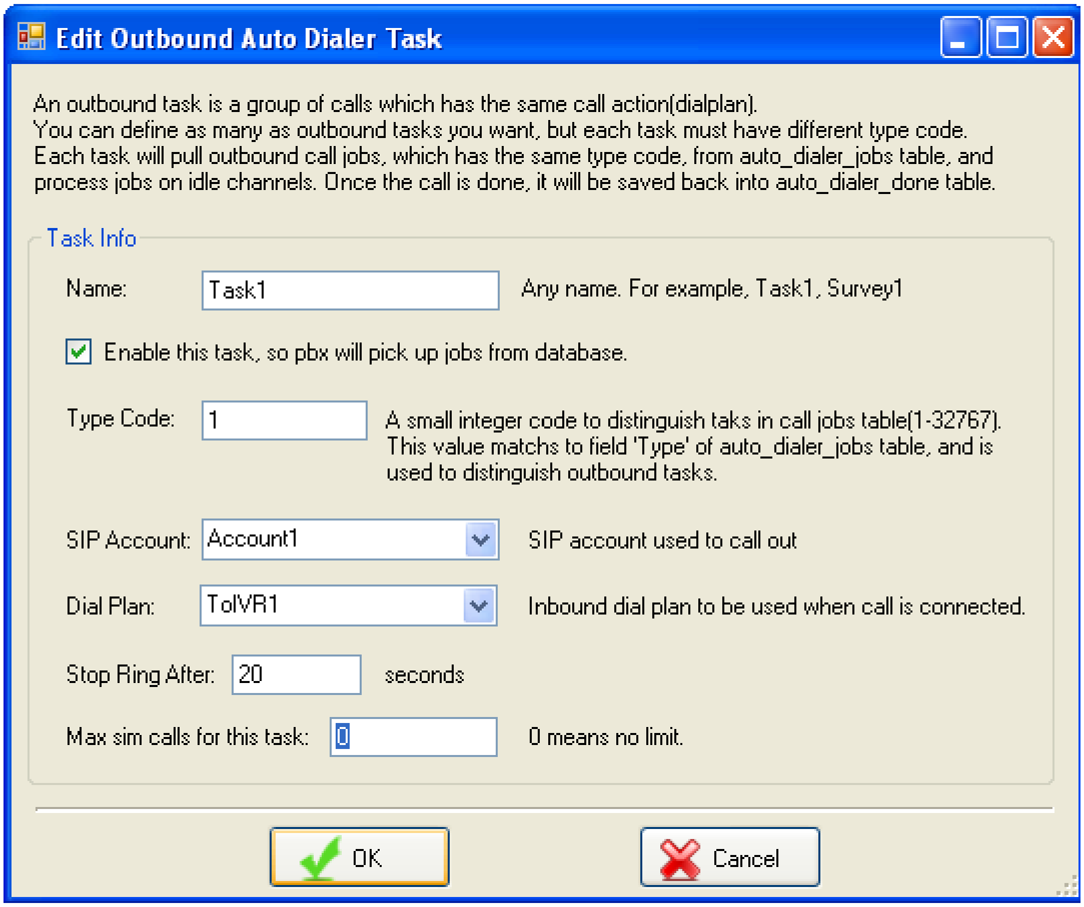


PBX will try to check auto_dialer_jobs every 2 seconds, to pull out outbound records, then dial the numbers out, then write the result back into auto_dialer_done table.

Steps to setup auto dialer tasks:

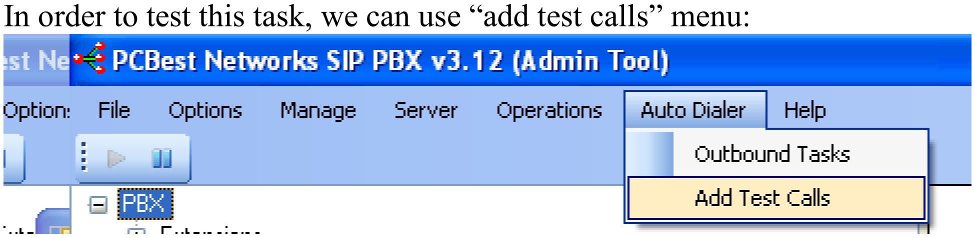


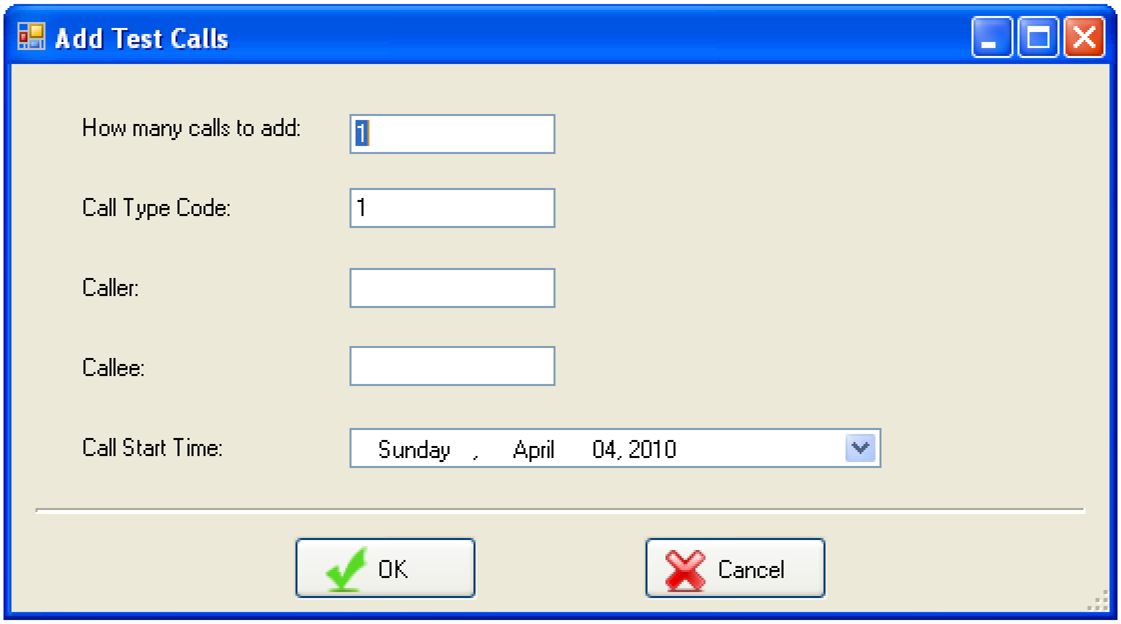

Auto Dialer Tasks
Auto Dialer Tasks are outbound jobs from database. You can use it to make outbound calls, then do special routes for connected calls. Typical auto dialer tasks can be:
Auto Survey Calls: You can specify an auto dialer task which presents an IVR menu for the connected calls. Once the 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:

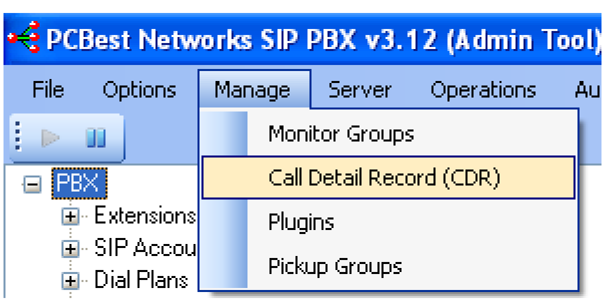




Give the type code 1, and caller and callee numbers, then click OK. PBX should be able to pick up the call job, and dial out to the number.

6 PBX Other Configurations

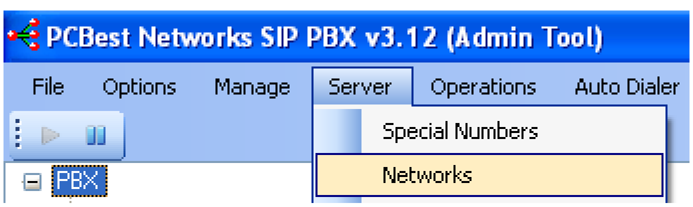
6.1 CDR



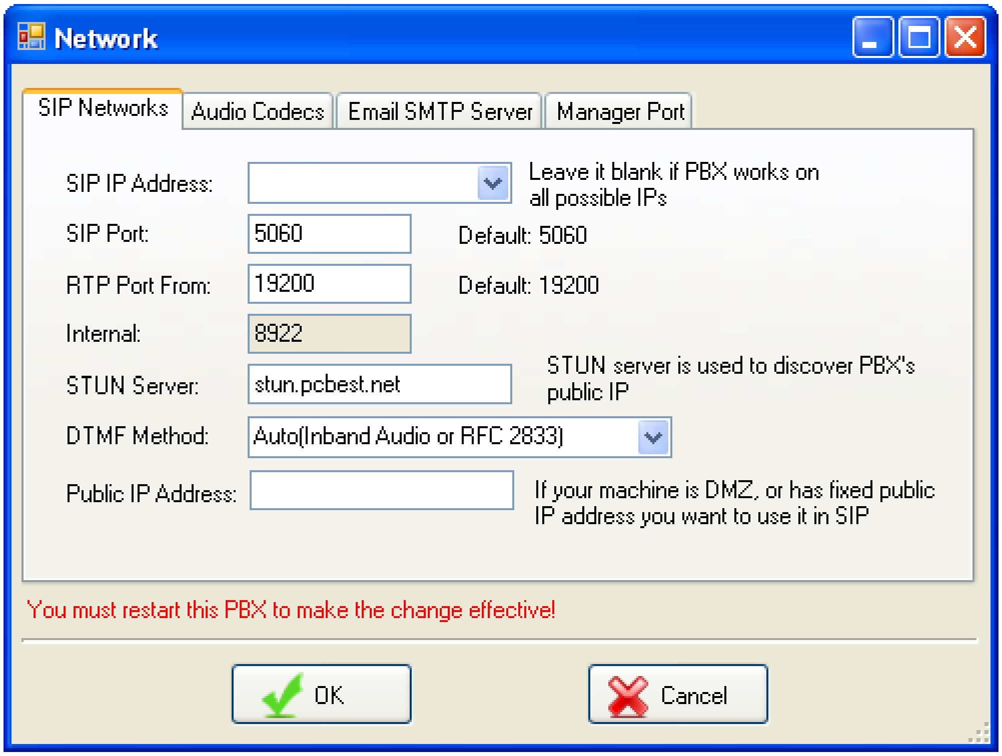
You can write CDR into database: (**Note** V2 must setup a database connection first)


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).

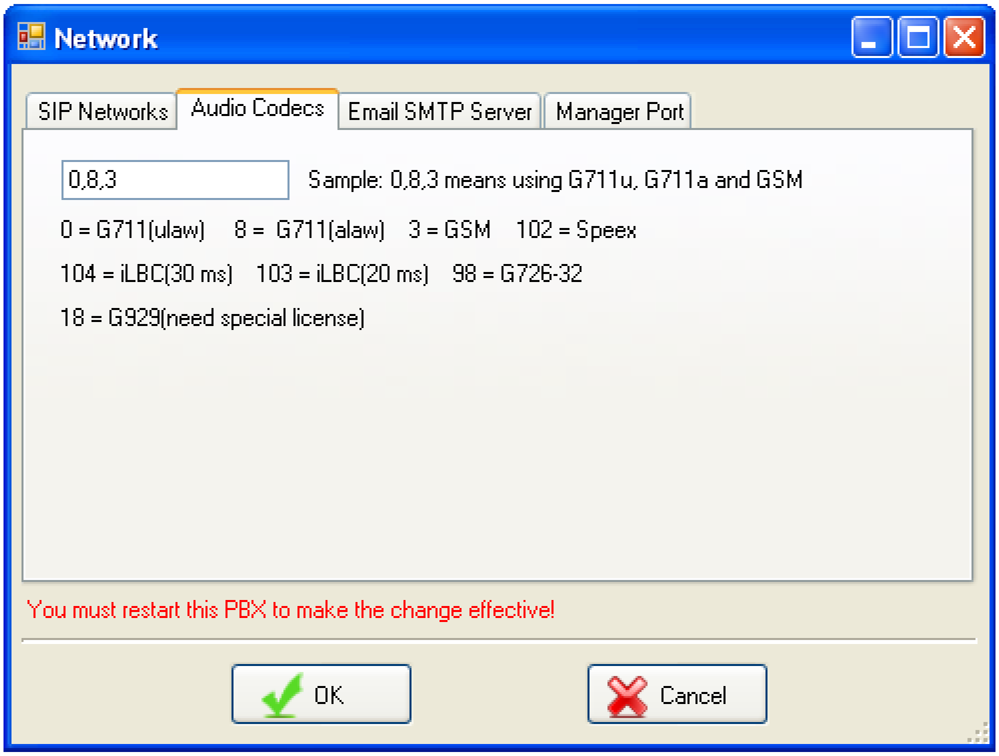
Internal: PBX uses this internal port for internal messages and events communication. It is not changeable.

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.

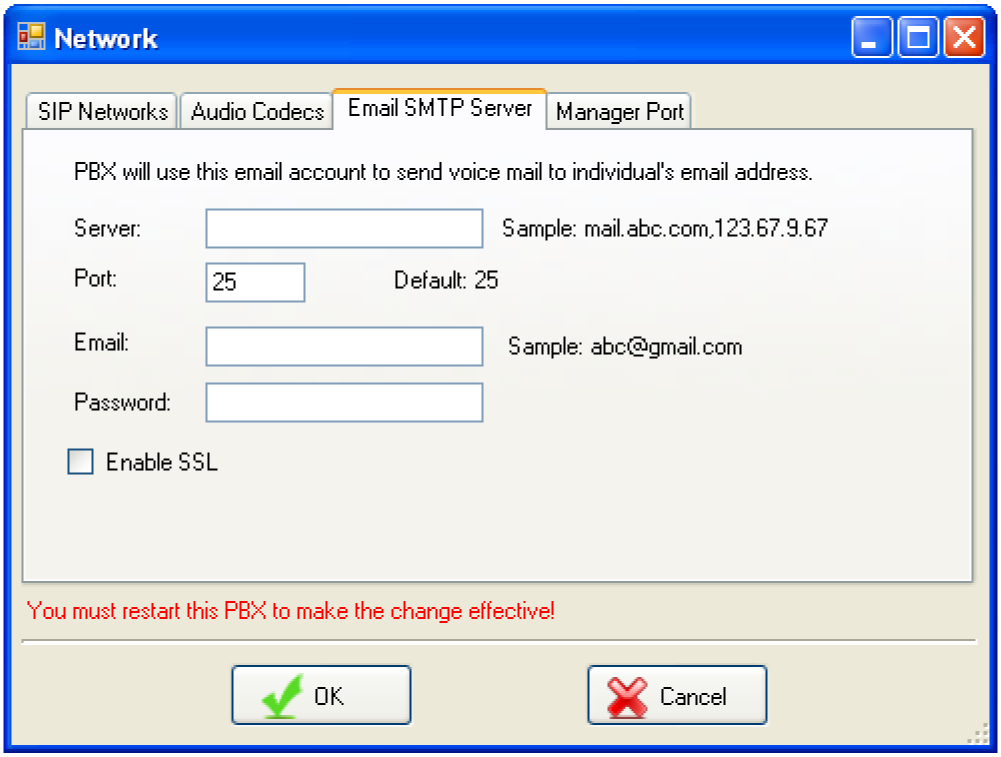
Public IP Address(V3 Only): In some case, for example, DMZ, you know your PBX are working on specific public IP address, so you can specify this field so PBX won't use internal IP address or ignore STUN server to get public IP.

Audio Codec Tab:



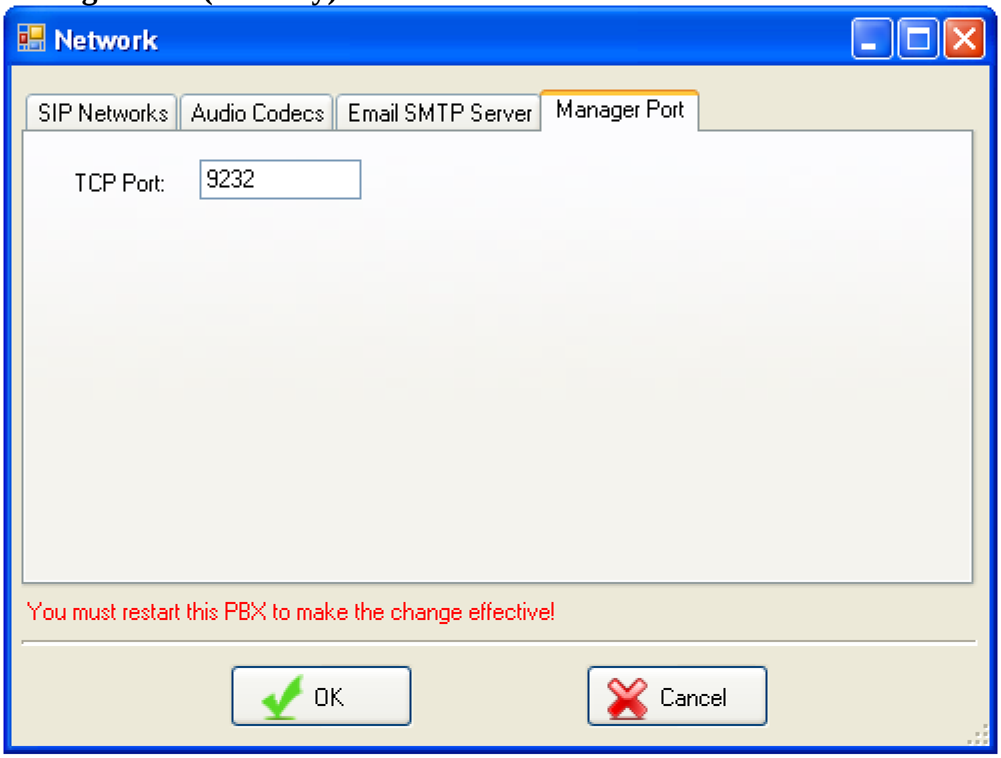
You can specify the PBX which audio codec in SIP SDP negotiation. When negotiating the audio codec, PBX will try to use the audio codec that is in the front of the list.
In above sample, the audio codec is 0,8,3. It means that g711 mulaw first, then g711 alaw, then GSM.

Email SMTP Server:



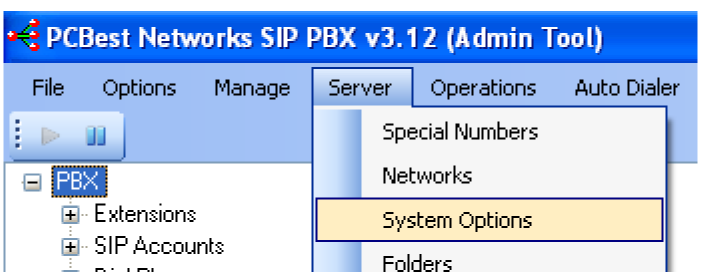
Server: Email server address. It can be an IP address or domain name.
Port: Email server port number.
Email: Email address that is used by PBX to send out email.
Password: Password for above email address.
Enable SSL: if it uses SSL.

Manager Port (V3 Only):

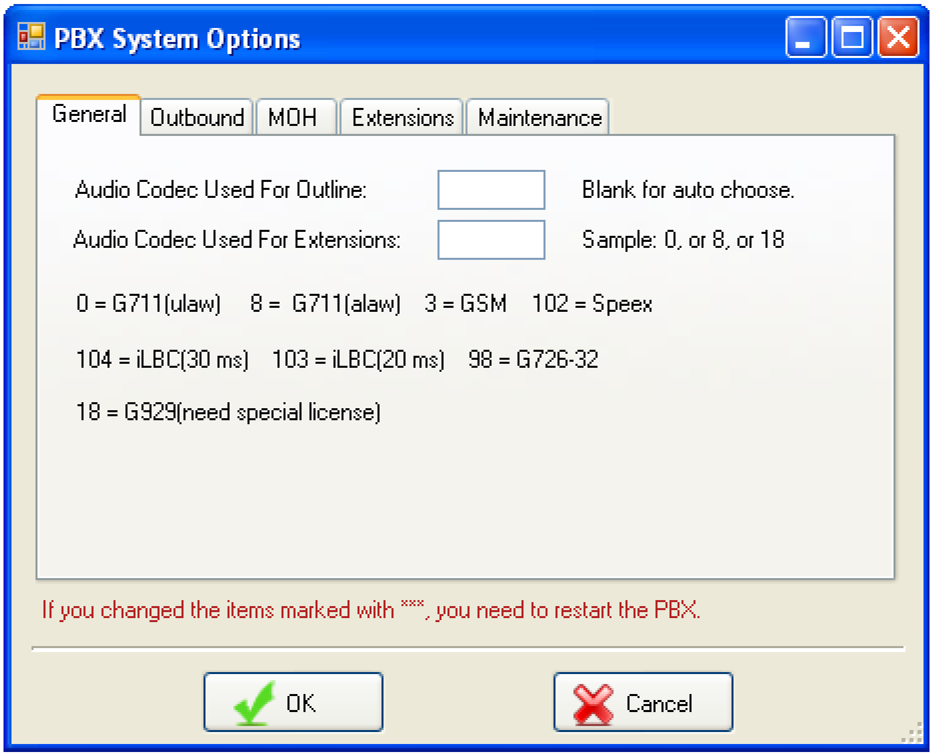


Manager port is used to for manager client to connect. PBX has a sample in SDK named "ManagerClient", which shows how to develop .NET application to receive events from PBX, or control PBX. Please refer to 7.2 about details.

6.3 System Options

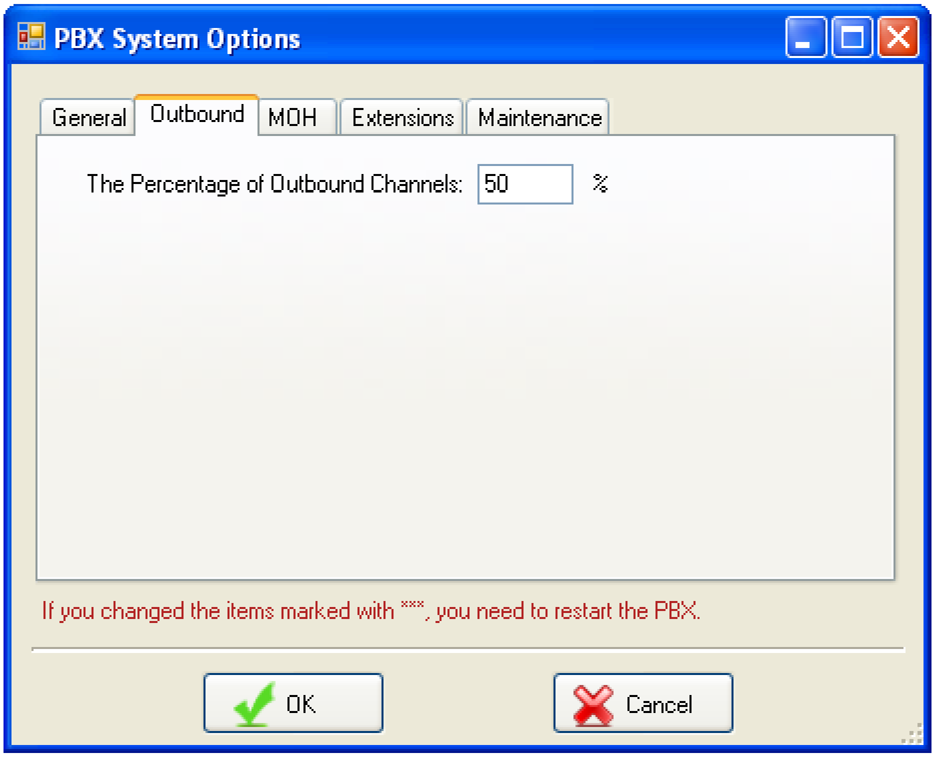


General Tab:



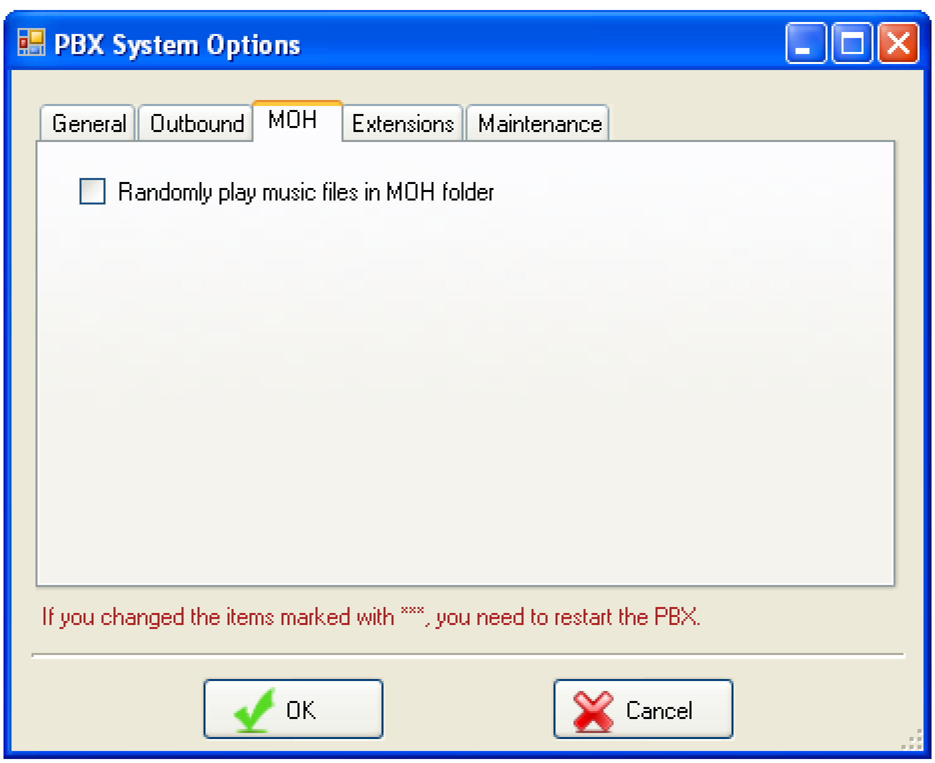
A typical example is that sometimes, you want low bandwidth audio codec using on the public network, but high quality audio codec on intranet.
Above dialog give you an option to specify the outline codec and internal codec.
For example, you can specify:
Outline: 18
Extensions: 0
It means PBX will do 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:

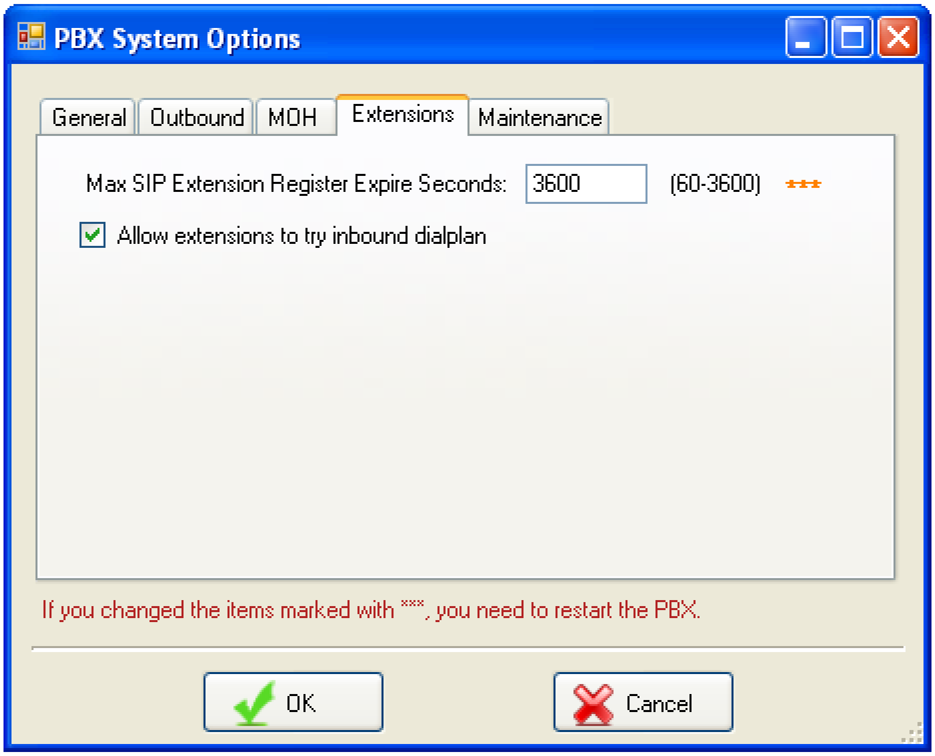


Percentage of outbound channels is for outbound calls. PBX default uses half channels for outbound, and keep half channels for inbound calls.

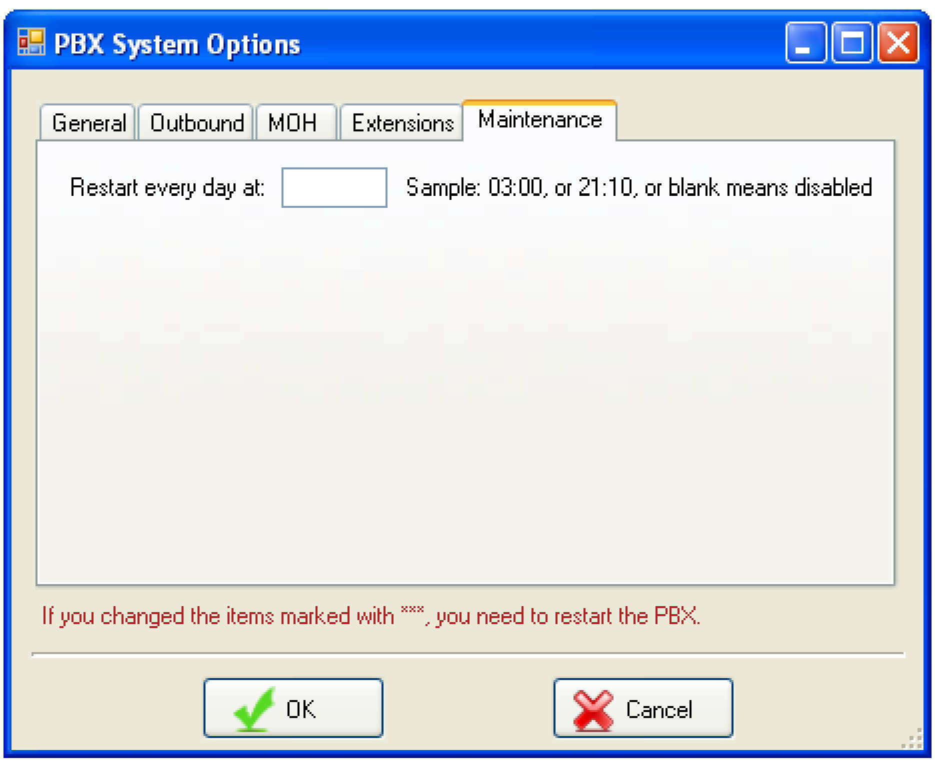
MOH Tab:



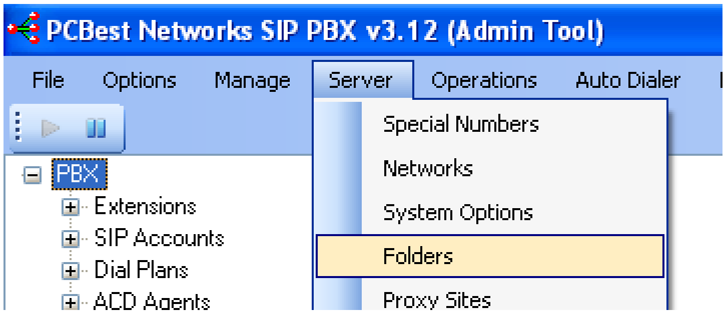
Extensions Tab:

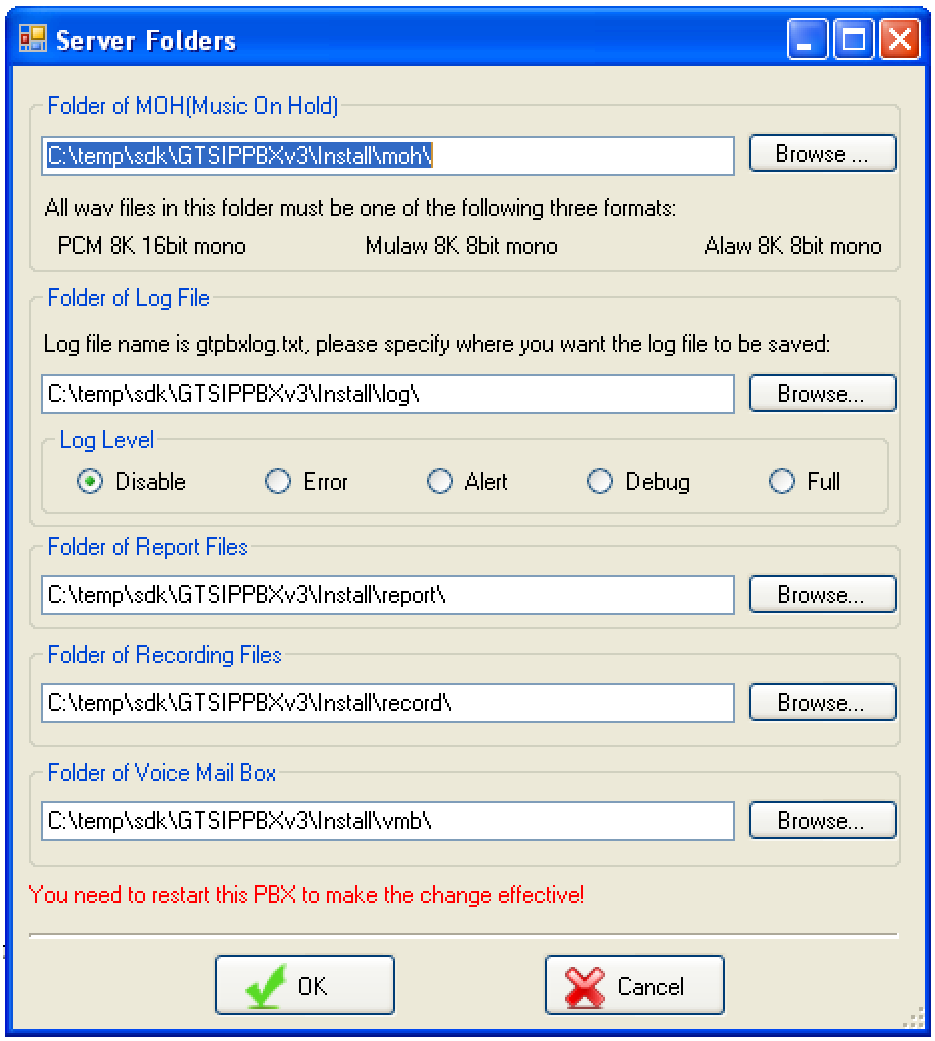


Maintenance:



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

Plugin allows you extend PBX's feature. Plugins are external dlls that exist in "plugin" sub folder, and are loaded when PBX starts. There are three C# plugin samples in pbx plugin sub folder, for three types of PBX plugins, "IVRMenu", "Init" and "Routine".

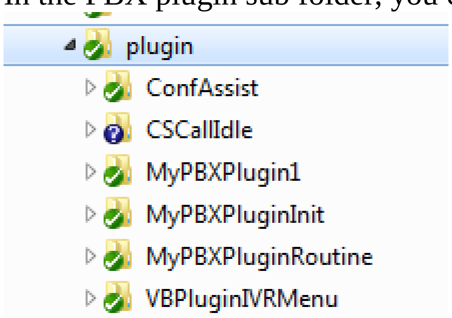
"IVRMenu" plugin is used to extend PBX's IVR feature. It allows you customize your own IVR menu, or do your special routes before it reaches extensions.

"Init" plugin is executed when pbx starts and stops. For example, you can use your own data from DB to set PBX parameters.

"Routine" plugin runs every one second, to let you do your own job for special purpose. For example, restarting PBX regularly, or adding more extensions.

There are also five call states plugin, when the call state changed.
"CallIdle"
"CallOffered"
"CallRinging"
"CallDialing"
"CallConnected"

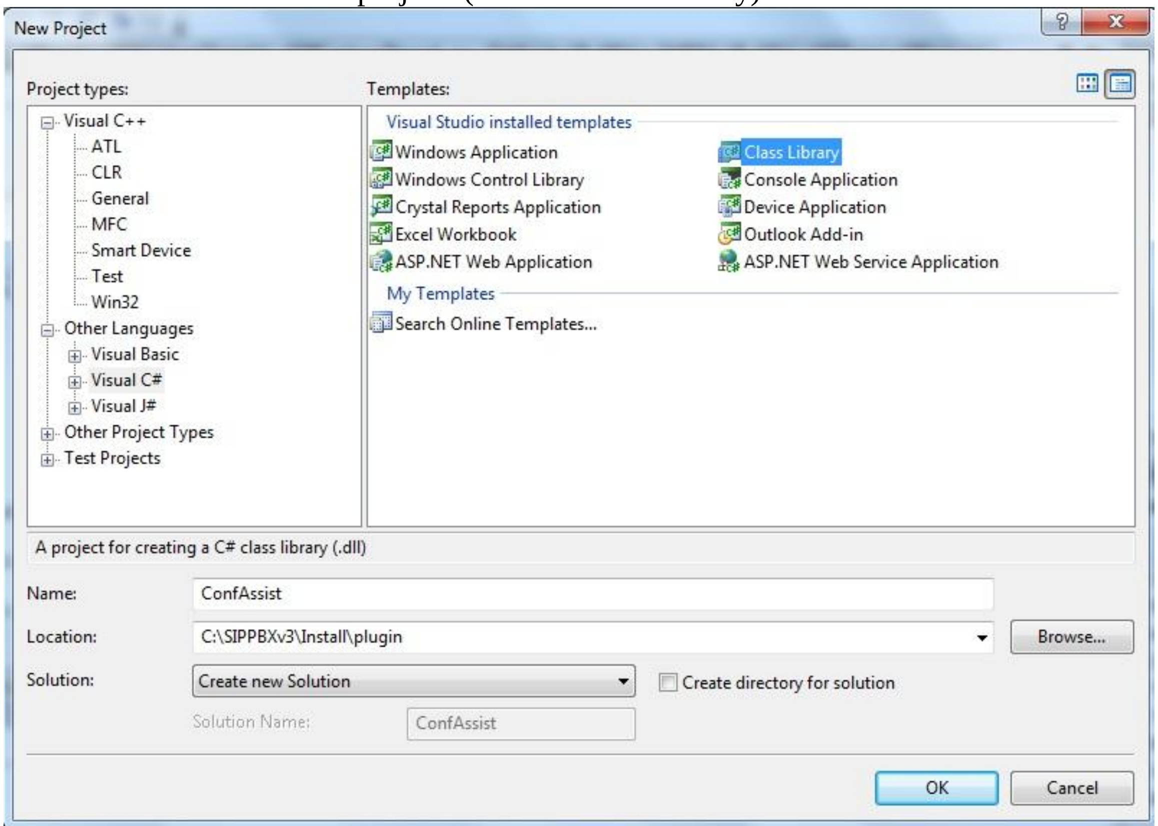
In the PBX plugin sub folder, you can find samples of plugin.



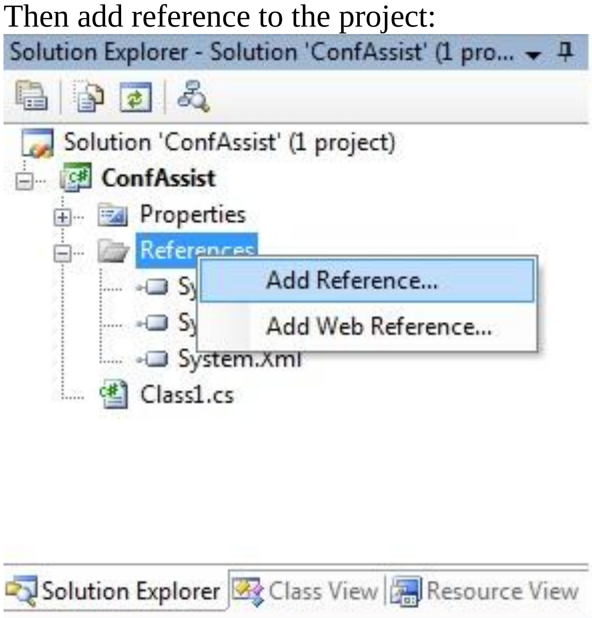
MyPBXPlugin1 is a sample for IVRMenu type plugin. C# code.
MyPBXPluginInit is a sample for Init type plugin. C# code.
MyPBXPluginRoutine is a sample for Routine type plugin. C# code.
VBPluginIVRMenu is a sample for IVRMenu type plugin in vb.net code.
CSCallIdle is a sample for call idle type plugin. C# code.
ConfAssist is a sample for IVRMenu type plugin which call advanced conference functions. C# code.

How to setup a plugin project?

1. New a vb.net or C# project: (Choose Class Library)

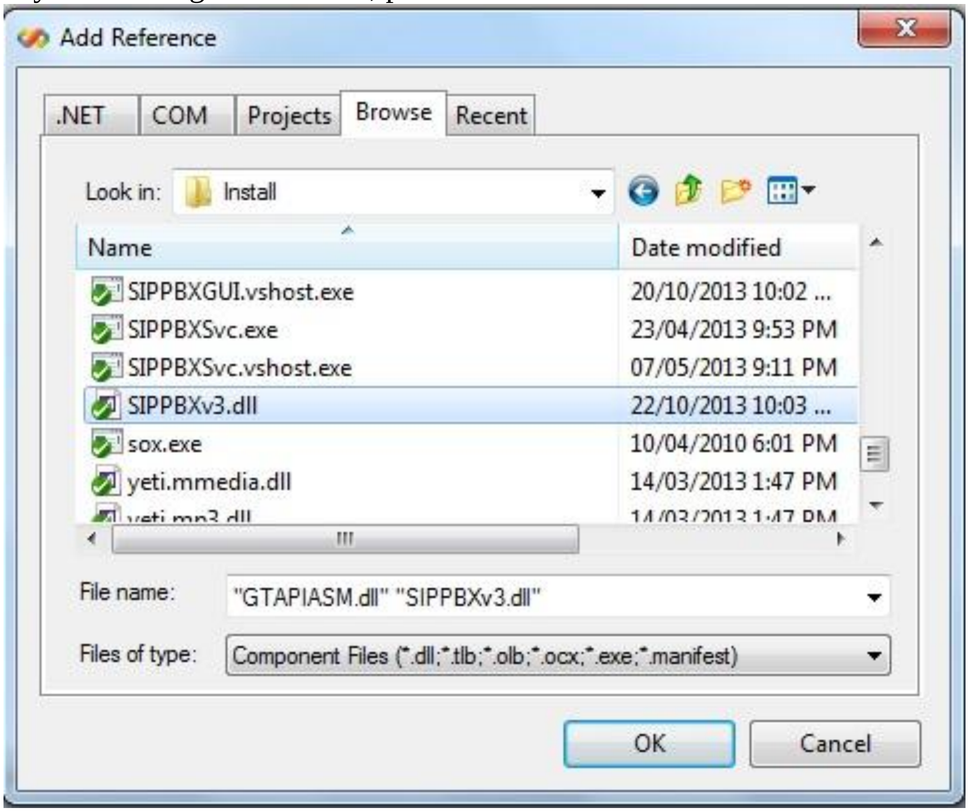


2. Then add reference to the project:



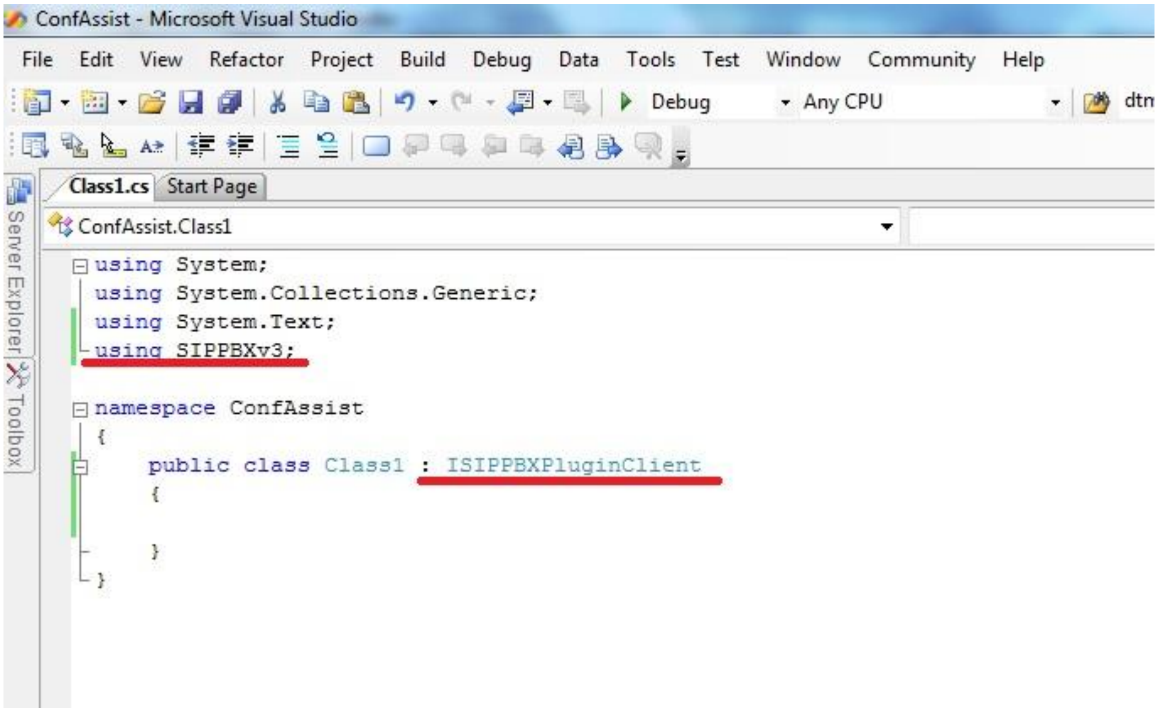
If you are using SIP PBX v2, please choose GTSIPPBX.exe in pbx installation folder.

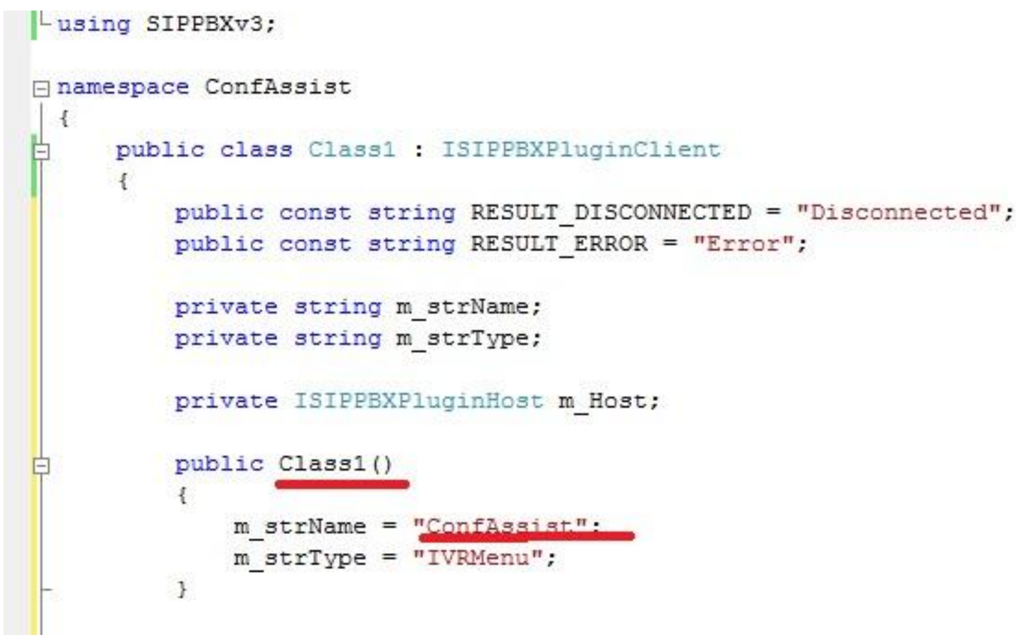
If you are using SIP PBX v3, please choose SIPPBXv3.dll.



Also, you need add GTAPIASM.dll as reference.
Note: DO NOT forget to add reference “System.Windows.Forms”.

3. Write a class which implements interface `ISIPPBXPluginClient`


ConfAssist - Microsoft Visual Studio
File Edit View Refactor Project Build Debug Data Tools Test Window Community Help
Debug
Any CPU
Server Explorer
Toolbox
Class1.cs
Start Page
ConfAssist.Class1
using System;
using System.Collections.Generic;
using System.Text;
using SIPPBXv3;
namespace ConfAssist
{
public class Class1 : ISIPPBXPluginClient
{
}
}

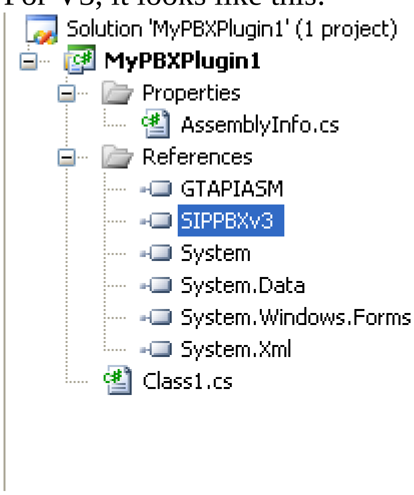

using SIPPBXv3;
namespace ConfAssist
{
public class Class1 : ISIPPBXPluginClient
{
public const string RESULT_DISCONNECTED = "Disconnected";
public const string RESULT_ERROR = "Error";
private string m_strName;
private string m_strType;
private ISIPPBXPluginHost m_Host;
public Class1()
{
m_strName = "ConfAssist";
m_strType = "IVRMenu";
}

```
public void Start()
{
    //get the caller and callee number
    string caller_addr = Host.Channel.caller_num;
    string callee_addr = Host.Channel.callee_num;

    string caller_num = GTAPIASM.GTAPIEnv.GetSIPAddressInfo(1, caller_addr);
    string callee_num = GTAPIASM.GTAPIEnv.GetSIPAddressInfo(1, callee_addr);

    //if it is a job of auto dialer task for human/answer machine detection,
    //use the following code to access detection result:
    /*
    if (Host.PBX_Channel.call_job != null)
    {
        switch (Host.PBX_Channel.call_job.DetectResult)
        {
            case 0: // = Answering Machine
                break;
```

Please refer to C# or VB.NET plugin sample code for this part.

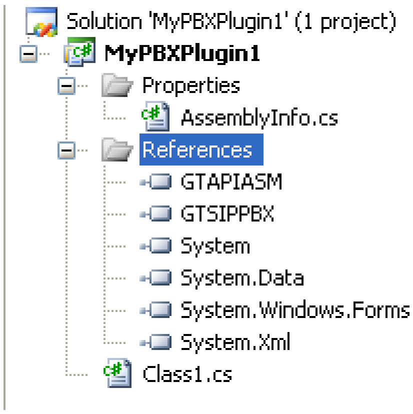
Let us open MyPBXPlugin1.

For V3, it looks like this:



Please change the references about GTAPIASM, and SIPPBXv3 if they are not available and pointing to right dlls.
SIPPBXv3 is SIPPBXv3.dll.

If you are using V2, it should looks like this:



GTSIPPBX refers to V2's GTSIPPBX.exe.

Please open class1.cs for less than 200 lines sample, which teaches you how to write the plugin IVR sample.

Plugin Built-in Methods and Functions

Host.DisplayMenu
Display a menu, and accept DTMF inputs.

Format:
```
string DisplayMenu(string audio_fn, int maxDigits, string termStr,
int timeOut);
audio_fn: Audio file name in full path.
maxDigits: The maximum digits to accept for the menu.
termStr: The string contains the digit which terminate the DTMF inputs.
In most of cases, it is "#".
timeOut: how long to wait. In milliseconds.
```

Return: DTMF string

Host.DisplayMenuEx
Display a multiple-audio menu, and accept DTMF inputs.

Format:
```
string DisplayMenu(List<string> audio_files, int maxDigits,
string termStr, int timeOut);
audio_files: Audio files in full path to be played.
maxDigits: The maximum digits to accept for the menu.
termStr: The string contains the digit which terminate the DTMF inputs.
In most of cases, it is "#".
timeOut: how long to wait. In milliseconds.
```

Return: DTMF string

Host.PlayAudio
Play an audio file.

Format:
```
string PlayAudio(string audio_fn, int maxDigits, string termStr,
int timeOut);
```

```
audio_fn: Audio file name in full path.
maxDigits: The maximum digits to accept for the menu.
termStr: The string contains the digit which terminate the DTMF inputs.
In most of cases, it is "#".
timeOut: how long to wait. In milliseconds.
```

Return: DTMF string

Host.PlayAudioEx
Play an audio file.

Format:
```
string PlayAudioEx(List<string> audio_files, int maxDigits,
string termStr, int timeOut);
audio_files: Audio files in full path to be played.
maxDigits: The maximum digits to accept for the menu.
termStr: The string contains the digit which terminate the DTMF inputs.
In most of cases, it is "#".
timeOut: how long to wait. In milliseconds.
```

Return: DTMF string

Host.RecordAudio
Record an audio file.

Format:
```
string RecordAudio(string audio_fn, int maxDigits, string termStr,
int timeOut);
audio_fn: Audio file name in full path.
maxDigits: The maximum digits to accept for the menu.
termStr: The string contains the digit which terminate the DTMF inputs.
In most of cases, it is "#".
timeOut: how long to wait. In milliseconds.
```

Return: DTMF string

Host.DetectDTMF
Detect DTMF keys.

Format:
```
string DetectDTMF(int maxDigits, string termStr, int timeOut);
maxDigits: The maximum digits to accept for the menu.
termStr: The string contains the digit which terminate the DTMF inputs.
In most of cases, it is "#".
timeOut: how long to wait. In milliseconds.
```

Return: DTMF string

Host.HangUp
Disconnect call.

Format: `int HangUp();`

Return: none

Host. WriteLog
Write a log information in the PBX GUI output and log.

Format: `WriteLog(string logInfo);`
`logInfo: the log text.`

Return: none

Host.ToExtension
Transfer this call to extension.

Format: `bool ToExtension(string exten_no);`
`exten_no: the extension number`

Return: bool, if succeed.

Host.ToIVRMenu
Send this call to IVR menu.

Format: `bool ToIVRMenu(string menu_name);`
`menu_name: the IVR menu name defined in PBX.`

Return: bool, if succeed.

Host.ToMonitorGroup
Send this call to monitor group.

Format: `bool ToMonitorGroup(string mg_name);`
`mg_name: the monitor group name defined in PBX.`

Return: bool, if succeed.

Host.ToHuntGroup
Send this call to ACD group.

Format: `bool ToHuntGroup(string acd_name, bool set_front);`
`acd_name: the hunt group name defined in PBX.`
`set_front: if set the call to the front of group so it can be answered immediately.`

Return: bool, if succeed.

Host.ToRingGroup
Send this call to ring group.

Format: `bool ToRingGroup(string rg_name);`
`rg_name: the ring group name defined in PBX.`

Return: bool, if succeed.

Host.ToVoiceMailBox
Send this call to voice mail box.

Format: `bool ToVoiceMailBox(string exten_no);`
`exten_no: the extension number of which voice mail box being used.`

Return: bool, if succeed.

Host.ToConferenceRoom
Send this call to a conference room.

Format: `bool ToConferenceRoom(string conf_name);`
`conf_name: conference room name defined in PBX.`

Return: bool, if succeed.

Host.ToPlugin
Send this call to another plugin.

Format: `bool ToPlugin(string plugin_name);`
`plugin_name: another plugin's name.`

Return: bool, if succeed.

Host.ToNumber
Forward this call to another phone number.

Format: `bool ToNumber(string number, SIPAccount sip_acct);`
`number: the number to forward.`
`sip_acct: sip account to use for this call`

Return: bool, if succeed.

Sample:

```
//to another outside number
//Host.ToNumber("<sip:123@192.168.1.100>", null);
//Host.ToNumber("<sip:6781992@callcentric.com>", null);
//or
//SIPAccount acct1;
//acct1.DisplayName = "any";
//acct1.UserName = "1234";
//acct1.DomainServer = "sip.callwithus.com";
//acct1.ProxyServer = "sip.callwithus.com";
//acct1.AuthName = "1234";
```

//acct1.Password = "xxxxx";
//Host.ToNumber("655112", acct1);

Host.DisconnectExtension
Disconnect(hang up) extension's call.

Format: bool `DisconnectExtension(string exten_no);`
`exten_no: extension number.`

Return: bool, if succeed.

Host.SetChanRunPlugin
Set channel to run another plugin.

Format: `bool SetChanRunPlugin(int ch, string plugin_name);`
```
ch: channel number.
plugin_name: the name of plugin.
```

Return: bool, if succeed.

Host.ResetChannel
Reset the channel. Disconnect the call if there is a call on the channel.

Format: `bool ResetChannel(int ch);`
`ch: channel number.`

Return: bool, if succeed.

Host.SetChanInConferenceRoom
Set channel into conference room.

Format: `bool SetChanInConferenceRoom(int ch, string conf_name, int opt);`
```
ch: channel number.
conf_name: conference room.
opt: 0 = take out of conference room. 1 = add into conference room. 2 =
monitor(listening only, not speaking)
```

Return: bool, if succeed.

Host.CreateConferenceRoom
Create a conference room

Format: `SIPConferRoom CreateConferenceRoom(string conf_name);`
`conf_name: conference room.`

Return: bool, if succeed.

Host.DestroyConferenceRoom
Destroy a conference room

Format: `void DestroyConferenceRoom(string conf_name);`
```
conf_name: conference room.
```

Return: bool, if succeed.

Host.GetConferenceRoomIndex
Get conference room index.

Format: `int GetConferenceRoomIndex(string conf_name);`
```
conf_name: conference room.
```

Return: the index of conference room.

Host.GetConferenceRoomHandle
Get conference room handle.

Format: `ulong GetConferenceRoomHandle(string conf_name);`
```
conf_name: conference room.
```

Return: the handle of conference room.

Host.GetConferenceRoomByName
Get conference room handle.

Format: `SIPConferRoom GetConferenceRoomByName(string conf_name);`
```
conf_name: conference room.
```

Return: the class of conference room.

Host.SetUserObj
Set user object for application, in order to retrieve it later

Format: `bool SetUserObj(int idx, object obj);`
```
idx: index of the object, based on 0.
obj: the object.
```

Return: if succeed.

Host.GetUserObj
Set user object for application, in order to retrieve it later

Format: `object GetUserObj(int idx);`
```
idx: index of the object, based on 0.
```

Return: the object

Host.GetChanUserObj
Get channel's object

Format: `object GetChanUserObj(int ch, int idx);`

```
ch: channel index based on 0.
idx: index of the object
```

Return: the object

Host.SetChanUserObj
Set channel's object

Format: `bool SetChanUserObj(int ch, int idx, object obj);`

```
ch: channel index based on 0.
idx: index of the object
obj: object
```

Return: if succeed.

Host.StartPBX
Start PBX

Format: `void StartPBX();`

Return: none.

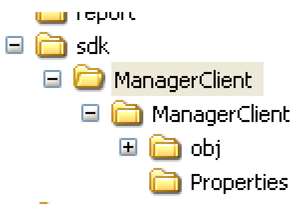
Host.StopPBX
Stop PBX

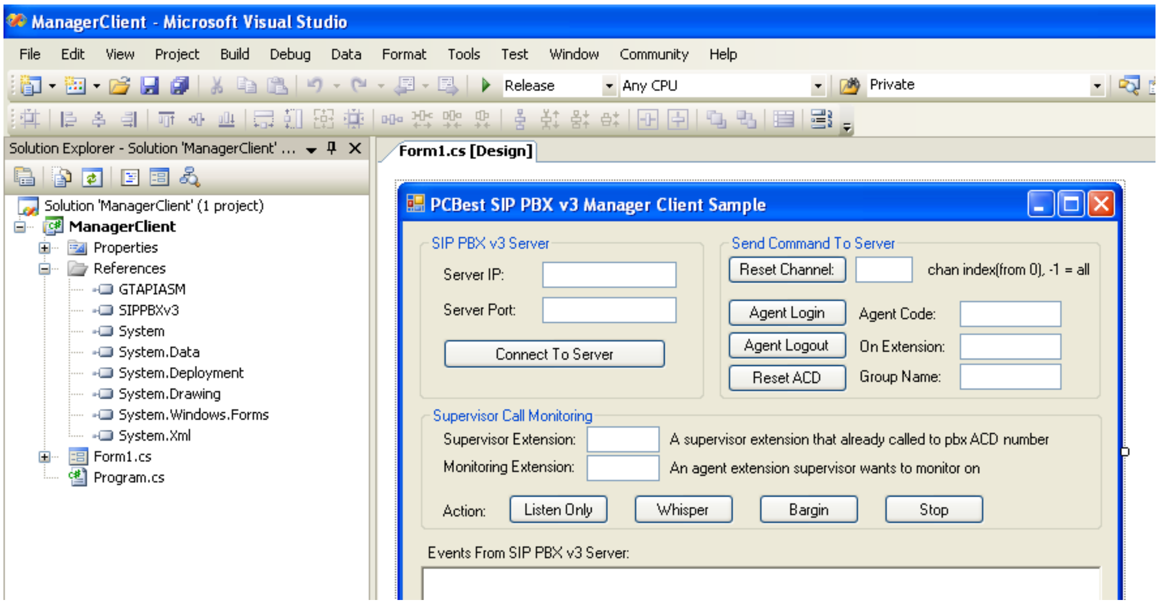
Format: `void StopPBX();`

Return: none

7.2 Manager Client Application (V3 only)

Manager client application is used for agent desktop computer to receive additional call information, or manager to control the PBX. Please open PBX SDK subfolder, you will see the a full source code of manage client application.





Currently manage client can receive those events:
Call events on each channel.
Registration events of extensions.
Agent login and logout event.

Manage client can also do those actions:

1. ***Connect/Disconnect to PBX server.***
2. ***Reset channels.***
3. ***Reset ACD group.***
4. ***Agent login and logout.***
5. ***Supervisor monitors extension.***
6. ***Dial a number for extension***
7. ***Make, Answer, and Hang up call on specific channel***
8. ***Hold and Transfer call on specific channel***
9. ***Run plug-in on the specific channel***
10. ***Do magic transfer for specific channel***
11. ***Extension status, Channel Status, and Agent Status events.***

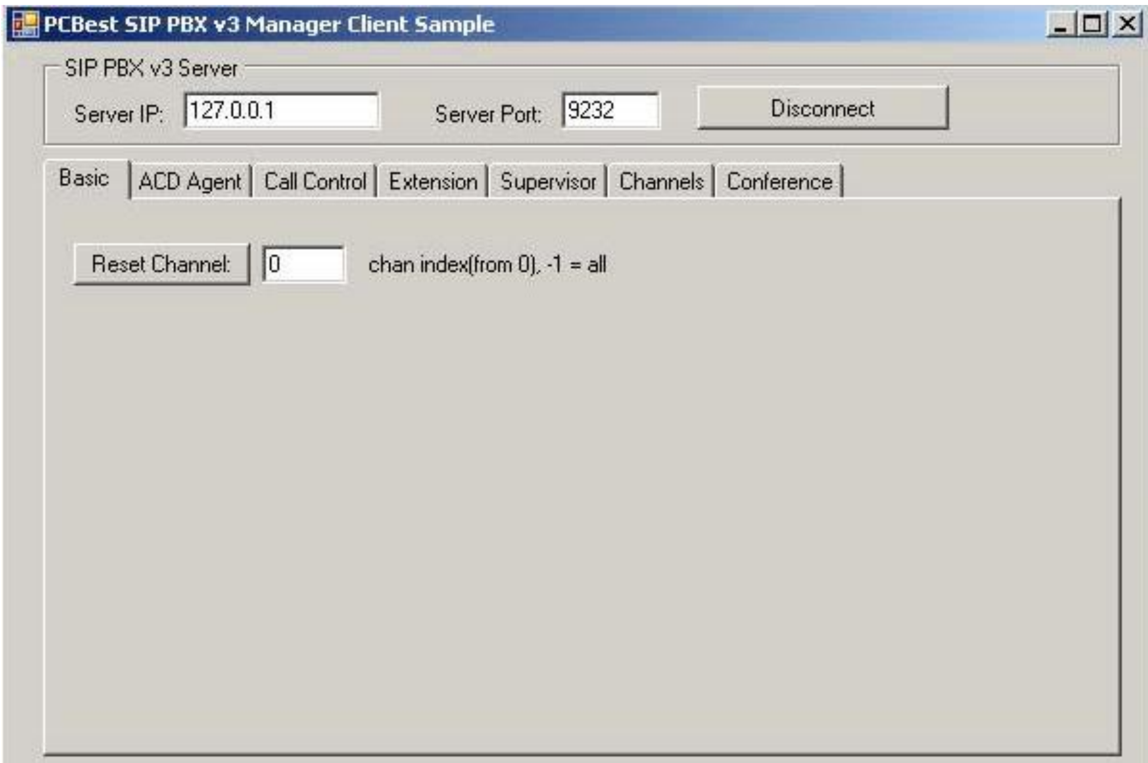
Methods and Events

ServerConnected
This event is triggered when manage client connected to server or disconnect to server

Format: `void ServerConnected(bool bConnected)`
`bConnected: connected or not`

Return: none

Channel related methods and events:

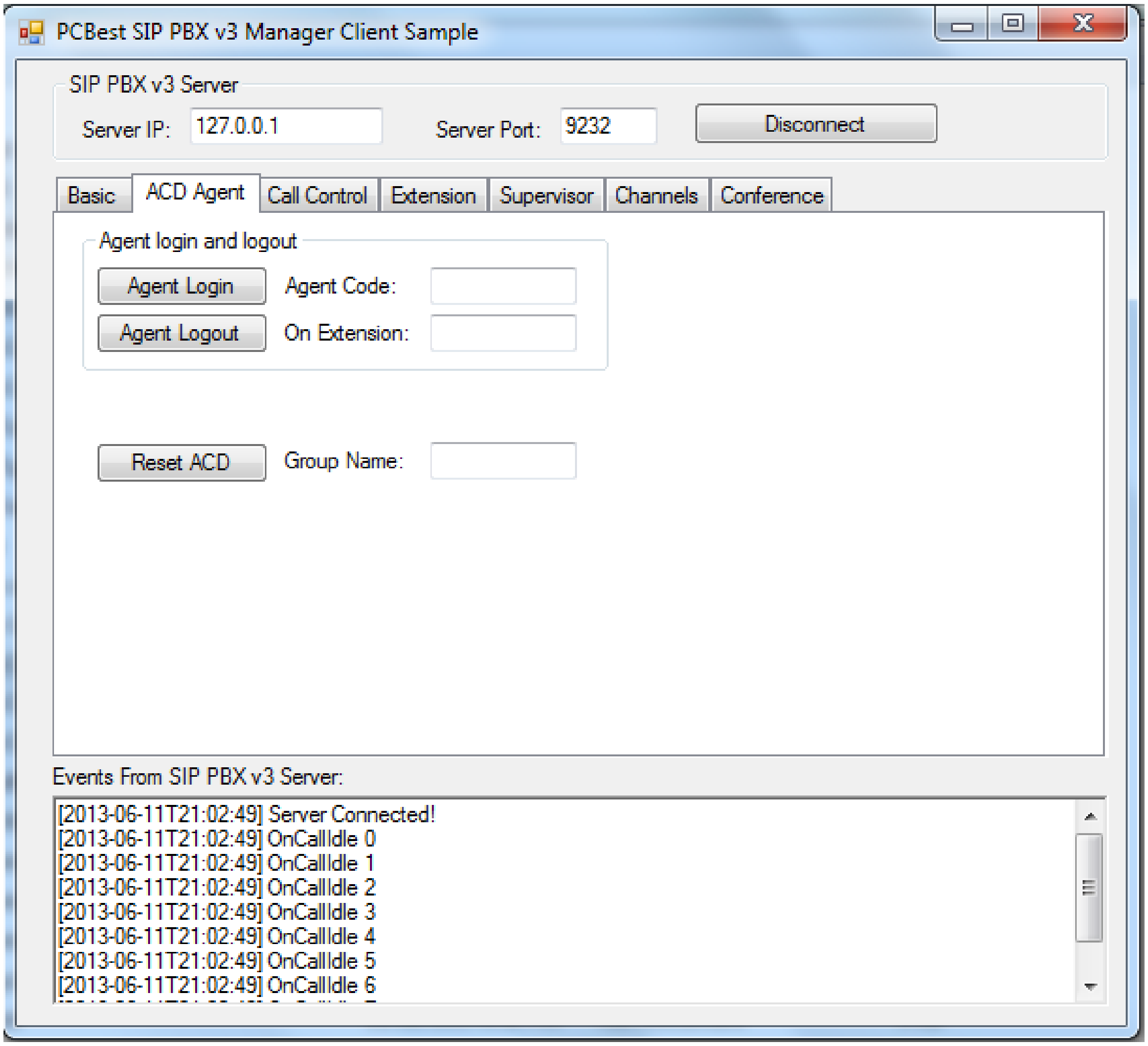


ResetChannel
Reset a channel. Disconnect a call on the channel if there is any.

Format: `void ResetChannel(int ch)`
`ch: the index of channel, based on 0.`

Return: none, but it will trigger OnCallIdle event if there was a call on this channel

Agent related methods:



AgentLogin

Specify an agent login on an extension. It will trigger the event `OnAgentLog`.

Format:
```
public void AgentLogin(string agentCode, string externNum, bool
bLogin, string p1, string p2, string p3)
agentCode: the code of agent
extenNum: the extension number
bLogin: true=login false=logout
p1,p2,p3: personal data for saving in database
```

Return: none, but it will trigger the event following.

OnAgentLog

The event when an agent login or logout.

Format:
```
public void OnAgentLog(bool bLogIn, string agentCode, string
extenNum, string p1, string p2, string p3)
bLogIn: true=login false=logout
agentCode: the code of agent
```

```
extenNum: the extension number
p1,p2,p3: personal data for saving in database
```

Return: none.

GetAgentStatus
Get the agent calling status. It will trigger the event `OnAgentStatus`.

Format: `void GetAgentStatus(string agentCode)`
```
agentCode: the code of agent
```

Return: none, it will trigger the event following.

OnAgentStatus
The event for agent status.

Format: `void OnAgentStatus(string agentCode, string atExten, string callStatus)`
```
agentCode: the code of agent
atExten: The extension number which agent is at(logged in).
callStatus: 0 = idle, 10 = offered, 20 = dialing, 21 = ringing, 30 = connected
```

Return: none

ResetACD
Reset a ACD group

Format: `void ResetACD(string acdName)`
```
acdName: the name of hunt group(ACD group).
```

Return: none

Call Control Related Methods and Events:

MakeCall
make a call out

Format:
```
string MakeCall(int ch, string caller, string callee)
acdName: the name of hunt group(ACD group).
ch: the index of channel
caller: the caller in sip address format: <sip:1234@abc.com>.
callee: the called id in sip format: <sip:456@def.com:5060>.
```

Return: the command id for later on to get the result

MakeCall
make a call out

Format:
```
string MakeCall(int ch, string caller, string callee, string
username, string passwd)
ch: the index of channel
caller: the caller in sip address format: <sip:1234@abc.com>.
callee: the called id in sip format: <sip:456@def.com:5060>.
username: the user name for outbound call credential
passwd: the password for outbound call credential
```

Return: the command id for later on to get the result

MakeCall
make a call out

Format:
```
string MakeCall(int ch, string caller, string callee, string
username, string passwd, string uri, string contact)
ch: the index of channel
```

```
caller: the caller in sip address format: <sip:1234@abc.com>.
callee: the called id in sip format: <sip:456@def.com:5060>.
username: the user name for outbound call credential
passwd: the password for outbound call credential
uri: the request URI in SIP invite
contact: the contact address in SIP invite
```

Return: the command id for later on to get the result

AnswerCall
answer an incoming call on a channel

Format: `void AnswerCall(int ch)`
```
ch: the index of channel
```

Return: none, but it will trigger the event OnCallConnected if succeed.

HangupCall
disconnect call on a channel

Format: `void HangupCall(int ch)`
```
ch: the index of channel
```

Return: none, but it will trigger the event OnCallIdle if succeed.

HangupCall
disconnect call on a channel

Format: `void HangupCall(int ch, int reasonCode, string reasonDesc)`
```
ch: the index of channel
reasonCode: reason code
reasonDesc: reason description
```

Return: none, but it will trigger the event OnCallIdle if succeed.

HoldCall
hold call on a channel

Format: `void HoldCall(int ch)`
```
ch: the index of channel
```

Return: none, but it will trigger the event OnCallHold if succeed.

TransferCall
blind transfer call on a channel

Format: `void TransferCall(int ch, string callee) //blind transfer`
```
ch: the index of channel
callee: transferee sip address, like <sip:78646@pcbest.net>
```

Return: none

TransferCall
consult transfer call on a channel

Format:
```
void TransferCall(int ch, string callee, int ch1) //consult
transfer
ch: the index of channel
callee: transferee sip address, like <sip:78646@pcbest.net>
ch1: the index of another channel which is the address above but
connected
```

Return: none

OnCallConnected
This event is triggered whenever there is a call connected

Format:
```
void OnCallConnected(int ch, string unique_id, string
dialplan_name, string audio_fn)
ch: the index of channel
unique_id: unique id to mark this call
dialplan_name: dialplan name will be used for this call
audio_fn: if recording, its file name.
```

Return: none

OnCallIdle
This event is triggered whenever a call got disconnected

Format:
```
void OnCallIdle(int ch, string unique_id, string dialplan_name,
string audio_fn)
ch: the index of channel
unique_id: unique id to mark this call
dialplan_name: dialplan name will be used for this call
audio_fn: if recording, its file name.
```

Return: none

OnCallRinging
This event is triggered whenever a outbound call is ringing(remote is ringing).

Format:
```
void OnCallRinging(int ch, string unique_id, string
dialplan_name, string audio_fn)
ch: the index of channel
unique_id: unique id to mark this call
dialplan_name: dialplan name will be used for this call
audio_fn: if recording, its file name.
```

Return: none

OnCallDialing
This event is triggered whenever a outbound call is dialing.

Format:
```
void OnCallDialing(int ch, string unique_id, string caller,
string callee, string dialplan_name, string audio_fn)
ch: the index of channel
unique_id: unique id to mark this call
caller: caller id
callee: callee id
dialplan_name: dialplan name will be used for this call
audio_fn: if recording, its file name.
```

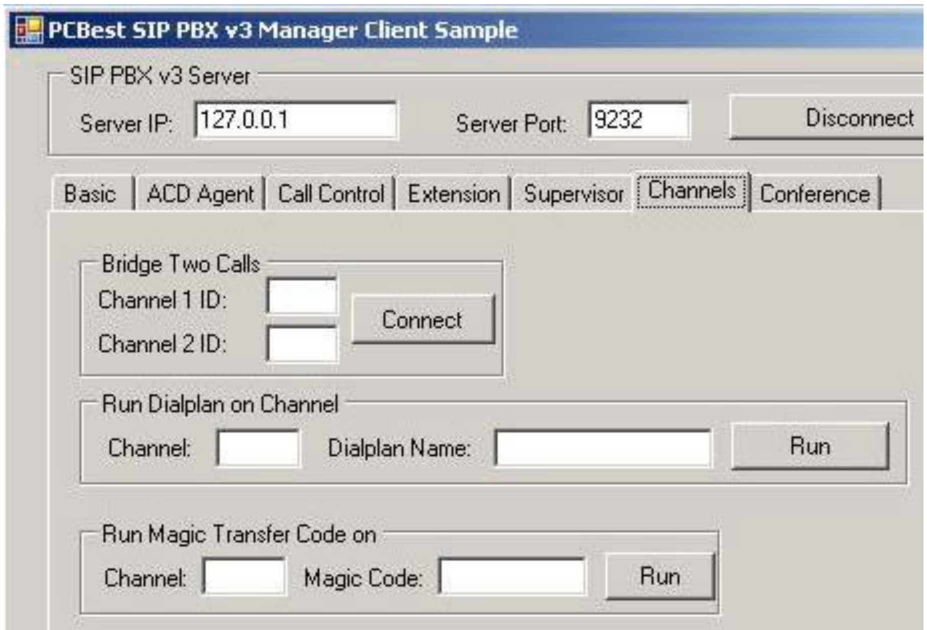
Return: none

OnCallOffered
This event is triggered whenever there is a new incoming call

Format:
```
void OnCallOffered(int ch, string unique_id, string caller,
string callee, string dialplan_name, string audio_fn)
ch: the index of channel
unique_id: unique id to mark this call
caller: caller id
callee: callee id
dialplan_name: dialplan name will be used for this call
audio_fn: if recording, its file name.
```

Return: none

Channel related methods and events:



MagicTransfer
magic transfer call

Format: `void MagicTransfer(int ch, string transCode)`
```
ch: the index of channel
transCode: The magic transfer code
```

Return: none

BridgeTwoCalls
brige the calls on two channels

Format: `string BridgeTwoCalls(int ch1, int ch2)`
```
ch1: the index of channel 1
ch2: the index of channel 2
```

Return: the command id for later to get the command status

RunDialPlan
run a dialplan on the channel

Format: `void RunDialPlan(int ch, string planName)`
```
ch: the index of channel
planName: the name of dialplan
```

Return: none

Extension related methods and events:

MakeExtensionCall
make a call to extension on specific channel

Format: `string MakeExtensionCall(int ch, string extnNum, string sCaller)`
```
ch: the index of channel
extnNum: the extension number
sCaller: caller id
```

Return: the command id

ExtenCallOut
Initiate a call from an extension to outside. It actually uses auto-dialer task to dial out then connect with extension once the call is connected.

Format: `void ExtenCallOut(string extnNum, string destNum, string sipAcctUserName, int ringTimeoutSec)`
```
extnNum: the extension number
destNum: the destination number
sipAcctUserName: the sip account name to be used for outbound call
ringTimeoutSec: how many seconds to wait in the ring
```

Return: none

ExtenCallOutEx
Initiate a call from an extension to outside. It actually uses auto-dialer task to dial out then connect with extension once the call is connected.

Format:
```
void ExtenCallOutEx(string extnNum, string destNum, string
sipAcctUserName, int ringTimeoutSec, bool enableDetect, bool
disconectAfterDetect)
extnNum: the extension number
destNum: the destination number
sipAcctUserName: the sip account name to be used for outbound call
ringTimeoutSec: how many seconds to wait in the ring
enableDetect: if enable human/answering machine detection
disconectAfterDetect: if disconnect call after detection is done.
```

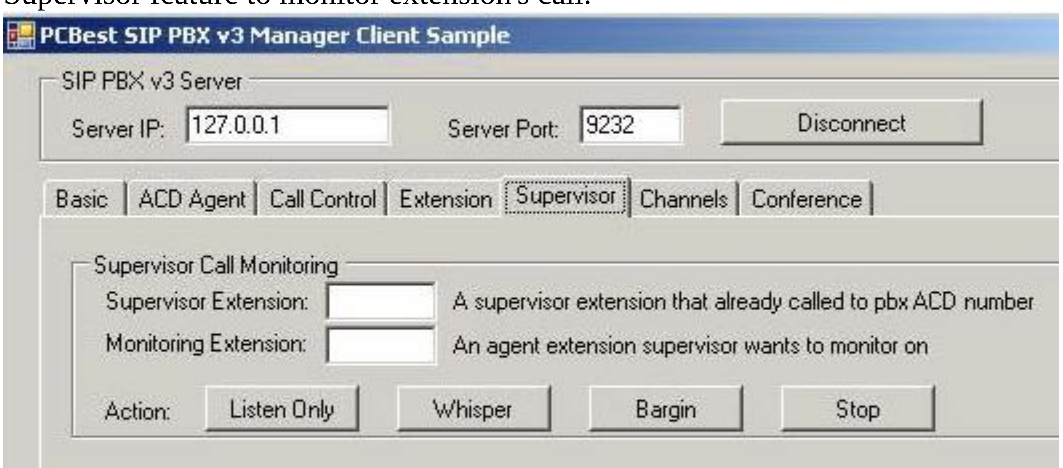
Return: none

OnExtenStatus
The event to reflect extension status

Format:
```
void OnExtenStatus(string extenNum, string callStatus)
extenNum: the extension number
callStatus: 0 = idle, 10 = offered, 20 = dialing, 21 = ringing, 30 =
connected
```

Return: none

Supervisor feature to monitor extension's call.



MonitorCall
Connect supervisor's extension with agent/user extension to allow supervisor monitor the current calls.

Format:
```
void MonitorCall(string extnSupervisor, string extnNormal, int
monitorType)
extnSupervisor: the supervisor extension
extnNormal: the extension number to be monitored.
monitorType: 0 = listen, 1 = whisper, 2 = talking(bargin), -1 = stop
monitoring(get out, withdraw)
```

Return: none

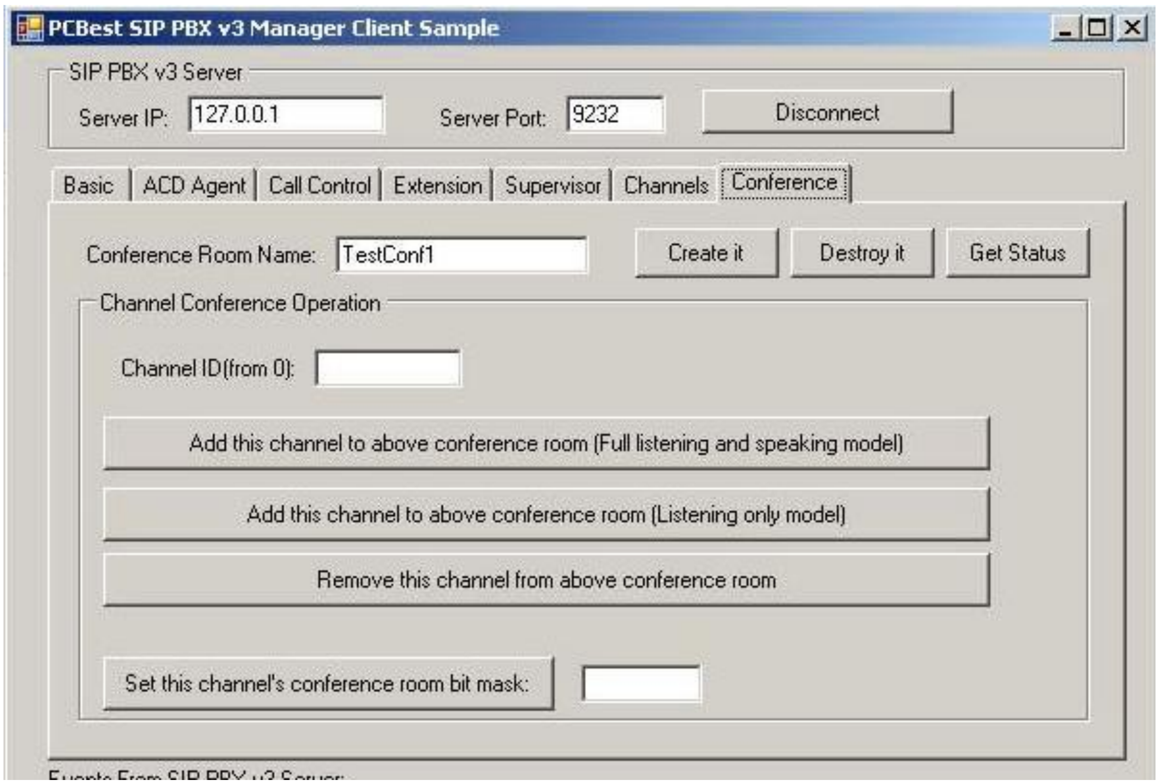
OnCallMonitoring
Monitoring call event

Format: `void OnCallMonitoring(string extenSupervisor, string extenNormal, int monitorType)`

```
extnSupervisor: the supervisor extension
extnNormal: the extension number to be monitored.
monitorType: 0 = listen, 1 = whisper, 2 = talking(bargin), -1 = stop
monitoring(get out, withdraw)
```

Return: none

Conference related methods and events



CreateConferenceRoom
Create a conference room on PBX dynamically.

Format: `void CreateConferenceRoom(string conf_name)`

```
conf_name: the name of conference room
```

Return: none

DestroyConferenceRoom

Destroy a conference room on PBX dynamically.

Format: `void DestroyConferenceRoom(string conf_name)`
`conf_name: the name of conference room`

Return: none

SetChanInConferenceRoom
This function is majorly used to send a channel into a conference room, or withdraw it.

Format: `void SetChanInConferenceRoom(int ch, string conf_name, int opt)`
`ch: the index of the channel`
`conf_name: the name of conference room`
`opt: 0 = take out of conference room. 1 = add into conference room. 2 = monitor(listening only, not speaking)`

Return: none

SetChanConferenceBitMask
Set channel's bitmask in conference room. Set channel's output when in conference room.
This function is used to disable the chan's output voice to other channels in the same conference.
Default channel mask is always 0xFFFFFFFF, which means output to all other channels in the conference room.
Every bit marks a channel. If the bit is 1, its voice can output to the channel.
The First channel in the conference room is 0x01.
The second channel in the conference room is 0x02.
The third channel in the conference room is 0x04.
So if you want the channel's output goes to the first channel, and the third channel, you can set this for this channel:
SetChanConfMaskch, 0x05); //which 0x05 = 0x01 + 0x04

Another example,
1st channel is connected with Agent. (Channel Index is 0, and it is the first channel set to the conference room)
2nd channel is connected with Customer. (Channel Index is 1, and it is the second channel set to the conference room)
3rd channel is supervisor. (Channel Index is 2, and it is the third channel set to the conference room)
They are all in the same conference room. Regularly if don't set anything, they can hear each other.
If supervisor only wants the agent hear his voice, not the customer, you can do so:
SetChanConfMask(2, 0x01);
It means that only the first channel get his voice.

Format: `void SetChanConferenceBitMask(int ch, uint bitMask)`
`ch: the index of the channel`
`bitMask:` bit mask to enable or disable output

Return: none

GetConferenceRoomStatus
Trigger the conference room event to get the status

Format: `void GetConferenceRoomStatus(string conf_name)`
`conf_name: the name of conference room`

Return: none, but the event OnConferenceRoomStatus will triggered.

OnConferenceRoomStatus
The event to receive current conference status

Format: `void OnConferenceRoomStatus(string roomName, string channels)`
`roomName: the name of conference room`
`channels: channel status in the conference room. the format is:`
channel,status;channel,status;channel,status
status: 1 = listen and speak, 2 = listening only(monitoring)

Return: none.

Please refer to the source code of manager client about full demonstration. The demo source cod is in C#, and if you are .NET developer, you can easily use it in your project. It provides very simple interfaces to use. But if you are like vb6, Delphi developer, and you want develop manager client application in your own language, here is guide how to do:

Assume you can use vb6 to open a TCP connection to IPPBXv3's manager port(you can set this in ippbxv3's GUI, default it is 9232). After connected, you will receive events like this:

command parameter1|parameter2|parameter3.....

For new incoming call, you will receive command CallOffered. Format like this:
CallOffered channel-id|unique-id|caller|callee|dialplan|recording-audio-filename

For call dialing out, you will receive command like this:
CallDialing channel-id|unique-id|caller|callee|dialplan|recording-audio-filename

If remote ringed for outbound call, you will receive:
CallRinging channel-id|unique-id|dialplan|recording-audio-filename

If call got connected, the event looks like:
CallConnected channel-id|unique-id|dialplan|recording-audio-filename

If call got disconnected, the command format is:
CallIdle channel-id|unique-id|dialplan|recording-audio-filename

There are other commands, and if you need, please contact PCBest Networks support for more details.

7.3 Database Development (V3)

PBX v3 is a completely database driven engine. It saves everything into database table. For example, real-time status of PBX are saved into status_xxx.

Tables:

cdr_xxx are CDR tables.

auto_dialer_xxx are auto dialer tables.

cfg_xxx are PBX configuration tables.
If you want to develop your own user interfaces, like web interface, to work with PBX, cfg_xxx tables are the tables you mostly need to deal with. Each cfg table has a field **ModTag**, which makes this record's status.
If you add or change a record, you need to set ModTag to 1. PBX service will later refresh its memory and set this tag back to 0.
If you want to remove(delete) the record, you need to set ModTag to 2. PBX service will later delete it from table.
When ModTag is 0, then it means there is no change on this record.

log_xxx are PBX real-time log table.

opt_cmd are PBX command table. PBX checks this table regularly to see if there are commands sent to PBX through DB.

status_xxx are PBX real-time status table.

voice_mailbox is voice mailbox table.

Here is the full list of database table of PBX v3:

- Tables
  - System Tables
  - dbo.auto_dialer_done
  - dbo.auto_dialer_jobs
  - dbo.cdr_acd
  - dbo.cdr_agent
  - dbo.cdr_exten
  - dbo.cdr_pbx
  - dbo.cfg_agents
  - dbo.cfg_autodialertasks
  - dbo.cfg_conferencerooms
  - dbo.cfg_dests
  - dbo.cfg_dialplans
  - dbo.cfg_extensions
  - dbo.cfg_huntgroups
  - dbo.cfg_ivrs
  - dbo.cfg_ivrsubitems
  - dbo.cfg_monitorgroups
  - dbo.cfg_parkingslots
  - dbo.cfg_pickupgroups
  - dbo.cfg_ringgroups
  - dbo.cfg_sipaccounts
  - dbo.cfg_sys
  - dbo.log_sys
  - dbo.opt_cmd
  - dbo.status_acd
  - dbo.status_agent
  - dbo.status_channel
  - dbo.status_conferenceroom
  - dbo.status_exten
  - dbo.status_parkingslot
  - dbo.status_sipaccount
  - dbo.voice_mailbox

For more detail info about database development of PBX v3, please contact PCBest at support@pcbest.net