

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

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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: <u>support@pcbest.net</u> Toll Free(USA & Canada): 1-888-733-6620 Local and International: 1-613-800-2202

2 Installing PBX

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

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



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

4. Click next.

← Setup - PCBest SIP PBX v3 version 3.70 —	×
Select Start Menu Folder Where should Setup place the program's shortcuts?	Ĵ
Setup will create the program's shortcuts in the following Start Menu folder.	
To continue, click Next. If you would like to select a different folder, click Browse.	
PCBest SIP PBX v3 Browse	
Back Next Cance	

5. Then confirm the installation.

		×
		(10)
any set	tings.	
	1	^ -
	>	
	any set	any settings.

It is done.

Setup - PCBest SIP PBX v3 version	3.70 — 🗆 ×
F	Completing the PCBest SIP PBX v3 Setup Wizard Setup has finished installing PCBest SIP PBX v3 on your computer. The application may be launched by selecting the installed shortcuts. Click Finish to exit Setup.
	Finish

Open Windows services:

Services							×		
File Action View Help									
🆏 Services (Local)	Services (Local)								
	SIP PBX v3	Name 🛆	Description	Status	Startup Type	Log On As	^		
	<u>Start</u> the service	Remote Packet Cap Remote Procedure Remote Procedure Remote Registry Removable Storage Routing and Remot Secondary Logon Security Accounts Security Center Security Center	Allows to c Provides th Manages t Enables re Offers rout Enables st Stores sec Monitors s Supports fil	Started Started Started Started Started Started	Manual Automatic Manual Automatic Manual Disabled Automatic Automatic Automatic Automatic	Local System Network S Network S Local Service Local System Local System Local System Local System Local System			
	Extended / Standard /	Shell HardWare Det SIP PBX v3 Smart Card SQL Server (SQLEX SQL Server Active SQL Server Browser SQL Server V5S Wri	Manages a Provides at Enables int Provides 5 Provides th	Started Started Started Started	Automatic Manual Automatic Disabled Automatic Automatic	Local System Local System Local Service Network S Network S Local System	~		

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":

<u> </u>		Open
	0	Run as administrator
a	_	Troubleshoot compatibility
gtsimplephone		
ዿ jre-7u11-windows-i58	К.	Scan for viruses
🙆 Microsoft Visual Studi		Add to archive
SQLEXPR_x86_ENU		Add to "SQLEXPRWT_x86_ENU-2012.rar"
SQLEXPR_x86_ENU-sp		Compress and email
SQLEXPRWT_x86_ENU	۱	Compress to "SQLEXPRWT_x86_ENU-2012.rar" and email
SQLEXPRWT_x86_ENU		Extract files
SOI ManagementStud		Extract Here
		Extract to SOLEXPRWT x86 ENU-2012\

Choose new SQL server stand-alone installation:

1 SQL Server Installation Center	- • ×
Planning Installation	New SQL Server stand-alone installation or add features to an existing installation Launch a wizard to install SQL Server 2012 in a non-clustered environment or to add features to an existing SQL Server 2012 instance.
Tools Tesources	Upgrade from SQL Server 2005, SQL Server 2008 or SQL Server 2008 R2 Launch a wizard to upgrade SQL Server 2005, SQL Server 2008 or SQL Server 2008 R2 to SQL Server 2012.
Options	
SQL Server 2012	

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

1 SQL Server 2012 Setup	anting 10. Second 20.1 second								
Setup Support Rules									
Setup Support Rules identify pr corrected before Setup can con	Setup Support Rules identify problems that might occur when you install SQL Server Setup support files. Failures must be corrected before Setup can continue.								
Setup Support Rules	Operation completed. Passed: 7. Failed 0. Warning 0. Skipped 0.								
License Terms									
Feature Selection									
Installation Rules	Show details >>	Re-run							
Instance Configuration	View detailed report								
Disk Space Requirements									
Server Configuration									
Database Engine Configuration									
Error Reporting									
Installation Configuration Rules									
Installation Progress									
Complete									
	< Back Next > Cancel	Help							

Accept license terms, and clieck Next:



📸 SQL Server 2012 Setup								
Feature Selection Select the Express features to install.								
Setup Support Rules	Features:	Feature description:						
License Terms Feature Selection Installation Rules Instance Configuration Disk Space Requirements Server Configuration Database Engine Configuration Error Reporting Installation Configuration Rules	Instance Features	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.						
Installation Progress Complete		Aiready 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	 Default instance Named instance: 	SQLEXPRESS					
Installation Rules Instance Configuration Disk Space Requirements Server Configuration Disk Disk Configuration	Instance ID: Instance root directory:	SQLEXPRESS C:\Program Files\	Microsoft SQL Serve	:r\			
Database Engine Configuration Error Reporting Installation Configuration Rules Installation Progress	SQL Server directory: Installed instances:	C:\Program Files\N	Aicrosoft SQL Serve	r\MSSQL11.SQLEXPR	ESS		
Complete	Instance Name	Instance ID	Features	Edition	Version		
			< Back	Next > Can	cel Help		

SQL Server 2012 Setup	19992 T	an Table			
Specify the service accounts and	d collation configuration.				
Setup Support Rules	Service Accounts Collation				
Feature Selection	Microsoft recommends that you u	use a separate account for each	SQL Server servi	ce.	
Installation Rules	Service	Account Name	Password	Startup Type	e
nstance Configuration	SQL Server Database Engine	NT Service\MSSQL\$SQL		Automatic	•
Disk Space Requirements	SQL Server Browser	NT AUTHORITY\LOCAL		Disabled	-
latabase Engine Configuration rror Reporting nstallation Configuration Rules nstallation Progress Complete					
1		< Back Next	> Can	cel He	elp

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

📸 SQL Server 2012 Setup								
Database Engine Configuration								
Specify Database Engine authentication security mode, administrators and data directories.								
Setup Support Rules	Server Configuration Data Directories User Instances FILESTREAM							
License Terms Feature Selection	Specify the authentication mode and administrators for the Database Eng	ine.						
Installation Rules	Authentication Mode							
Instance Configuration	Windows authentication mode							
Disk Space Requirements	 Mixed Mode (SOL Server authentication and Windows authentication) 							
Server Configuration								
Database Engine Configuration	specify the password for the SQL Server system administrator (sa) account							
Installation Configuration Rules	Enter password:							
Installation Progress	Confirm password:							
Complete	Specify SQL Server administrators							
	Yonge-PCW7\Yonge (Yonge)	SOL Server administrators						
		have unrestricted access						
		to the Database Engine.						
	Add Current User Add Bemove							
	And carrent over Andam Inchlore							
< Back Next > Cancel Help								

🚼 SQL Server 2012 Setup	
Error Reporting	
Help Microsoft improve SQL Se	erver features and services.
Setup Support Rules License Terms Feature Selection Installation Rules	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.
Disk Space Requirements Server Configuration Database Engine Configuration Error Reporting	See the Microsoft SQL Server 2012 Privacy Statement for more information. Read more about Microsoft Update and Automatic Update.
Installation Configuration Rules Installation Progress Complete	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:

📸 SQL Server 2012 Setup	anting \$3. Second \$5. onterco.		x			
Complete Your SQL Server 2012 installation	on completed successfully with product up	ndates.				
Setup Support Rules	Information about the Setup operation of	or possible next steps:				
License Terms	Feature	Status				
Feature Selection	Management Tools - Basic	Succeeded				
Installation Rules	Database Engine Services	Succeeded				
Instance Configuration	SQL Server Replication	Succeeded				
Disk Space Requirements	SQL Browser	Succeeded				
Server Configuration	SQL Writer	Succeeded				
Database Engine Configuration			- 1			
Error Reporting						
Installation Configuration Rules	Details:					
Installation Progress	Viewing Product Documentation	for SOL Server				
Complete	Viewing Product Documentation	IUI JUL JEIVEI	=			
	Only the components that you use to view and manage the documentation for SQL Serve been installed. By default, the Help Viewer component uses the online library. After instal SQL Server, you can use the Help Library Manager component to download documentation your local computer. For more information, see <u>Use Microsoft Books Online for SQL Server ">http://go.microsoft.com/fwlink/?LinklD=224683>. Summary log file has been saved to the following location: C:\Program Files\Microsoft SQL Server\110\Setup Bootstrap\Log\20130422 172129\Summary Y </u>					
	PCW7 20130422 172129.txt	Close Help				

Recycle Bin Adobe		Open
Reader XI		Run as administrator
	-	Troubleshoot compatibility
		Troubleshoot compatibility
	\mathbf{K}_{1}	Scan for viruses
Google		Open file location
Chrome		Add to archive
		Add to "Ssms.rar"
		Compress and email
	8	Compress to "Ssms.rar" and email
		Pin to Taskbar
Kasnershy		Pin to Start Menu
Kaspersky Internet Security 2013		
Maintenance		Restore previous versions
Microsoft Silverlight		Send to 🔸
Microsoft Silverlight 3 SDK		Cut.
Microsoft Silverlight 4 SDK		cut
Microsoft SOL Server 2008		Сору
Microsoft SQL Server 2012		Delete
Download Microsoft SQL Server Co	0	Rename
Import and Export Data (32-bit)		Descrition
🧏 SQL Server Management Studio	-	Properties
퉬 Configuration Tools		Computer
🍶 Integration Services		Compare

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.

📑 Connect to Server	X				
SQL Server 2012					
Server type:	Database Engine 👻				
Server name:	YONGE-PCW7\SQLEXPRESS -				
Authentication:	Windows Authentication				
User name:	Yonge-PCW7\Yonge 👻				
Password:					
	Remember password				
Connect	Cancel Help Options >>				

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



New Database						x
Select a page General	🔄 Script 🔻 📑	Help				
Poptions Filegroups	Database name:		sippbxv3			
	Owner:		<default></default>			
	√ Use full-text in	ndexing				
	Database files:					
	Logical Name	File Type	Filegroup	Initial Size (MB)	Autogrowth / Maxsize	
	sippbxv3	Rows	PRIMARY	3	By 1 MB, Unlimited	
	sippbxv3_log	Log	Not Applicable	1	By 10 percent, Unlimited	
Connection						
Server: YONGE-PCW7\SQLEXPRESS						
Connection: Yonge-PCW7\Yonge						
View connection properties						
Progress						
Ready	٠	III		Add	Remove	4
	·			(OK Cance	

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.

sippbxv3 - Notepad	
File Edit Format View Help	
Please set the <database> section for DB connection.</database>	
Registry: From Windows Registry. (Not implemented)	Authentication Mode
XML: From this XML file.(Not implemented) >	 Windows authentication mode Mixed Mode (SQL Server authentication and Windows authentication)
<cfgsource>Database</cfgsource>	
<database> <enabled>true</enabled></database>	Specify the password for the SQL Server system administrator (sa) account.
<pre><!-- DBT/De: 0 = MS_S0L_2005_Express_1 = MS_S0L_2005_2 = MS_S0L_2005_2</pre--></pre>	Enter password:
> <pre>construction construction construct</pre>	Confirm password:
<pre><!-- DBServer: blank = local, otherwise give the server name or--> cppServer: </pre>	Specify SQL Server administrators
<pre><dbbel ver=""></dbbel> </pre>	
<br 0 = SQL Authentication, 1 = Windows Authentication	
<authtype>0</authtype>	
<br if AuthType is SQL Authentication, then please give the Use for connecting SQL server. > <username>sa</username> <password>xxxxxx</password> 	erName and Password

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):

Services							×
File Action View	Help						
← → 💽 🗗 🖸) 🗈 😫 🖬 🕨 = II =>						
🍇 Services (Local)	Services (Local)						
	SIP PBX v3	Name 🗡	Description	Status	Startup Type	Log On As	^
	<u>Start</u> the service	Remote Packet Cap Remote Procedure Remote Procedure Remote Registry Removable Storage Routing and Remot Secondary Logon Security Accounts Security Accounts	Allows to C Provides th Manages t Enables re Offers rout Enables st Stores sec Monitors s Supports fil	Started Started Started Started Started Started Started	Manual Automatic Manual Automatic Manual Disabled Automatic Automatic Automatic	Local System Network S Network S Local Service Local System Local System Local System Local System Local System	
		Shell Hardware Det	Provides n	Started	Automatic	Local System	=
		SIP PBX v3			Manual	Local System	-
		Smart Card	Manages a Drouidos st	Charland	Manual	Local Service	
		SOL Server (SQLEX	Frables int	Started	Disabled	Network S	
		SQL Server Browser	Provides S	Started	Automatic	Network S	
		SQL Server VSS Wri	Provides th	Started	Automatic	Local System	~
	Extended Standard						_

If you get popup, and cannot start the service:

Services (Local)								
SIP PBX v3	Name		Description	Status				
	🔍 Secure Socket Tunneling P	rotocol Service	Provides support for the Secure Socket Tunneling.					
Start the service	Security Accounts Manage	er	The startup of this service signals other services t	Started				
	Security Center		The WSCSVC (Windows Security Center) service S					
	Server		Supports file, print, and named-pipe sharing over	. Started				
	🌼 Shell Hardware Detection		Provides notifications for AutoPlay hardware eve	Started				
	🕵 SIP PBX v3							
	🔍 Skype Updater		Enables the detection, download and installation					
	🎑 Smart Card		Manages access to smart cards read by this com					
	🔍 Smart Card Removal Polici	v	Allows the system to be configured to lock the us					
	SNMP Trap Services		Resident from the second proceeded by faculty or as-	×				
	🔍 Software Protec							
	SPP Notification		an ian an I anal Camputer started and then started					
	🔍 SQL Server (SQL 🛛 👔	Some services st	op automatically if they are not in use by other service	es				
	🔍 SQL Server Acti	or programs.	-,,,,,					
	🔍 SQL Server Brov							
	SQL Server VSS							
	SSDP Discovery		0	к				
	Superfetch							
	😪 Symantec pcAnywhere Ho	ist Service	Allows Remote pcAnywhere users to connect to t					

Please check: a. Event Viewer:

Event Viewer		1		
File Action View Help				
🗢 🔿 🗾 🖬				
🛃 Event Viewer (Local)	Administrative Events Numbe	r of events: 4,925		
Custom Views	Number of suggests 4.025			
Maninistrative Events	V Number of events: 4,925			
Windows Logs	Level	Date and Time	Source	Event ID 🔺
Applications and Services Lo	Error	23/04/2013 9:54:37 PM	SIP PBX v3	0
Cisco	Error	23/04/2013 9:42:06 PM	SIP PBX v3	0
Thermot Events	🕕 Error	23/04/2013 9:41:03 PM	SIP PBX v3	0 _
Key Management Service	1			•
Media Center	Event 0. SIP PBX v3			X
Microsoft				
📔 Microsoft Office Diagnos	General Details			
Microsoft Office Sessions	Friendly View XMI	View		
Windows PowerShell		VIEW		
Subscriptions				<u>~</u>
	+ System			
	- EventData			=
	Eventoutu			
		Service cannot be started	d. System.Exception: LoadConfic	() cannot
		access DB! at SIPPBXv3.5	SIPPBXMain.LoadConfig() in	
		C:\temp\projects\GTSIPF	PBXv3/SIPPBXv3/SIPPBXv3/SIPP	'BXMain.cs:line
		1115 at SIPPBXv3.SIPPB	XMain.StartServer() in	-

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

	🍌 audio	*	Name			
	🍌 log			abt		
	퉬 moh		PDAV5-IC	Jg.txt		
	퉬 plugin					
	퉬 record					
	鷆 report					
	퉬 sdk					
	鷆 temp					
	鷆 vmb					
	鷆 web					
	PBXv3-log.txt - Notepad					
	File Edit Format View Help)				
	23/04/2013 9:37:25 PM: 23/04/2013 9:40:38 PM: 23/04/2013 9:41:03 PM: 23/04/2013 9:42:06 PM: 23/04/2013 9:54:16 PM: 23/04/2013 9:54:37 PM: 23/04/2013 9:59:47 PM:	LOa LOa LOa LOa LOa LOa	adConfig() adConfig() adConfig() adConfig() adConfig() adConfig() adConfig()	cannot cannot cannot cannot cannot cannot	access access access access access access access access	DB!. DB!. DB!. DB!. DB!. DB!. DB!.
1			-			

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...".

💣 Windows Defender Firewall with	h Advanced Security						-	×
File Action View Help								
🗢 🄿 🙍 🖬 🗟 🖬								
Minuows Derender Firewall witl	Inbound Rules					Actions		
Inbound Rules	Name	Group	Profile	Enabled	Acti ^	Inbound Pulse		
Connection Security Rules	🥑 csharpsipphone.exe		Public	Yes	Allo 🔇	New Rule	\supset	
> 🔜 Monitoring	🔮 csharpsipphone.exe		Public	Yes	Allo	Filter by Profile		•
	🔮 Dropbox		All	Yes	Allo	Tilhan hu Chata		
	🔮 gtsimplephone.exe		Public	Yes	Allo	Y Filter by State		
1	🕑 gtsimplephone.exe		Public	Yes	Allo	Filter by Group		•
	🔮 Microsoft Lync		Public	Yes	Allo	View		•
	Microsoft Lync		Public	Yes	Allo	Defeet		
1	Ø Microsoft Lync UcMapi		Public	Yes	Allo	G Kerresh		
1	Ø Microsoft Lync UcMapi		Public	Yes	Allo	📑 Export List		
	Ø Microsoft Office Outlook		Public	Yes	Allo	Help		
	PCBest SIP PBX		All	Yes	Allo			
	🔮 VR2 WinApp		Public	Yes	Allo			
	VR2 WinApp		Public	Yes	Allo			
	@{Microsoft.AAD.BrokerPlugin_1000.190	@{Microsoft.AAD.BrokerPlu	Domai	Yes	Allo			

Choose "program", then "next".

🔗 New Inbound Rule V	Vizard	×
Rule Type		
Select the type of firewall ru	le to create.	
Steps:	What type of rule would you like to create?	
 Rule Type Program 		
ActionProfile	 Program Rule that controls connections for a program. 	
 Name 	 Port Rule that controls connections for a TCP or UDP port. 	
	Predefined: @FirewallAP1.dll -80200	
	Rule that controls connections for a Windows experience.	
	Custom rule.	
	< Back Next > Cancel	

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

💣 New Inbound Rule Wizard		×
Action		
Specify the action to be taken wh	en a connection matches the conditions specified in the rule.	
Steps:		
Rule Type	What action should be taken when a connection matches the specified conditions?	
Program	Allow the connection	
Action	This includes connections that are protected with IPsec as well as those are not.	
Profile	Allow the connection if it is easy m	
Name	 Allow the connection if it is secure This includes only connections that have been authenticated by using IPsec. Connections will be secured using the settings in IPsec properties and rules in the Connection Security Rule node. Customize Block the connection 	

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:



Display Name	User Name	Domain	Proxy

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 in	coming calls
	🗸 ок	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.



Extension	Name	Email	

Add an	extension			
Basic	Advanced	Voice Mail Box	Call Forwarding	
Exte	ension:	101	(Sample This is a	: 101, 1001. Must be unique to the whole PBX, lso the user name for SIP extension)
Use	er Name:	Mike		(Sample: Bob wall, Mike Smith)
Pas	sword:			(The password for SIP extension registration)
Ema	ail:	Mike@mycompar	ny.com	
Exte	ension Type:	Normal	-	
Virtu	ual Extension	Outbound Address	s or Number:	
(Us 123	e outbound di @sipprovider	alplan rule to set o .com)	utbound numbe	r, or use SIP address format like:
IP E	Extension Auth	nization Type:	Proxy	•
		🧹 Add Extensio	n	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.



Name	Sound File	D	TMF Accept					
rompt: C	:\ivmer	iu.wav			laan dala ann C		Browse	
--	---	---	--	---	---	--	--	-----------
ress 2 for rulaw or al	custome aw wav	e file, or 8k 16bit r	for billing departm nono PCM wav fi	nent. Stay on the extension nent. Stay on the	h, please dial it now. C he line for operator. So	ound file	e must be 8k 8b	it mono
/lillisecond	s to wait	for DTMF keys:	15000	Millisecond	s.			
Menu Opti	ons							
DTMF		Then play:(if bla	nk, jump to action	n directly)	Action:	[Destinations	
1	•			Browse	Dial extension	•	101	•
N/A	•			Browse	To another menu	•		Ŧ
N/A	•			Browse	To another menu	•		Ŧ
N/A	•			Browse	To another menu	•		Ŧ
N/A	•			Browse	To another menu	•		Ŧ
N/A	•			Browse	To another menu	•		Ŧ
N/A	•			Browse	To another menu	•		Ŧ
Accep	t Extens	sions. Note: if you	enable this optior	n, your custom	er may need to wait		Millise	conds
on the menu detem millise begini	e menu, l to accp nine if us conds to ng with	because of the fac et 1 to forward cal ers want to reach see if users have 6 or 7 for example	ct: if you have ext Is to ACD group. ACD group or wa more inputs. To (Leave 8 or 9 for	tensions begini When users st ant to reach an avoid delay us outbound rule)	ng with 1, like 101,10, ay on this meanu, and extemsion. So pbx w er experience, you car	2. Also input 1 ill wait a n set yo	you set above (), pbx needs to above amount o our extensions)TMF f

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

🤞 РСВ	est Networ	ks SIP PBX v	3.10 (Adm	nin Tool)
File	Options	Manage	Server	Operations
1 ≥ 1	SIP A	Accounts(Ex	ternal Line	es)
⊡ · PB)	Exter	nsions		Plan N
-	Hun	t Groups(AC	CD)	1
	Park	ing Slots		2
	Ring	Groups		ut 1
	IVR I	Menus		ut2
	Con	ference Roo	ms	.t3
	Dial	Plans		
	Out			

Plan Name	Call Direction	Called ID	Templete	

Add a dialplan Inbound1.

Plan Name:	Inbound1		Any name you like to give for this plan
Call Direction:	Inbound Outbound		Which call direction the plan is for
Caller Number:		•	Blank if no limit on caller
Called Number:	4321	•	Use * for any number, and ? for any one digit.
Plan Templete:	Auto Attendant(IVR Menu)	•	IVR1 -
Pre-strip:			Outbound called number pre-strip text For example: prestrip text for called number 9* is 9.
Pre-append:			Pre-append string after pre-strip.
Use SIP Accour	nt:	-	Which SIP account you want to use for oubound call

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.

🔫 РСВ	est Networl	cs SIP PBX v	/3.10 (Adn	nin Tool)					
File	Options	Manage	Server	Operati	ons				
1 ≥ 1	SIP A	ccounts(Ex	ternal Line	es)					
	Exter	nsions							
	Hunt	t Groups(AC	CD)						
	Parki	ing Slots							
	Ring	Groups							
	IVR N	/lenus							
	Conf	erence Roo	ms						
	Dial	Plans							
	_								
Autor Incon for ca	natical Call D ning calls will all center app	ps istribution Hu be automatio lication.	unt Group is cally distribu	s a group of uted to exte	f extensi ensions b	ions that by order	can ar This f	nswer ca reature is	lls. excellent gents
Nam	ie	Тур	e		Agents				

Basic Age	Group ents Advance	-	
Name:	ACD1	Any n be un	name you like to give to this ACD hunt group. Must nique.
Type:	Linear 👻	With a the lo In a c "roun In mo which	a linear hunt group, calls are always delivered to west-numbered available line. sircular hunt group, the calls are distributed id-robin''. ist-idle hunting, calls are always delivered to hever line has been idle the longest.
Play	music when waiting. Mus	ic folder:	C:\temp\projects\GTSIPPBX Browse
	🖌 ок		Cancel

Then add one ACD huntgroup:

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

Reference Add Hunt Group
Basic Agents Advance
Please select agents that can accept calls in this ACD group.
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)
Agents: Available Extensions or Agents: 101 102 103 <<== ==>> 103
OK Cancel

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.

Basic True Cel	and de Estanciano en Aconte	
Time Sci	Tedule Extensions of Agents	
Plan Name:	Inbound2	Any name you like to give for this plan
Call Direction:	Inbound Outbound	Which call direction the plan is for
Caller Number:		 Blank if no limit on caller
Called Number:	4321	✓ Use * for any number, and ? for any one digit.
Plan Templete:	Automatic Call Distribution(ACD)	✓ ACD1
Pre-strip:		Outbound called number pre-strip text For example: prestrip text for called number 9* is 9.
Pre-append:		Pre-append string after pre-strip.
Use SIP Accourt	t:	 Which SIP account you want to use for oubound call
	🖌 Finish	Cancel

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.

Plan Name:	Out Plan 1	Any name you like to give for this plan
Call Direction:	Inbound Outbound	Which call direction the plan is for
Caller Number:		Blank if no limit on caller
Called Number:	9* -	Use *for any number, and ?for any one digit.
Plan Templete:	Auto Attendant(IVR Menu)	
Pre-strip:	9	Outbound called number pre-strip text For example: prestrip text for called number 9* is 9.
Pre-append:		Pre-append string after pre-strip.
Use SIP Accou	nt: 4321 4321 -	Which SIP account you want to use for oubound call

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 $9x(x ext{ is real phone number you want to reach outside})$, 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

File Options Manage	v3.11 (A Serve	dmin To er Op	ool) erations Aut	to Dialer Help			23	Webphone - Wind
E BX		Euton	Namo	Status	Contact	Idle Emm	Char	× Norton · ② · ③ ·
Extensions		101	Mike		192 169 101 42	2010-02-20 11-55-26		
- 101		102	Pab		102.100.101.43	2010-03-20 11:55:20		
- 103		102	BOD		192.100.101.43	2010-03-20 11:55:24	4	🔶 Favorites 🛛 🚖 🌈 Suggested Sites 🔻
		103	Allen	Offline		2010-03-20 11:30:43	5	
		104	Super	Offline		2010-03-20 11:30:43	3	e webphone
Hunt Groups(ACD) — Parking Slots — Ring Groups U/R Menus — Monitor Groups — Conference Rooms — Conference Rooms — Conference Rooms ① - Channels [2010-03-20 11:26:02] SIP PBX v [2010-03-20 11:32:18] Extension [2010-03-20 11:32:21 Extension [2010-03-20 11:32:21 Extension [2010-03-20 11:35:25] DoCallFon defined.	3 Servic Plugin D Plugin D 101 just 102 just warding F	e is NC emo) T register register orExter	102 SIP: "101"< New Call 1 2 4 5 7 8 • 0	sip:101@192.168.	101.43> Options Answei Hungup Hold		•	101 102 Ringing 9 1 9 2 8 1 2 3 8 1 2 3 8 4 5 6 2 7 8 9 2 * 0 # 2
GTSIPRegistrar GTSIPSDK audio(8K8BitMulaw) bin doc			Dynamically I Dynamically I Mute Spea	Recording	Switch Sound in a live of Show Camer	Device all M M PM		 Ø,

PC Best Networks SIP PBX Reference

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

Edit IVR Menu			- 8 4	
IVR Menu Name:	IVR1	Please give an ur	nique name, like "IVR1", '	"MainMenu".
Prompt: c:\ivme	nu.wav			Browse
Sample sound: We press 2 for custome mulaw or alaw way	elcome to ABC company. If y er support. press 3 for billing ve file, or 8k 16bit mono PCN	ou know the extension department. Stay on th 4 wav file.	, please dial it now. Other e line for operator. Sound	wise, press 1 for sales. file must be 8k 8bit mono
Milliseconds to wai	t for DTMF keys: 15000	Milliseconds	l.	
Menu Options				
DTMF	Then play:(if blank, jump t	to action directly)	Action	Destinations
1 👻		Browse	Dial extension 🔹	101 🗸
N/A 👻		Browse	To another menu 🔻	IVR1 -
N/A 👻		Browse	To another menu 👻	IVR1 -
N/A 👻		Browse	To another menu 🔹	IVR1 -
N/A 👻		Browse	To another menu 🔹	IVR1 -
N/A 👻		Browse	To another menu 👻	IVR1 -
N/A 👻		Browse	To another menu 👻	IVR1 -
Accept Exten	sions. Note: if you enable th	is option, your custome	r may need to wait	Milliseconds
on the menu, menu to accp determine if u milliseconds to begining with	because of the fact: if you h let 1 to forward calls to ACD sers want to reach ACD grou o see if users have more inpu 6 or 7 for example(Leave 8	nave extensions beginin group. When users sta up or want to reach an uts. To avoid delay use or 9 for outbound rule).	ng with 1, like 101,102. Al y on this meanu, and inpu externsion. So pbx will wa r experience, you can set	so you set above DTMF ut 1, pbx needs to ait above amount of ; your extensions
	🖌 ок		X Cancel	

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.

Basic	Advanced	Voice Mail Box	Call Forwarding	
Ext	ension:	104	(Sample: This is a	101, 1001. Must be unique to the whole PBX, so the user name for SIP extension)
Use	er Name:	Super		(Sample: Bob wall, Mike Smith)
Pa	ssword:			(The password for SIP extension registration)
Em	ail:			
Ext	ension Type:	Virtual)
Virt	ual Extension	Valooana Addre	ss or Number:	91234567
(Us 123	e outbound d 3@sipprovider	ialplan rule to set com)	outbound numbe	r, or use SIP address format like:
IP	Extension Aut	hrization Type:	Proxy	

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.

File	Options	Manage	Server	Operations
∃ ⊳	SIP	Accounts(Exte	ernal Line	es)
	3) Exte	ensions		- F
	Hur	nt Groups(ACI	D)	
	Parl	king Slots		
L	Ring	g Groups		
	IVR	Menus		
	Cor	ference Roon	ns	
	Dial	Plans		

Name	Ring Type	Destinations	

Dasic		
Name:	rg1 Please give any name to this ring group	
Type:	Ring all destinations at one time Ring destinations by order	
Destinatio	ins	
101 102 103	Add	
	Delet	e
Music whe	en waiting	_
Play m	usic when waiting. Music folder: Browse	
Voice Mail	I Box	
Enable	Settings	

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.

🤫 PCE	📢 PCBest Networks SIP PBX v3.12 (Admin					in
File	Opt	ions	Manage	Server	Operatio	ns
		SIP Accounts(External Lines)		1		
😑 PB		Exte	ensions			Г
۲		Hunt Groups(ACD)				
		Parking Slots				
		Ring Groups				
	IVR Menus					
	Conference Rooms					
	Dial Plans					
Monitor Groups						

•	Parking Slots			
	Name	Number		
	🔶 Add	📔 Edit	Delete	🖌 ок

.....

•	Edit Parking Slo	t			
	Parking slot is used to park a call, which can be picked up later by dialing the parking slot's number.				
	After an agent answersed a call, he/she can input the parking slot's number to park this call. Once the call is parked successfully, the agent's call will be automatically disconnected, and another agent can dial the parking slot's number to pick up that call.				
	Basic Advance				
	Parking Slot Name:	PK1	Any name. Sample: Slot 1		
	Number:	*61	Sample: *61, #10,		
	Music On Hold				
	Play music who	en call parked			
	Music files from:		Browse		
		🖌 ок	Cancel		

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 **.

🔫 PCBest Networks SIP I	BX v3.12 (Admin Tool)			
File Options Manage	Server Operations Auto Dialer			
; 🕨 🔟	Special Numbers			
E PBX	Networks			
Extensions	System Options			
101	Folders			
103	Proxy Sites			
- SIP Accounts	Load Balance			
- Dial Plans	Database Connection			
Basic Magic Transfer Code Magic transfer code is a extensions, agents or dia For example, if blind tran extension 101 can alway Or extension 101 can pr has called number set to (NOTE: in order to transl needs to put a # key at	le lode that extension can use it to trans lplans. sfer code is defined as *#, then in a ca s press *#102 to transfer the call to 10 sss *#1234567# to transfer the call to o 1234567. er the call to another dialplan, the exter he end of dialplan number)	fer call to other II, the 2. dialplan which nsion side		
Blind Transfer Code: *# Attended Transfer Code: **				
ок	Cancel]		

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:

🔜 Add SIP Account	
Basic DIDs	
Display Name: ToGW	Sample: Bob Wall, Mike Keeler
User Name: 100	Sample: 7184773245, 1001, or Mike
SIP Domain: 192.168.1.19	Sample: pobest.net, voip.com
SIP Proxy: 192.168.1.10	Sample: pobest.net, usually same as domain
Authorization: 100	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 inco uncheck this option because it is a r	Any as GW doesn't check your authentication fake account
🖌 ок	Cancel

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.

PC Best Networks SIP PBX Reference

🔜 Dian Plan		
Basic Time Sc	hedule Extensions or Agents	
Plan Name:	ToGWDialplan	Any name you like to give for this plan
Call Direction:	 Inbound Outbound 	Which call direction the plan is for
Caller Number:	~	Blank if no limit on caller
Called Number:	9× 🗸	Use * for any number, and ? for any one digit.
Plan Templete:	Auto Attendant(IVR Menu)	~
Pre-strip:	9	Outbound called number pre-strip text
Pre-append:		Pre-append string after pre-strip.
Use SIP Accou	nt: ToGW 100 💌	Which SIP account you want to use for oubound call
	OK	Cancel

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.



🔡 Conference Room		
Name		
Add	Edit Delete	🖌 ок

🖶 Edit Conference Room	
Conference Room Name:	
Conf1	
Must be unique. Sample: Conf 1, Tech Conf Room,	
🖌 OK 🛛 🎽 Canc	el

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.

	Add an extension	
	Basic Advanced	Voice Mail Box Call Forwarding
	Extension:	1002 (Sample: 101, 1001. Must be unique to the whole PBX, This is also the user name for SIP extension)
	User Name:	Bob (Sample: Bob wall, Mike Smith)
	Password:	(The password for SIP extension registration)
	Email:	
	Extension Type:	Virtual 👻
	Virtual Extension	Outbound Address or Number: *@OutPlan1
	(Use outbound d outbound dialpla *@sipprovider.cc *@outbound-dial	alplan rule to set outbound number, sample like 9123456, if you have defined n for 9*. Or use SIP address format like: 123@sipprovider.com, or m. * means forward the original called id. You can also use plan-name, which means forwarded original called id to an outbound dialplan)
	IP Extension Aut	nrization Type: Proxy -
-		Add Extension Xancel

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.

🖳 Dian Plan	
Basic Time Schedule Extensions or Agents	
Plan Name:	Any name you like to give for this plan
Call Direction: 💿 Inbound 🔘 Outbound	Which call direction the plan is for
Caller Number:	Blank if no limit on caller
Called Number: 9*1	Use * for any number, and ? for any one digit.
Plan Templete: Call Forward 💌	OutPlan1 👻
Pre-strip:	Outbound called number pre-strip text For example: prestrip text for called number 9* is 9.
Pre-append:	Pre-append string after pre-strip.
Use SIP Account:	Which SIP account you want to use for oubound call
Finish	Cancel

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.

🖳 Dian Plan		
Basic Time Sch	nedule Extensions or Agents	
Plan Name:		Any name you like to give for this plan
Call Direction:	Inbound Outbound	Which call direction the plan is for
Caller Number:		Blank if no limit on caller
Called Number:	8765 💌	Use * for any number, and ? for any one digit.
Plan Templete:	Music Server 👻	C:\temp\projects\GTSIPPBXv3\Ir 👻
Pre-strip:		Outbound called number pre-strip text For example: prestrip text for called number 9* is 9.
Pre-append:		Pre-append string after pre-strip.
Use SIP Account	t: 🔍 👻	Which SIP account you want to use for oubound call
	🖌 Finish	Cancel

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.

🖳 Dian Plan	
Basic Time Schedule Extensions or Agents	
Plan Name:	Any name you like to give for this plan
Call Direction: 💿 Inbound 💿 Outbound	Which call direction the plan is for
Caller Number:	Blank if no limit on caller
Called Number: 8765 🗸	Use * for any number, and ? for any one digit.
Plan Templete: Echo Test 🗸	C:\temp\projects\GTSIPPBXv3\lr 👻
Pre-strip:	Outbound called number pre-strip text For example: prestrip text for called number 9* is 9.
Pre-append:	Pre-append string after pre-strip.
Use SIP Account:	Which SIP account you want to use for oubound call
Finish	Cancel

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.

🔫 PCBest Networks SIP I	PBX v3.	12 (Admin 1	Fool)		
File Options Manage	Server	Operations	Auto Dialer		
	Spe	ecial Numbers			
B PBX	Nel	tworks		-	
	Sys	stem Options			
-					
😸 PBX Special Numbers	5				
Basic Magic Transfer Co	de				
Special ACD numbers fo	r extensior	ns to call:	×9000		
ACD Agent log in numbe	[*71			
ACD Agent log out numb	ACD Agent log out number:				
Pickup Group Short Cod	e:	[#		
Voice Mail Box Number:		[*91		
🖌 ок			🄾 Cancel]	

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:								
📢 PCBest Networks SIP PBX v3.12 (Admin								
File	Op	tions	Manage	Server	Operatio	ns		
		SIP Accounts(External Lines)						
😑 PB		Extensions						
Ē.		Hunt Groups(ACD)						
		Parking Slots						
		Rino	i Groups					

Steps to setup agents

🖶 ACD Hunt Grou	ps		
Automatical Call Dist Incoming calls will b for call center applic	ribution Hunt Group e automatically distri ation.	is a group of extensions that buted to extensions by order.	can answer calls. This feature is excellent
			Set Agents
Name	Туре	Agents	
<			>
Add	Edit	🔲 Delete	🖌 ок
🐣 ACD Agents			
ACD agents are the p in on an extension to The phone numbers f their code and passw	eople who can ansu answer calls. After th or logging in and ou ord for logging in an	wer Hunt Group's calls from a he work is done, an agent mu t can be set in Special Numb d out. You can set the promp	ny extensions. An agent must first log Ist log out before leaving. ers option. Usually agents will give ts here:
			Prompts
Code	Status	LoginTime	
3010	Offline	N/A	
🕂 Add	Edit	💻 Delete	🖌 ок

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:

🔜 Edit Extension	
 Basic Advanced Voice Mail Box Call Forwarding 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. Inable Call Recording 	
Method to answer ACD calls Once registered Once connected with pbx special number(*9000)	
Rest Interval(In Seonds): 0 Used for ACD Group when distributing calls to the extension. This will give the extension some seconds interval for next call.	nis
Update Extension 🔀 Cancel	.:

Enable agent call recording:

🔜 Edit Agent		
Agent informatio	n	
Name:	Grace	Optional. Any name. For example, Agent1, Bob, Grace
Code:	3010	Digits only. Must be unique. For example, 72000, 2100, 401
Password:	****	Password for logging in and out. Digits only.
🕑 Enable C	all Recording	
	🖌 ок	. Cancel

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: PCBest Networks SIP PBX v3.12 (Admin Tool)						
File Options	Mar	nage	Server	Operations	Auto	
		Mon	itor Groups	;		
		Call	Detail Reco	ord (CDR)		
Extensions		Plug	ins			
		Pick	up Groups			
102	_				_	

🔡 Monitor Grou	ıps		
A monitor group v the call.	vorks like a conference roor	n. The supervisor can mo	nitor extensions and even Barge-In to
Name	Number		
	1	1	
🔶 Add	Edit	🛑 Delete	🖌 ок

 Edit Moni	or Group	
Basic Pas	sword Keys to operate	Extensions
Name:	MonitorGroup1	Any name for this group. Must be unique in whole system.
Number:	*910	Special number for supervisor to call. If it is blank, you can still route call to monitor group in dialplan.
	ΟΚ	Cancel

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.

📢 PCBest Networks SIP PBX v3.12 (Admin Tool)							1)
File Optio	ons	Mar	nage	Server	Operations	A	uto
: D 🖬			Mon	itor Groups	;		
B PBX			Call Detail Record (CDR)				
Extensions			Plug	ins			
- 101 - 102		Pickup Groups					
1(13	_				_	

PC Best Networks SIP PBX Reference

🔜 Pickup Groups	;		
Pickup Group define call(in ringing status) If one agent wants t short code + agent o Defaultly the pickup	es a group of agents by just entering pic o pick up the ringing code or extension c short code is #. It c	s or extensions, in which, one ca kup short code. g call in another group, he or sho ode. san be set in the menu Server/S	an pick up another's e has to enter pickup pecial Numbers.
Name	Туре	Members	
<			>
🔶 Add	Edit	🔲 Delete	🗹 ок
🛃 Edit Pickup Gr	oup		
- Pickup Group			
Name: Pickup@	ìroup1	Any name, like group1	
Type: Extensio	n	~	
 ✓ 101 ✓ 102 			
V 103			
1	ОК	🔀 Cancel	

Pickup Group Short Code is defined in special number:

🖶 PBX Special Numbers	
Basic Magic Transfer Code	
Special ACD numbers for extensions to call:	*9000
ACD Agent log in number:	*71
ACD Agent log out number:	*72
Pickup Group Short Code:	#
Voice Mail Box Number:	*91
🖌 ок	X Cancel

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.

🕰 PCBest Networks SIP PBX v2.48						
File	Options	Manage	Ser	ver	Operations	Auto Dialer
			Special Numbers			
🕀 PB	X		Networks			
			System Options			
			Folders			
			Proxy Sites			
			Load Balance			
			Database Connection			ion
· · · · · · · · · · · · · · · · · · ·						

🔡 Database	Connection					
- SQL Server						
DB Type:	SQL Server Express 2005/2008					
DB Server:						
	Blank if db server is local, otherwise give the machine name.					
DB Name:	gtsippbxv2					
	Database name. Please create a database first time. It is "gtsippbxv2" deafultly.					
Authenticat	ion Type: SQL Authentication					
User Name:	sa					
	Defaultly it is "sa".					
Password:						
🥏 Test	Connection V K Kancel					

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.

Nicrosoft SQL Server Management Studio Expr	ess		
File Edit View Query Designer Tools Window C	Community	Help	
📜 New Query 🛅 📂 🗔 🥥 📴 🏙 🥻 🎽	Ŧ		
📴 🏢 🕺 🔛 Change Type 🕶 🦿 🦓 🕼 🛅	i .		
Object Explorer 🗸 🗸	× _ 1	able - dbo.auto_diale	r_done / Table
량 🥂 = 🖻 🝸		ID	Туре
🕑 🥅 System Tables	*	NULL	NULL
🕀 💷 dbo.auto_dialer_done			
🗉 🔳 idbo.auto_dialer_jobs			
Dec.cor_aco			

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:								
📢 PCBest Networks SIP PBX v3.12 (Admin Tool)								
File	Options	Manage	Server	Operations	Aut	o Dialer	Help	
Outbound Tasks								
E PBX						Add Te:	st Calls	
÷.	Extensions				_			

•	🖁 Auto Dialer Tasks 📃 🗖 🔀							
ļ c	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:							
/ t r	Auto Survey Calls: You o he customer chose an o ecord into database like	an specify an au ption, then forwa this: IVRMenu1,	to dialer task whic rd the call to anot 1;IVRMenu2,2;	ch presents an IVR menu ther menu, and so on. The	for the connected calls. Once e customer choices will be			
(s f	Call me back: Your custo stored into pbx's auto dia orward the call to an ext	omer can give a p Iler call jobs table. ension(or agent).	hone number to o . The pbx will call	call back on your website. the number, and once th	The phone number will be e call is connected, then			
(ii	CRM, Message Broadca ncrease your sale.	sting, and other a	applications: Broa	dcast your messages to a	large of phone numbers to			
	Name	Status	Type Code	SIP Account	Dial Plan			
_								
_								
-								
-								
	Add Edit Delete							

🗏 Edit Outbound Auto Dialer Task 📃 🗖 🔀					
An outbound task is a group of calls which has the same call action(dialplan). You can define as many as outbound tasks you want, but each task must have different type code. Each task will pull outbound call jobs, which has the same type code, from auto_dialer_jobs table, and process jobs on idle channels. Once the call is done, it will be saved back into auto_dialer_done table.					
Task Info]			
Name:	Task1	Any name. For example, Task1, Survey1			
🗹 Enable th	is task, so pbx will pick up jobs fr	om database.			
Type Code:	Type Code: 1 A small integer code to distinguish taks in call jobs table(1-32767). This value matchs to field 'Type' of auto_dialer_jobs table, and is used to distinguish outbound tasks.				
SIP Account:	Account1	SIP account used to call out			
Dial Plan:	TolVR1	Inbound dial plan to be used when call is connected.			
Stop Ring Aft	er: 20 seconds				
Max sim calls	for this task:	0 means no limit.			
	🖌 ок	Cancel			

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:

est Ne 🕶	t Ne <mark>📢 PCBest Networks SIP PBX v3.12 (Admin Tool)</mark>								
Option:	File Op	tions	Manage	Server	Operations	Aut	o Dialer	Help	_
	▶ 11						Outbou	nd Tasks	
•	B PBX						Add Tes	st Calls	

😸 Add Test Calls			
How many calls to add:	0		
Call Type Code:	1		
Caller:			
Callee:			
Call Start Time:	Sunday , April	04, 2010	v
	🖌 ок	Cancel	

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)	
🗹 Enable	Write to text file	Vrite to database
🗹 Enable	Write to text file	Vrite to database
Extension CDR		
🗹 Enable	Write to text file	✓ Write to database
Agent Login and Logo	ut Record	
🗹 Enable	Write to text file	✓ Write to database
	🖌 ок 🔰 🎽	Cancel

6.2 Networks

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

🔜 Network				
SIP Networks Audio Codecs Email SMTP Server Manager Port				
SIP IP Address:	Leave it blank if PBX works on all possible IPs			
SIP Port:	5060 Default: 5060			
RTP Port From:	19200 Default: 19200			
Internal:	8922			
STUN Server:	stun.pcbest.net STUN server is used to discover PBX's public IP			
DTMF Method:	Auto(Inband Audio or RFC 2833)			
Public IP Address:	If your machine is DMZ, or has fixed public IP address you want to use it in SIP			
You must restart this PBX to make the change effective!				
OK Kancel				

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:
-------	-------	------

🔜 Network		
SIP Networks Audio Codecs Email SMTP Server Manager Port 0,8,3 Sample: 0,8,3 means using G711u, G711a and GSM 0 = G711(ulaw) 8 = G711(alaw) 3 = GSM 102 = Speex 104 = iLBC(30 ms) 103 = iLBC(20 ms) 98 = G726-32 18 = G929(need special license)		
You must restart this PBX to make the change effective!		
OK 🏹 Cancel		

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:

💀 Network				
SIP Networks Audio Codecs Email SMTP Server Manager Port				
PBX will use this email account to send voice mail to individual's email address.				
Server: Sample: mail.abc.com,123.67.9.67				
Port: 25 Default: 25				
Email: Sample: abc@gmail.com				
Password:				
Enable SSL				
You must restart this PBX to make the change effective!				
OK K Cancel				

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):

💀 Network	
SIP Networks Audio Codecs Email SMTP Server Manager Port TCP Port: 9232	
You must restart this PBX to make the change effective!	

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:

.	PBX System Options
	General Outbound MOH Extensions Maintenance
	Audio Codec Used For Outline: Blank for auto choose.
	Audio Codec Used For Extensions: Sample: 0, or 8, or 18
	0 = G711(ulaw) 8 = G711(alaw) 3 = GSM 102 = Speex 104 = iLBC(30 ms) 103 = iLBC(20 ms) 98 = G726-32 18 = G929(need special license)
	If you changed the items marked with ***, you need to restart the PBX.
	OK Kancel

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:

🖶 PBX System Options	
General Outbound MOH Extensions Maintenance The Percentage of Outbound Channels: 50 %	
If you changed the items marked with ***, you need to restart the PBX.	

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

MOH Tab:

💀 PBX System Options
General Outbound MOH Extensions Maintenance Randomly play music files in MOH folder
If you changed the items marked with ***, you need to restart the PBX.

Extensions Tab:

🔜 PBX System Options	
General Outbound MOH Extensions Maintenance	
Max SIP Extension Register Expire Seconds: 3600 (60-3600)	
Allow extensions to try inbound dialplan	
If you changed the items marked with ***, you need to restart the PBX.	
V OK 🎇 Cancel	

Maintenance:

💀 PBX System Options
General Outbound MOH Extensions Maintenance Restart every day at: Sample: 03:00, or 21:10, or blank means disabled
If you changed the items marked with ***, you need to restart the PBX.

6.4 Folders and Logs

PCBest Networks SIP	PBX v3.12 (Admin Tool)			
File Options Manage	Server Operations Auto Dialer I			
	Special Numbers			
E PBX	Networks			
i Extensions	System Options			
⊡ Dial Plans	Folders			
ACD Agents	Proxy Sites			
🔜 Server Folders				
Folder of MOH(Music On Ho	ld)			
C:\temp\sdk\GTSIPPBXv3	\Install\moh\	Browse		
All wav files in this folder mu PCM 8K 16bit mono	st be one of the following three formats: Mulaw 8K 8bit mono AI	aw 8K 8bit mono		
← Folder of Log File				
Log file name is gtpbxlog.txt,	please specify where you want the log file to	o be saved:		
C:\temp\sdk\GTSIPPBXv3	\Install\log\	Browse		
Log Level				
💿 Disable 🛛 🔿 E	irror 🔿 Alert 🔿 Debug	🔘 Full		
Folder of Report Files				
C:\temp\sdk\GTSIPPBXv3	\Install\report\	Browse		
Folder of Recording Files				
C:\temp\sdk\GTSIPPBXv3	\Install\record\	Browse		
Folder of Voice Mail Box				
C:\temp\sdk\GTSIPPBXv3	\Install\vmb\	Browse		
You need to restart this PBX to make the change effective!				
🖌 (K 🏼 🎽 Cancel			

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:

P Network				
SIP Networks Audio	/Video Codecs Email SMTP Server RTP SIP Account Other			
SIP IP Address:	Leave it blank # BX works on all possible IPs			
SIP Port:	5060 Default: 5060			
SIP Protocol:	UDP TCP			
RTP Port From:	19200 Default: 19200			
RTP Port Space:	4 The port space between each channel. Default 4			
Internal:	8922			
STUN Server:	STUN server is used to discover PBX's public IP			
DTMF Method:	Auto(Inband Audio or RFC 2833) -			
Public IP Address:	If your machine is DMZ, or has fixed public IP address you want to use it in SIP			
You must restart this PBX to make the change effective!				
	✓ OK			

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 reregister 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
Γ	Basic DIDs Peer2Peer RTP
	Lit is a SIP trunk
	Display Name: Sample: Bob Wall, Company1, Trunk1
	UserName UserName: Sample: 7184773245, 1001, or Mike
	DomainServer SIP Domain: Sample: pcbest.net, voip.com
	ProxyServer SIP Proxy: Sample: pcbest.net, usually same as domain
	SIPProtocol SIP Protocol: UDP O TCP O SIPS(TLS)
	Authorization: Sample: 7845, usually same as UserName
	Password: Your secret code
	Expire Sec Expire Duration: 3600 In seconds, default is 3600 = 1 hour
	RegWithProxyServer Register with SIP proxy server to receive incoming calls
_	✓ OK ∑ Cancel

	Add SIP Account	_	x
Ba	asic DIDs Peer2Peer RTP DIDS		7
	Add	1	
	Rem	ove	
	AcceptOtherID Use above first DID or original caller id for outbound call(Usually check this box only this SIP account accepts from id rather than usemame) AppendExtenID Append extension's id to above first DID as caller id when extension calls out	y if your	
	V OK		.44

Add SIP Account	-	x
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 SI 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 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, an outgoing calls using this sip account will be forwarded to the extension's ip. Check the following option to enable this feature: MappedExten 1001	IP NAT. d	
V Cancel		

🖳 🛛 Add SIP Acc	ount	– – X
Basic DIDs Peer2Peer RTP		
UseRTP		
Use SRTP for calling out from this SIP account		
🚽 ок	💥 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 ext	tension			-		x
[Basic	Advanced	Voice Mail Box	Call Forwarding	Outbound	SIP	RTP			
	Use Exte Rea	erName ension: alName		(Sample: This is al	101, 1001. M so the user na	lust be u ame for \$	inique to t SIP exten:	the whole sion)	PBX,	
	Use	r Name:			(Sample: Bob	wall, M	like Smith)		
	Pas	sword:			(The passwo	rd for SI	P extensi	on registra	ation)	
	Ema	an Bil: Witter Torrol								
	Exte	ension Type: ualExtenDe	Normal stAddr	*						
	Virtu	ual Extension	Outbound Addre	ss or Number:	annala lika	0100450	C if you b	ave dafia		
	(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 Extensi	on		Cancel				.4
										¥
				Add an ex	tension					~
	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.									
	Met	hable Call Re	ecording Record							
	Meu	Once regi	stered O On	ce connected wit	h pbx special	number	(*9000)			
		-	RestSe	conds						
	Rest	Interval(In S	eonds): 0	Used exten	for ACD Grou sion. This will	ıp when give the	distributin extensio	ng calls to n some	this	
		OnlyAgentl CD agent mu MultipleCa	Login ust login to use thi 11	secor s extension. Usua	lds interval fo Ily only check	r next ca this opt	all. tion if it is	call cente	er.	
		now this exte	ension to accept m	nuitipie calls simult	aneously.					
-			🞺 Add Extensi	on	×	Cancel				

Add an extension	x
Basic Advanced Voice Mail Box Call Forwarding Outbound SIP RTP VMBON ✓ Enable Voice Mail Box Set Voice Mail Box	
Add Extension Kancel	

	Edit Voice Mail Box	-)	C
VMBPrompt					
Voice Prompt:	C:\SIPPBXv3\audio\Please-leave-your-message-after-beep.wav	Brov	/se		
	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				
VMBEmail	voice mail box instead of giving a voice me name.				
Email:					
VMBMaxLen; Max Duration:	The email address that voice mail audio file will be sent. eth 600 Max length of each voice mail in seconds. Default value is 600 seconds, which is 10 minutes.				
VMBPasswor Password:	rd Digits only. Maximum 4 digits.				
	VK Xancel				.4

	Add an extension	_		x
ſ	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 automion			x
	Add an extension	_ 1	_	

			Ac	id an ext	ension			
	Basic	Advanced	Voice Mail Box Call	Forwarding	Outbound	SIP	RTP	
	A1tP1 Outbo	honeNumb ound Caller I	er D:					
	Set a acco	n unique cal unt must ena	ller id for outbound calls able "Use above first DI	from this ext D or original	tension. Note	e: the ou	tbound dia	alplan's sip
	in cf Pin/ <i>i</i>	g_attr table	s, the <u>attr tyoe is 1002</u> e:	, item id is A code ex The code Extension tone, then the end.	the extension has t tension has t length can b user will nee dial the out	ion user to dial firs be 3-6 dig ed to call bound dia	name, va st into PB) jits. Like (pin code t alplan nun	Itue is pin code X, then call out. 331, 2468. first to hear a dial aber with '#' sign at
-			Add Extension		×	Cancel		

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	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 reregister. If you are seeing this extesion has become offline since this change, then please set it back to 0.	
-	Add Extension Kancel	_

			Add an ex	tension		_ □	x
Basic	Advanced SRTP Extension is e	Voice Mail Box	Add an ex Call Forwarding	tension	RTP		
		🖌 Add Extensi	on	X Can	cel]	

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

ng Dian	Plan 📃 🗖 🗙
Basic Time Schedule Extensions or Agents	
DialPlanName Plan Name:	Any name you like to give for this plan
CallDirection Call Direction: Inbound Outbound	Which call direction the plan is for
Catler Caller Number:	Blank if no limit on caller. Use * for any number, and ? for any one digit. You can use @ for calls on
Called Called Number:	specific IP/domain in SIP format. For example: *@192.168.0.2
CallPlan Plan Templete: Auto Attendant(IVR Menu)	
OutboundPreStrip Pre-strip:	Outbound called number pre-strip text For example: prestrip text for called number 9* is 9.
OutboundPrepend Pre-append: OutboundSIPAcct	Pre-append string after pre-strip.
Use SIP Account:	Which SIP account you want to use for oubound call
Alter SIP Account:	Second SIP account in case the first one is offline
Finish	

🖳 Dian Plan 🔄 🗖 🗙
Basic Time Schedule Extensions or Agents
✓ Enable TimeLimited TimeStartHour:TimeStartMinute TimeEndHour:TimeEndMinute Time: From To: Ex: 9:00 and 17:00, 19:30 and 6:30 Monday Tuesday Wednesday Thursday Saturday Sunday TimeDay
Finish Kancel

	Dian Plan	_ D X
Basic Time Schedule Extensio	ns or Agents	
 All Extensions or Agents ExtenMembers 	O Selected Extensions or Agents	Set Extensions or Agents
	Finish 🔀 C	iancel

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

Α Α	Add Hunt Group		-		x
Basic Agents Advance					
Name Name: Type Type: Linear ✓	Any name you like to give to this be unique. With a linear hunt group, calls a the lowest-numbered available I In a circular hunt group, the call "round-robin". In most-idle hunting, calls are al- whichever line has been idle the In most-skill hunting, calls alway who have bigger number of skill	s ACD hur re always ine. s are distri ways deliv e longest. s go to the level.	nt grou delive ibuted vered to e agen	p. Mu ned to p	ist
Music when waiting PlayMOH	MOHDir				
Play music when waiting. Music f	iolder:		Brows	se	
PromptQueuePosition Prompt queue position when wait	ting				
🖌 ОК	X Cancel				
• - A	dd Hunt Group		_		x
Basic Agents Advance Please select agents that can accept of AgentType AgentType Agent Type:	calls in this ACD group. se extension, calls will be delivere	ad to exter			
Agent (Calls will be Agent (Calls will be the ACD group name Agents: Agents	deliver to the extension which ag deliver to the agent who has sign ab y calling ACD login special nur Available Extension 1001 1002 1003	ient has lo ned in expl nber) ons or Age	nsions) ogged i licitly fo	n on) pr	

Add Hunt Group – 🗖 🗙
Basic Agents Advance
Voice Mail Box VMBOn Enable DTMF: Settings
Route Calls To Dialplan DialplanName
DTMF: To Dialplan:
Waiting Timeout or No Agents Available Waiting Timeout If calls waiting in the ACD queue for 0 (0 means never timeout) seconds or there are no agents logged in, then route the call to: WaitTimeoutTo CallForwardingType O Dialplan Defined Above
Choice for queue full MaxNumOfCalls
Maximum Number of Calls in Queue 10 0 means no limit CallForwardingPlan
O To Dialplan:
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

•	Edit Parking Slot	. 🗆 X				
Parking slot is used to park a call, which can be picked up later by dialing the parking slot's number.						
After an agent answerse Once the call is parked another agent can dial t	ed a call, he/she can input the parking slot's number to park this successfully, the agent's call will be automatically disconnected, the parking slot's number to pick up that call.	call. , and				
Basic Advance						
Name						
Parking Slot Name:	Any name. Sample: Slot 1					
DTMFStr Number:	Sample: *61, #10,					
Music On Hold PlayMOH Play music when	ı call parked					
Music files from:	MOHDir Browse	÷				
1	OK Cancel					

Edit Parking Slot
Parking slot is used to park a call, which can be picked up later by dialing the parking slot's number.
After an agent answersed a call, he/she can input the parking slot's number to park this call. Once the call is parked successfully, the agent's call will be automatically disconnected, and another agent can dial the parking slot's number to pick up that call.
Basic Advance
Voice Mail Box VMBOn VMBDTMF
Enable Voice Mail Box DTMF: Voice Mail Box Settings
Route Calls To Dialplan DiaplanName
DTMF: To Dialplan:
Waiting Timeout Wait Timeout
If the call waits in this Park Slot for 0 (0 means never timeout)
seconds, then route the call to: WaitTimeoutTo
Voice Mail Box Defined Above
✓ OK Cancel

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

Edit Ring Group	-		x
Basic Name Name: Please give any name to this rin Type: Type: Ring all destinations at one time Ring destinations by	ig grou order	q	
Destinations saved in cfg_dests			
	+ []	Add Edit Delet	e
Music when waiting MOHDir PlayMOH Play music when waiting. Music folder:	Brov	wse	
Voice Mail Box VMBOn Enable Settings Answer call first, then ring destinations. AnswerCallFirst			
V Cancel			

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

	PagingGroupEdit 📃 🗖 🗙
Basic Name: Number:	Name Please give a unique name to this paging group DID Unique number in system to reach this paging group. ie 120
Destinatio	ns Defined in cfg_dests
	Delete

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

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a.	2			Edit Mor	nitor (Group			-		x
	Basic Nan Nun	Password Na ne: Nu nber:	Keys to operate	Extensions Any name for Special num route call to	or this gr	roup. Must b supervisor t group in dia	oe unique o call. If it alplan.	in whole s	ystem ou ca	n still	
			🖌 ок			💥 Car	ncel				

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

🖳 Edit Monitor Group	-		x	
Basic Password Keys to operate Extensions KeyWhisper Key to start whispering to agent: 1 When monitoring, press this key to whispering to agent: KeyBargeIn When monitoring, press this key to speak. When monitoring, press this key to speak. KeyBargeOut When monitoring, press this key to jump or conversation, and choose another extension continue.	r to a ut of ion to	gent.		
✓ OK 🌋 Cancel				55.F

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

	Edit Agent 📃 🗖 🗙
Agent information Name Name: Name: Code Code: Password Password: SkillLevel: SkillLevel: 0 RecordCall Enable Call Recording	Optional. Any name. For example, Agent 1, Bob, Grace Digits only. Must be unique. For example, 72000, 2100, 401 Password for logging in and out. Digits only. 0 - 100. The bigger skill number, the higher priority agent has to get calls from ACD.
🖌 ок	

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

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e Ed	it IVR Menu		_ D X				
cfg_ivrs.MenuName IVR Menu Name: Prompt: Prompt: Sample sound: Welcome to ABC company. If you know the extension, please dial it now. Otherwise, press 1 for sales. press 2 for customer support. press 3 for billing department. Stay on the line for operator. Sound file must be 8k 8bit mono mulaw or alaw wave file, or 8k 16bit mono PCM wav file. Milliseconds to wait for DTME keys: 15000							
Menu Options							
cfg TWF subitems. Then play if blank, jump to action	directly) Action: IVRMenuActio		stinations MenuTransferTo				
N/A V	Browse To another menu		:BestAA 🗸				
N/A v	Browse To another menu	PC	;BestAA ✓				
N/A 🗸	Browse To another menu	i v PC	BestAA 🗸				
N/A v	Browse To another menu	PC	;BestAA ✓				
N/A 🗸	Browse To another menu	PC	;BestAA 🗸				
N/A v	Browse To another menu	i v PC	BestAA 🗸				
cfg_ivrs.DTMFAcceptExten Accept Extensions. Note: if you enable this option	cfg_ivrs.DTM , your customer may need to wait	AcceptEx	ctenWaitMS Milliseconds				
on the menu, because of the fact: if you have extensions begining with 1, like 101,102. Also you set above DTMF menu to accpet 1 to forward calls to ACD group. When users stay on this meanu, and input 1, pbx needs to determine if users want to reach ACD group or want to reach an extension. So pbx will wait above amount of milliseconds to see if users have more inputs. To avoid delay user experience, you can set your extensions begining with 6 or 7 for example (Leave 8 or 9 for outbound rule).							
🖌 ок	X Cancel						

9.13 cfg_autodialertasks

Name: the name of this auto dialer tasks

Enabled: 1 = enabled, 0 = disabled

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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

Edit Outbour	nd Auto Dialer Task 📃 🗖 🗙				
An outbound task is a group of calls which has the s You can define as many as outbound tasks you war Each task will pull outbound call jobs, which has the process jobs on idle channels. Once the call is done	ame call action(dialplan). it, but each task must have different type code. same type code, from auto_dialer_jobs table, and , it will be saved back into auto_dialer_done table.				
Name Name: Enabled Enabled this task, so pbx will pick up jobs fro	Any name. For example, Task 1, Survey 1 om database.				
JypeCode Type Code: 0 A small integer code to distinguish taks in call jobs table(1-32767). This value matchs to field 'Type' of auto_dialer_jobs table, and is used to distinguish outbound tasks. Please give a unique value each task.					
SIPACCT SIP Account: 120147_srtp v DialPlan	SIP account used to call out				
Dial Plan: Echotest RingTimeout Stop Ring After: 0 seconds	Inbound dial plan to be used when call is connected.				
MaxSimCalls Max sim calls for this task: 0	0 means no limit.				
🖌 ок	Cancel				

🖳 Edit Outbound Auto Dialer Task 📃 🗖	x
An outbound task is a group of calls which has the same call action(dialplan). You can define as many as outbound tasks you want, but each task must have different type code. Each task will pull outbound call jobs, which has the same type code, from auto_dialer_jobs table, and process jobs on idle channels. Once the call is done, it will be saved back into auto_dialer_done table.	
Basic Advanced	
EnableDetect	
Enable human voice and answering machine detection.	
Detection result will be saved into DetectResult column of auto_dialer_done table. 0 = Answering Machine 1 = Human voice 2 = Fax -1 = silence (no voice at all) -2 = detected voice, but unknown. -3 = unknown DiscAfterDetect Disconnect call after detection is done. Don't run the dialplan of this task.	
Otherwise(if uncheck), it will run the dialplan set in the Basic tab. Sometimes you want to do different things according to detection result. You can achieve this by writing your own plugin and set it in the dialplan.	
V Cancel	

9.14 cfg_pickupgroups

Name: the name of this pickup group

MemberType: 0 = extension, 1 = agent

Members: a list of extensions or agents
Pickup Group Name Name: Any name, like group 1 Member Type	
Type: Extension V	Any name, like group1
Members	
 ✓ 1001 ✓ 1002 ☐ 1003 	

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
C:\SIPPBXv3\audio\conf-join.wav Browse User leaving prompt: LeavePcompt
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
V Cancel

Edit Conference Room	
Basic Advanced DiscCall ✓ ✓ Disconnect last user's call when others have left.	
Host Password Protection Host PW Password:	
RecordCall Record Conference Conversation	
	4

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

	Call	Time Limit	Rule	_ □	X
Edit Rule - Dialplar Name: Seconds Second	Name can be dialplan can pull down the cor extensions, or sip acc It also accepts ? and characters. For example, if you wa 2 a mount of outboun limit all callers starting s: 0	name, extension nbo box to select ounts. *.? means any or ant to only give all d call time, you ca with 3, you can s Roundup Seco Some system ro here you set 60	name, sip accou already existing ne character, and 1 3 digits extensio an set name as 2 et 3*. conds onds: 60 oundup usage as 0. Some system r	unt name. You dialplans, d * means any m starting with ???. If you wan s minutes, so ound up to 6	r t
	🖌 ок	seconds.	X Cancel	I	