

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

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1 Introduction

PC Best Networks provides NO.1 Windows VOIP development kits to business customers. **PC Best IP-PBX** is a proprietary, Windows-based PBX system developed as a response to the growing needs of businesses who want to deploy voice-over-the-internet through a simple, easy to manage platform. There is no difference in the use of **PC Best IP-PBX** whether you are a one-person business or a company with tens or hundreds of staff. Powerful, flexible, light and user-friendly, **PC Best IP-PBX** can be set up and run within 30 minutes on any of your working computer, with great features like, Auto Attendant, ACD(Automatic Call Distribution), MOH(Message On Hold), Ring Group, Call Parking, Pickup Group, Conference, Auto-Dialer, Database Reports, and Plug-in.

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

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

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

Increase Productivity

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

Save time

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

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

Enhance business image

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

Improved Customer Services

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

PC Best IP-PBX FEATURES

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

Unified Communications and Mobility

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

Supported Codec (Voice Compression)

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

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

REQUIREMENT:

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

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

2 Installing PBX

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

Name 🔺	Size	Туре	Date Modified	
PBXv3Setup.msi	7,459 KB	Windows Installer P	4/6/2010 11:56 AM	
🔯 setup.exe	421 KB	Application	4/6/2010 11:56 AM	

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

🛃 SIP PBX v3
Welcome to the SIP PBX v3 Setup Wizard
The installer will guide you through the steps required to install PCBest Networks SIP PBX v3 on your computer.
WARNING: This computer program is protected by copyright law and international treaties. Unauthorized duplication or distribution of this program, or any portion of it, may result in severe civil or criminal penalties, and will be prosecuted to the maximum extent possible under the law.
Cancel < Back Next >

Click next.

🛃 SIP PBX v3		
Select Installation Folde	۲	
The installer will install SIP PBX v3 to the	following folder.	
To install in this folder, click "Next". To in	istall to a different folder, enter it be	elow or click "Browse".
Folder:		
C:\Program Files\PCBest Networks\S	IP PBX v3\	Browse
		Disk Cost
Install SIP PBX v3 for yourself, or for a	nyone who uses this computer:	
C <u>E</u> veryone		
Just me		
	Cancel < <u>B</u> ack	Next >

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

🛃 SIP PBX v3			
Confirm Installation			E .
The installer is ready to install SIP PBX v.	3 on your computer.		
Click "Next" to start the installation.			
	Cancel	< <u>B</u> ack	Next >

Then confirm the installation.

₿ SIP PBX v3	
Installation Complete	
SIP PBX v3 has been successfully installed.	
Click "Close" to exit.	
Please use Windows Update to check for any critical updates to the .NET Framewo	rk.
Cancel < <u>B</u> ack	<u>C</u> lose

It is done.

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

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

2. Run "msiexec /i PBXv3Setup.msi" in the command line, in the directory of two installation files:

Name 🔺	Size	Туре	Date Modified
PBXv3Setup.msi	7,459 KB	Windows Installer P	4/6/2010 11:56 AM
🔂 setup.exe	421 KB	Application	4/6/2010 11:56 AM

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

For V3, you n	leed to prepare databa	ase for PBX v3	before y	<mark>ou cai</mark>	<mark>1 run it.</mark>		
Services							×
File Action View	Help						
) 🗟 😰 🖬 🕨 = 🗉 🕨						
🆏 Services (Local)	Services (Local)						
	SIP PBX v3	Name 🛆	Description	Status	Startup Type	Log On As	^
		🆏 Remote Packet Cap	Allows to c		Manual	Local System	
	Start the service	🍓 Remote Procedure	Provides th	Started	Automatic	Network S	
		🍓 Remote Procedure	Manages t		Manual	Network S	
		🎇 Remote Registry	Enables re	Started	Automatic	Local Service	
		🍓 Removable Storage			Manual	Local System	
		🍓 Routing and Remot	Offers rout		Disabled	Local System	
		🆓 Secondary Logon	Enables st	Started	Automatic	Local System	
		🆓 Security Accounts	Stores sec	Started	Automatic	Local System	
		🎇 Security Center	Monitors s	Started	Automatic	Local System	_
		Server 😳	Supports fil	Started	Automatic	Local System	=
		Shell Hardware Det	Provides n	Started	Automatic	Local System	
		SIP PBX v3			Manual	Local System	-
		🎇 Smart Card	Manages a		Manual	Local Service	
		SQL Server (SQLEX	Provides st	Started	Automatic	Network S	
		SQL Server Active	Enables int		Disabled	Network S	
		SQL Server Browser	Provides S	Started	Automatic	Network S	
	!	SQL Server VSS Wri	Provides th	Started	Automatic	Local System	$\mathbf{\mathbf{v}}$
	Extended / Standard /						

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

For V2, you don't have to setup database in order to run.

V2 is NOT a service application, so you won't see it in Service list like above picture.

4. Setup **Database**.

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

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

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

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

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

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

64bit OS download one of the following:

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

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

Jan 1997 - 1997		Open
	0	Run as administrator
		Troubleshoot compatibility
gtsimplephone		
ዿ jre-7u11-windows-i58	<u>K</u> .	Scan for viruses
🙆 Microsoft Visual Studi	۲	Add to archive
SQLEXPR_x86_ENU	۱	Add to "SQLEXPRWT_x86_ENU-2012.rar"
SQLEXPR_x86_ENU-sp	۲	Compress and email
SQLEXPRWT_x86_ENU		Compress to "SQLEXPRWT_x86_ENU-2012.rar" and email
SQLEXPRWT_x86_ENU	8	Extract files
SQLManagementStud		Extract Here
		Extract to SOLEXPRWT x86_ENU-2012\

Choose new SQL server stand-alone installation:

1 SQL Server Installation Center	
Planning Installation Maintenance	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 Resources	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:

📸 SQL Server 2012 Setup	entry (i) fees (k) verses	- • ×
Setup Support Rules Setup Support Rules identify pi corrected before Setup can con	roblems that might occur when you install SQL Server Setup support files. Failures must be tinue.	
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	Operation completed. Passed: 7. Failed 0. Warning 0. Skipped 0. Show details >> View detailed report	Re-run
	< Back Next > Cancel	Help

📸 SQL Server 2012 Setup	
License Terms To install SQL Server 2012, you Setup Support Rules License Terms Feature Selection Installation Rules Instance Configuration	u must accept the Microsoft Software License Terms.
Disk Space Requirements Server Configuration Database Engine Configuration Error Reporting Installation Configuration Rules Installation Progress Complete	 which includes the media on which you received it, if any. The terms also apply to any Microsoft updates, supplements, Internet-based services, and support services.
Complete	 Copy Print I accept the license terms. Send feature usage data to Microsoft. Feature usage data includes information about your hardware configuration and how you use SQL Server and its components. See the Microsoft SQL Server 2012 Privacy Statement for more information.
	< Back Next > Cancel Help

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	n e ID for the instance of SQL	. Server. Instance II) becomes part of th	e installation path.	
Setup Support Rules License Terms Feature Selection	 Default instance Named instance: 	SQLEXPRESS			
Installation Rules Instance Configuration Disk Space Requirements Server Configuration	Instance ID: Instance root directory:	SQLEXPRESS C:\Program Files	\Microsoft SQL Serv	er\	
Database Engine Configuration Error Reporting Installation Configuration Rules Installation Progress	SQL Server directory: Installed instances:	C:\Program Files\	Microsoft SQL Serve	r\MSSQL11.SQLEXPRI	ESS
Complete	Instance Name	Instance ID	Features	Edition	Version
	1		< Back	Next > Cano	cel Help

PC Best Networks SIP PBX Reference

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.

1 SQL Server 2012 Setup						
Database Engine Configuration						
Specify Database Engine authen	ication 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 (SQL Server authentication and Windows authentication)					
Server Configuration	Specify the password for the SOL Server system administrator (sa) account	t				
Error Reporting	Enter password:					
Installation Configuration Rules						
Installation Progress						
Complete	Specify SQL Server administrators					
	Yonge-PCW7\Yonge (Yonge)	SQL Server administrators				
		to the Database Engine.				
	Add Current User Add Remove					
	< Back Next >	Cancel Help				

PC Best Networks SIP PBX Reference

📸 SQL Server 2012 Setup	
Error Reporting	
Help Microsoft improve SQL Se	erver features and services.
Setup Support Rules License Terms Feature Selection Installation Rules Instance Configuration	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	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	and the local difference	_ _ ×				
Complete Your SQL Server 2012 installation	on completed successfully with product up	idates.				
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						
Error Reporting						
Installation Configuration Rules	Details:					
Installation Progress	Viewing Product Documentation	for SOL Server				
Complete						
	Only the components that you use to view and manage the documentation for been installed. By default, the Help Viewer component uses the online library. SQL Server, you can use the Help Library Manager component to download do your local computer. For more information, see <u>Use Microsoft Books Online for ">http://go.microsoft.com/fwlink/?LinklD=224683>. Summary log file has been saved to the following location:</u>					
	C:\Program Files\Microsoft SQL Server\J PCW7 20130422 172129.txt	10\Setup Bootstrap\Log\20130422 172129\Summary Yonge-				
		Close Help				

Recycle Bin Adobe		Open
Reader XI	0	Run as administrator
		Troubleshoot compatibility
	К	Scan for viruses
Google		Open file location
Chrome		Add to archive
	۱	Add to "Ssms.rar"
	۱	Compress and email
	۱	Compress to "Ssms.rar" and email
		Pin to Taskbar
Kasnersky		Pin to Start Menu
퉬 Kaspersky Internet Security 2013		
Maintenance		Restore previous versions
Microsoft Silverlight		Send to
Microsoft Silverlight 3 SDK		
퉬 Microsoft Silverlight 4 SDK		Cut
Microsoft Silverlight 5 SDK		Сору
Microsoft SQL Server 2008		
퉬 Microsoft SQL Server 2012		Delete
Download Microsoft SQL Server Co	۲	Rename
🖳 Import and Export Data (32-bit)		Properties
y SQL Server Management Studio	-	riopentes
🃗 Configuration Tools		Computer
🍶 Integration Services		Computer

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 Serve	er 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		100				x
Select a page	🛒 Script 🔻 🖪	Help				
General						
	Database name:		sippbxv3			
	Owner:		<default></default>			
		devine				
	[v] Use full-text in	idexing				
	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	•			Add	Remove	•
				(OK Canc	el

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

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

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

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

sippbxv3 - Notepad	and a				
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) > <cfgsource>Database</cfgsource> <database> <enabled>true</enabled></database>	 Windows authentication mode Mixed Mode (SQL Server authentication and Windows authentication) Specify the password for the SQL Server system administrator (sa) account. — Enter password: 				
DBType: 0 = MS SQL 2005 Express, 1 = MS SQL 2005, 2 = MS SQL > <dbtype>0</dbtype>	Confirm password:				
<br DBServer: blank = local, otherwise give the server name or> <dbserver></dbserver>	Specify SQL Server administrators Yonge-PCW7\Yonge (Yonge) S				
<dbname>sippbxv3</dbname>					
<br 0 = SQL Authentication, 1 = Windows Authentication					
<authtype>0</authtype>					
<pre><!-- if AuthType is SQL Authentication, then please give the User for connecting SQL server--></pre>	Name and Password				
4					

6. Start SIP PBX v3 service

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

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	Allows to C Provides th Manages t Enables re Offers rout Enables st Stores sec Monitors s	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	
		Shell Hardware Det	Provides n	Started	Automatic	Local System	
		SIP PBX v3			Manual	Local System	-
		🎇 Smart Card	Manages a		Manual	Local Service	
		SQL Server (SQLEX	Provides st	Started	Automatic	Network S	
		SQL Server Active	Enables int	<i>-</i>	Disabled	Network S	
		SQL Server Browser	Provides 5	Started	Automatic	Network 5	
	Extended Standard	- Marchaller And Multi-	Fromues ut	Starteu	Maconiacie	Local System	

If you get popup, and cannot start the service:

Services (Local)					
SIP PBX v3	Name	~		Description	Status
	🔍 Secure Socket T	unneling Pro	otocol Service	Provides support for the Secure Socket Tunneling	
<u>Start</u> the service	🔍 Security Accourt	nts Manager		The startup of this service signals other services t	Started
	Security Center			The WSCSVC (Windows Security Center) service	Started
	🔍 Server			Supports file, print, and named-pipe sharing over	Started
	🔍 Shell Hardware	Detection		Provides notifications for AutoPlay hardware eve	Started
	端 SIP PBX v3				
	🔍 Skype Updater			Enables the detection, download and installation	
	🔍 Smart Card			Manages access to smart cards read by this com	
	🔍 Smart Card Ren	Joval Policy		Allows the system to be configured to lock the us	
	🔍 SNMP Trap	Services		Realistic log rescoge generated by local or to-	×
	🔍 Software Protec				
	SPP Notification		The SID DRY v2 of	envice on Local Computer started and then stopped	
	🔍 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	
	Superfetch				
	🔍 Symantec pcAn	where Host	t 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 Number	of events: 4,925		
Custom Views Administrative Events	Vumber 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
	🕕 Error	23/04/2013 9:41:03 PM	SIP PBX v3	0 +
Key Management Service	1 I I I I I I I I I I I I I I I I I I I	III		- F
Media Center	Event 0, SIP PBX v3			X
▷ Microsoft	Details			
Microsoft Office Diagnos	General Details			
Windows PowerShell	Friendly View	View		
Subscriptions				
	+ System			
	+ System			E
	- EventData			
		Service cannot be started	d. System.Exception: LoadConfig()	cannot
		access DB! at SIPPBXv3.5	SIPPBXMain.LoadConfig() in	
		C:\temp\projects\GTSIPF	PBXv3\SIPPBXv3\SIPPBXv3\SIPPB	XMain.cs:line
		1115 at SIPPBXv3.SIPPB	XMain.StartServer() in	-

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

퉬 audio	٠	Name		<u> </u>	
🍌 log			a tyt		
퉬 moh		E PDA03-10	y.ut		
퉬 plugin					
퉬 record					
퉬 report					
퉬 sdk					
퉬 temp					
퉬 vmb					
퉬 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:		adConfig() adConfig() adConfig() adConfig() adConfig() adConfig() adConfig()	cannot cannot cannot cannot cannot cannot	access access access access access access access	DB!. DB!. DB!. DB!. DB!. DB!.

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

PC Best Networks SIP PBX Reference

			WinPcap	1	plication to change the ct
All Progr	ams 👂	Z	Wireshark		
		6	Lexmark 7600 Series	•	📄 Manager Client Sample
		m	Microsoft ASP.NET 2.0 AJAX Extensions	×	🥰 SIP PBX v3 Admin Tool
🦺 start	Ø5	m	PCBest Networks SIP PBX v3	•	📄 SIP PBX v3 Reference

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

🔫 PC I	Best Networ	ks SIP PBX v	/3.56 (Adr	nin Tool)				- • ×
File	Options	Manage	Server	Operations	Auto Dialer	Search	Help	
	×							
St	art PBX							
[2013-04	4-22T22:39:3	7] DBVersion	3.56 AppV	ersion:3.56				
[2013-0	4-22T22:39:3	7] SIP PBX v	3 Service is	s NOT running! F	Please click Start	PBX menu.		

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

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 receiv	e incoming calls

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	

Basic	Advanced	Voice Mail Box C	Call Forwarding	
Ext	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	ssword:			(The password for SIP extension registration)
Em	ail:	Mike@mycompan	y.com	
Ext	ension Type:	Normal	-	
Virt	ual Extension	Outbound Address	or Number:	
(Us 123	e outbound d 3@sipprovider	ialplan rule to set ou r.com)	tbound numbe	er, or use SIP address format like:
IP	Extension Aut	hrization Type:	Proxy	•

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

RCBest Networks SIP PBX v3.11	. (Adr	nin Too	d)					- D X	🏉 Webphone - Wind 💻 🗉 🗙
File Options Manage Se	rver	Oper	ations A	uto Dialer	He	lp			🔾 🗢 🖉 http://www.p. 🔻 🔯
E 🕨 🗰									× Norton Q . @
⊟ • PBX	*		Exten	Name		Status	Contact	IdleFrom	
Extensions		•	101	Mike	8	Up(Idle)	192.168.101.43	2010-03-20 11:30	x 🔁 🕇
102			102 下	Poh	8	Offline		2010-03-20 11:30	
103 104	≡		103	Allen	8	Offline		2010-03-20 11:30	🖕 Favorites 🛛 🚖 🏈 Suggested Sites 🔻
SIP Accounts			104	Super	2	Offline		2010-03-20 11:30	🖉 Webphone
Dial Plans Inbound1 Inbound2 ACD Agents 1001 1002 - Hunt Groups(ACD) - ACD1 - Parking Slots Disa Groups	•	•		IT	1			,	
[2010-03-20 11:26:02] SIP PBX v3 Ser [2010-03-20 11:27:02] Plugin(Int Plugin [2010-03-20 11:30:46] Pugin(Int Plugin [2010-03-20 11:32:18] Extension 101 ju	vice is n Dem n Dem ust reg	s NOT ru o) Trace o) Trace jistered!	Inning! Pleas ISIP Accou Its contact in	e click Start nts: 1 Dialpla nts: 1 Dialpla fo: <sip:101< td=""><td>PBX i ans: 2 ans: 2 @192</td><td>menu. Extensions: 4 Extensions: 4 2.168.101.43:</td><td>7720></td><td></td><td></td></sip:101<>	PBX i ans: 2 ans: 2 @192	menu. Extensions: 4 Extensions: 4 2.168.101.43:	7720>		

3.2 Auto Attendant

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



Name	Sound File	DTMF Accept

	vanie.			lease give an u	nique name, like TVP	VI , D		
'rompt: C	: wwmer	iu.wav					Browse	
oress 2 for on nulaw or ali	nd: We custome aw wav	come to ABC con r support, press 3 e file, or 8k 16bit r	npany. It you kno for billing depart nono PCM wav f	ow the extension ment. Stay on th file.	n, please dial it now. (ne line for operator. So)therw ound fi	ise, press 1 for sale le must be 8k 8bit r	s. nono
Ailliseconds	s to wait	for DTMF keys:	15000	Millisecond	s.			
Menu Optio	ons							
DTMF		Then play:(if bla	ink, jump to actio	on directly)	Action:		Destinations	
1	•			Browse	Dial extension	•	101	•
N/A	•			Browse	To another menu	•		-
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 optio	on, vour custome	er may need to wait		Milliseco	nds
on the menu t determ millised beginir	menu, l to accpo ine 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 ex Is to ACD group. ACD group or w more inputs. To (Leave 8 or 9 for	tensions begining When users state vant to reach and avoid delay user outbound rule).	ng with 1, like 101,10 ay on this meanu, and extemsion. So pbx w er experience, you ca	2. Also I input ill wait n set y	you set above DT 1, pbx needs to above amount of your extensions	MF

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)	Extensions						
-	Hunt Groups(ACD)						
	Parking Slots						
	Ring Groups						
	IVR Menus						
	Conference Rooms						
	Dial	Plans					
Out							

Plan Name	Call Direction	Called ID	Templete	

Add a dialplan Inbound1.

Time Se	Extensions of Agenta		
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:	Ŧ	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	ks SIP PBX v	/3.10 (Adn	nin Tool)					
File	Options	Manage	Server	Operatio	ons				
1 ≥ 1	SIP A	Accounts(Ex	ternal Line	es)	1				
	Exter	nsions		1					
	Hunt	t Groups(AC	CD)						
	Parking Slots								
	Ring Groups								
	IVR Menus								
	Conference Rooms								
	Dial	Plans							
ACD Autor Incon for ca	Hunt Grou matical Call D ning calls will all center app	ps Distribution Hu be automation dication.	unt Group is cally distribu	a group of ited to exte	extensions by	ons that o y order.	can an: This fe	swer calls ature is e	s. excellent
Nam	ie	Туре	e	A	Agents				
-	5 Add						1		
Basic Ag	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 log In a c "round In most 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 never line has been idle the longest.						
Music w ▼ Play	Music when waiting Image: Music when waiting Music 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:

🖳 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 <<==
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.

Dian Plan	adula Estansione or Agente		
Time Sch	Edule 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 Account	t:	Ŧ	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:	OutPlan1	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)	v
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 PBX		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 IVR Menu Name:	IVR1	Please give an u	nique name, like "IVR1	I", "MainMenu".
Promot: c:\ivme	nu.way		•	Browse
Sample sound: We press 2 for custom mulaw or alaw wa	elcome to ABC compa er support. press 3 for ve file, or 8k 16bit mor	any. If you know the extension billing department. Stay on th no PCM wav file.	n, please dial it now. Oth ne line for operator. Sou	herwise, press 1 for sales. und file must be 8k 8bit mono
Milliseconds to wa	it for DTMF keys: 1	5000 Millisecond	s.	
Menu Options				
DTMF	Then play:(if blank	, jump 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 Exter	nsions. Note: if you en	able this option, your custome	er may need to wait	Milliseconds
on the menu, menu to accy detemine if u milliseconds t begining with	because of the fact: bet 1 to forward calls t sers want to reach A0 o see if users have m 6 or 7 for example(Le	if you have extensions beginir o ACD group. When users sta CD group or want to reach an ore inputs. To avoid delay use ave 8 or 9 for outbound rule).	ng with 1, like 101,102. ay on this meanu, and ii externsion. So pbx will er experience, you can	Also you set above DTMF nput 1, pbx needs to wait above amount of set your extensions

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.

Sasic	Advanced	Voice Mail Box	Call Forwarding	
Ext	ension:	104	(Sample: This is al	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)
Pas	sword:			(The password for SIP extension registration)
Ema	ail:			
Ext	ension Type:	Virtual		
Virt	ual Extension	Valooana Addre	ss or Number:	91234567
(Us 123	e outbound d @sipprovider	ialplan rule to set com)	outbound numbe	r, or use SIP address format like:
IP E	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				
1 ≥ 1	SIP A	Accounts(Ext	ternal Line	es)				
	Extensions							
	Hunt Groups(ACD)							
	Parking Slots							
	Ring	Groups						
	IVR N	Menus						
	Con	ference Roo	ms					
	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.

🤫 PCI	Best	Netw	vorks SIP	PBX v3.	12 (Adm	in	
File	Opt	tions	Manage	Server	Operatio	ns	
		SIP Accounts(External Lines)					
😑 PB3	Extensions						
	Hunt Groups(ACD)						
		Parking Slots					
		Ring) Groups				
	IVR Menus						
	Conference Rooms						
		Dial	Plans				
	Mon	itor an	oups				

 Parking Slots			
Name	Number		
🔶 Add	Edit	Delete	🖌 ок

🛃 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						
	Parking Slot Name	PK1	Any name. Sample: Slot 1				
	Number:	*61	Sample: *61, #10,				
	Music On Hold						
	🔲 Play music whe	en call parked					
	Music files from:		Browse				
-							
	OK 🏹 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 Aut	o Dialer			
; D 00	Special Numbers				
BX	Networks				
Extensions	System Options				
- 101	Folders				
102	Proxy Sites				
SIP Accounts	Load Balance				
Dial Plans	Database Connection				
PBX Special Numbers Basic Magic Transfer Code Magic transfer code is a code that extension can use it to transfer call to other extensions, agents or dialplans. For example, if blind transfer code is defined as "#, then in a call, the extension 101 can always press "#102 to transfer the call to 102. Or extension 101 can press "#1234567# to transfer the call to dialplan which has called number set to 1234567. (NOTE: in order to transfer the call to another dialplan. the extension side					
Blind Transfer Code: *#					
OK 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				
Any as GW doesn't check your authentication uncheck this option because it is a fake account					
OK Stancel					

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.

📰 Dian Plan				
Basic Time Schedule 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 For example: prestrip text for called number 9* is 9.			
Pre-append:	Pre-append string after pre-strip.			
Use SIP Account: ToGW 100	Which SIP account you want to use for oubound call			
🖌 ок	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:				
Lonf1) 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 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)					
	IP Extension Aut	nrization Type: Proxy 💌				
-	Add Extension Cancel					

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 PBX v3.12 (Admin Tool)					
File Options Manage	Server	Operations	Auto Dialer		
; Þ 🔟	Spe	cial Numbers			
BX	Net	works			
	Sys	tem Options			
-					
😸 PBX Special Numbers	5				
Basic Magic Transfer Co	de				
Special ACD numbers fo					
ACD Agent log in numbe	er:	[*71		
ACD Agent log out num	ber:	[*72		
Pickup Group Short Cod	le:		#		
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 Options Manage Server Operations SIP Accounts(External Lines) 😑 PB Extensions Đ Hunt Groups(ACD) Ð Parking Slots ÷. Rina Groups

🖬 ACD Hunt Groups					
Automatical Call Distribution Hunt Group is a group of extensions that can answer calls. Incoming calls will be automatically distributed to extensions by order. This feature is excellent for call center application.					
	Set Agents				
Name	Туре	Agents			
<					
Add	📔 Edit	Delete	СК		
ACD Agents					
ACD agents are the people who can answer Hunt Group's calls from any extensions. An agent must first log in on an extension to answer calls. After the work is done, an agent must log out before leaving. The phone numbers for logging in and out can be set in Special Numbers option. Usually agents will give their code and password for logging in and out. You can set the prompts here:					
·			Prompts		
Code	Status	LoginTime			
3010	Offline	N/A			
Add	📔 Edit 📃	Delete	🖌 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 Image: Porwarding 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. Image: Provide the call Recording Method to answer ACD calls	
Unce registered Unce connected with pbx special number("9000) Rest Interval(In Seonds): O Used for ACD Group when distributing calls to this extension. This will give the extension some seconds interval for next call.	
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 can monitor group.					
📢 PCBest Networks SIP PBX v3.12 (Admin Tool)					
Mai	nage	Server	Operations	Aut	
Monitor Groups					
	Call	Detail Reco	ord (CDR)		
	Plug	ins			
	Pick	up Groups			
	Ma	Manage Manage Call Plug Pick	Manage Server Monitor Groups Call Detail Reco Plugins Pickup Groups	Manage Server Operations Monitor Groups Call Detail Record (CDR) Plugins Pickup Groups	

Steps to setup a call monitor group:

Pickup Groups

Intermediate

A monitor group works like a conference room. The supervisor can monitor extensions and even Barge-In to the call.

Name

Number

Image: Contract of the call

Image: Contract

•5	Edit Mon	tor Group	
	Basic Pa	ssword 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.
		🖌 ок	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)						
File Options	Mai	nage	Server	Operations	Auto	
		Mon	itor Groups	5		
😑 PBX		Call Detail Record (CDR)				
Extensions		Plug	ins			
101 102		Pick	up Groups			
- 103	_				_	

🔜 Pickup Group	5					
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.						
Name	Туре	Members				
<			>			
🔶 Add	Edit	🔲 Delete	🗹 ок			
🔜 Edit Pickup Gr	oup					
← Pickup Group ——						
Name: Pickup(Group1	Any name, like group1				
Type: Extensi	on	×				
 ✓ 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			
			Nel	tworks		
				System Options		
				Fol	ders	
			Proxy Sites			
			Load Balance			
Database Connection					ion	
·						

🔡 Databas	e Connection
SQL Serve	er
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.
Authentica	ation Type: SQL Authentication
User Name	e: sa
	Defaultly it is "sa".
Password:	
Jes Tes	t Connection 🛛 🗹 OK 🛛 🎽 Cancel

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.

🍢 Microsoft SQL Server Management Studio Express							
File Edit View Query Designer Tools Window Comm	iunity ł	Help					
📜 New Query 🕞 😅 🖃 🎒 📴 📴 🐉 😁 🖕							
🔋 📰 🕺 📰 🛛 Change Type 🕶 ! 🥺 🕼 🖕							
Object Explorer 🚽 🗸 🗸	Т	able - dbo.auto_diale	r_done / Table				
📑 🛃 🔳 🏹		ID	Туре				
🗉 🗀 System Tables 🛛 🔼	*	NULL	NULL				
🕀 🔲 dbo.auto_dialer_done							

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

Steps to setup auto dialer tasks:						
📢 PCBest Networks SIP PBX v3.12 (Admin Tool)						
File Options Manage Server Operations	Auto Dialer Help					
Outbound Tasks						
PBX Add Test Calls						
Extensions						

Stong to get up out o dialer tasks

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

😸 Edit Outbou	ind Auto Dialer Task	
An outbound task You can define a Each task will pu process jobs on id	k is a group of calls which has the s many as outbound tasks you w II outbound call jobs, which has th dle channels. Once the call is dor	e same call action(dialplan). ant, but each task must have different type code. he same type code, from auto_dialer_jobs table, and he, it will be saved back into auto_dialer_done table.
Task Info		
Name:	Task1	Any name. For example, Task1, Survey1
🗹 Enable thi	is task, so pbx will pick up jobs fro	om database.
Type Code:	1 A small integr This value m used to distin	er code to distinguish taks in call jobs table(1-32767). atchs to field 'Type' of auto_dialer_jobs table, and is iguish 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 Afte	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 <mark>📢</mark> PCBest Networks SIP PBX v3.12 (Admin Tool)				
Option: File Options M	1anage Server	Operations	Auto Dialer	Help
			Outbou	nd Tasks
			Add Te:	st Calls

😸 Add Test Calls			
How many calls to add:	1		
Call Type Code:	1		
Caller:			
Callee:			
Call Start Time:	Sunday , April	04, 2010	•
	🖉 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	Vrite 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)			
File Options Manage	Server	Operations	Auto Dialer
Image: A state of the state	Sp	ecial Numbers	
E PBX	Networks		

🔜 Network	
SIP Networks Audio	Codecs Email SMTP Server Manager Port
SIP IP Address:	Leave it blank if PBX works on
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 PB	X to make the change effective!
[✓ OK Xancel

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 ∑ 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 Xancel

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 DK Kancel	

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	
Extensions SIP Accounts	System Options	
· Dial Plans	Folders	
■ ACD Agents	Proxy Sites	
🔜 Server Folders		
Folder of MOH(Music On H	ld)	
C:\temp\sdk\GTSIPPBXv3	\Install\moh\	Browse
All way files in this folder mu	st be one of the following three formats:	
PCM 8K 16bit mono	Mulaw 8K 8bit mono Ala	aw 8K 8bit mono
- Folder of Log File		
Log file name is gtpbxlog.txt,	please specify where you want the log file to	be saved:
C:\temp\sdk\GTSIPPBXv3	\Install\log\	Browse
CLog Level		
💿 Disable 🛛 🔿 I	rror 🔿 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 t	o 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

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

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

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

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

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

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



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

ew Project	1.4		8
Project types:		Templates:	
Visual C++ ATL CLR Genera MFC Smart Test Win32 Visual Visual Visual Visual Other Proj Test Project	+ al Device guages Basic C# J# iect Types cts :reating a C# class library	Visual Studio installed templates Windows Application Windows Control Library Crystal Reports Application Excel Workbook ASP.NET Web Application My Templates Search Online Templates	값 Class Library 값 Console Application 값 Device Application 값 Outlook Add-in 값 ASP.NET Web Service Application
Name:	ConfAssist		
Location:	C:\SIPPBXv3\Inst	all\plugin	✓ Browse
Solution:	Create new Soluti	on 🔹	Create directory for solution
	Solution Name:	ConfAssist	

How to setup a plugin project?

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

2. Then add reference to the project: Solution Explorer - Solution 'ConfAssist' (1 pro... ▼ ↓ Solution 'ConfAssist' (1 project) ConfAssist Properties References Add Reference... System.Xml Class1.cs

alution Explorer	🐼 Class View	Resource View
------------------	--------------	---------------

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

NET	COM	Projects	Browse	Recent			
Look	: in: 退 l	Install			•	G 👌 📂 🗔 -	
Nan	ne		^			Date modified	*
	SIPPBXGU	JI.vshost.ex	e			20/10/2013 10:02	
	SIPPBXSv	c.exe				23/04/2013 9:53 PM	Л
	SIPPBXSv	c.vshost.ex	e			07/05/2013 9:11 PM	Л
	SIPPBXv3	.dll				22/10/2013 10:03	
	sox.exe					10/04/2010 6:01 PM	1 =
5	yeti.mme	dia.dll				14/03/2013 1:47 PM	1
-	veti mn?	411			-	1//02/2012 1-/7 DA	1 -
-			11				+
File n	ame:	"GTAPIASI	M.dll" "SIP	PBXv3.dll"			•
Files	of type:	Component	Files (*.dll;	*.tlb;*.olb;*.o	ocx;*.e	xe;*.manifest)	•
					_		

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

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

3. Write a class which implements interface ISIPPBXPluginClient

```
🥠 ConfAssist - Microsoft Visual Studio 👘
 File Edit View Refactor Project Build Debug Data Tools Test Window Community Help
🔚 🕶 🕶 🖬 🚰 🛃 🎒 👗 🗈 🅦 🔊 🔹 🖓 🖛 💭 🗠 🖓 Debug
                                                        - Any CPU
                                                                            🗕 🧭 dtn
| 🖪 💁 🛦 | 律律| 🗏 😫 | 🗆 🖉 🗣 🔍 🔒 🔍 🖕
   Class1.cs Start Page
Server Explorer Toolbox
   🔧 ConfAssist.Class1
                                                               -
    using System;
     using System.Collections.Generic;
    using System.Text;
    -using SIPPBXv3;
    namespace ConfAssist
     {
          public class Class1 : ISIPPEXPluginClient
          {
          }
     L}
Lusing SIPPBXv3;
  namespace ConfAssist
   {
       public class Class1 : ISIPPEXPluginClient
  F
       1
           public const string RESULT DISCONNECTED = "Disconnected";
            public const string RESULT ERROR = "Error";
            private string m strName;
            private string m strType;
           private ISIPPEXPluginHost m Host;
           public Class1()
  È
            {
                m_strName = "ConfAssist":
                m strType = "IVRMenu";
            }
```

```
public void Start()
{
    //get the caller and callee number
    string caller_addr = Host.Channel.caller_num;
   string callee addr = Host.Channel.callee num;
    string caller_num = GTAPIASM.GTAPIEnv.GetSIPAddressInfo(1, caller addr);
    string callee_num = GTAPIASM.GTAPIEnv.GetSIPAddressInfo(1, callee_addr);
    //if it is a job of auto dialer task for human/answer machine detection,
    //use the following code to access detection result:
    1*
    if (Host.PBX Channel.call job != null)
    {
        switch (Host.PBX_Channel.call_job.DetectResult)
        {
           case 0: // = Answering Machine
               break;
```

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

Let us open MyPBXPlugin1.



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

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



GTSIPPBX refers to V2's GTSIPPBX.exe.

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

Plugin Built-in Methods and Functions

Host.DisplayMenu

Display a menu, and accept DTMF inputs.

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

Return: DTMF string

Host.DisplayMenuEx

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

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

Return: DTMF string

Host.PlayAudio

Play an audio file.

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

Return: DTMF string

Host.PlayAudioEx

Play an audio file.

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

Return: DTMF string

Host.RecordAudio

Record an audio file.

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

Return: DTMF string

Host.DetectDTMF

Detect DTMF keys.

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

Return: DTMF string

Host.HangUp Disconnect call.

Format: int HangUp();

Return: none

Host. WriteLog

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

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

Return: none

Host.ToExtension

Transfer this call to extension.

Format: bool ToExtension(string exten_no);
exten no: the extension number

Return: bool, if succeed.

Host.ToIVRMenu

Send this call to IVR menu.

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

Return: bool, if succeed.

Host.ToMonitorGroup

Send this call to monitor group.

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

Return: bool, if succeed.

Host.ToHuntGroup

Send this call to ACD group.

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

Return: bool, if succeed.

Host.ToRingGroup Send this call to ring group.

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

Return: bool, if succeed.

Host.ToVoiceMailBox

Send this call to voice mail box.

Format: bool ToVoiceMailBox(string exten_no); exten no: the extension number of which voice mail box being used.

Return: bool, if succeed.

Host.ToConferenceRoom

Send this call to a conference room.

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

Return: bool, if succeed.

Host.ToPlugin Send this call to another plugin.

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

Return: bool, if succeed.

Host.ToNumber

Forward this call to another phone number.

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

Return: bool, if succeed.

Sample:

//to another outside number //Host.ToNumber("<sip:123@192.168.1.100>", null); //Host.ToNumber("<sip:6781992@callcentric.com>", null); //or //SIPAccount acct1; //acct1.DisplayName = "any"; //acct1.UserName = "1234"; //acct1.DomainServer = "sip.callwithus.com"; //acct1.ProxyServer = "sip.callwithus.com"; //acct1.AuthName = "1234"; //acct1.Password = "xxxxx"; //Host.ToNumber("655112", acct1);

Host.DisconnectExtension

Disconnect(hang up) extension's call.

```
Format: bool DisconnectExtension(string exten_no);
exten no: extension number.
```

Return: bool, if succeed.

Host.SetChanRunPlugin

Set channel to run another plugin.

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

Return: bool, if succeed.

Host.ResetChannel

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

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

Return: bool, if succeed.

Host.SetChanInConferenceRoom

Set channel into conference room.

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

Return: bool, if succeed.

Host.CreateConferenceRoom

Create a conference room

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

Return: bool, if succeed.

Host.DestroyConferenceRoom

Destroy a conference room

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

Return: bool, if succeed.

Host.GetConferenceRoomIndex

Get conference room index.

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

Return: the index of conference room.

Host.GetConferenceRoomHandle

Get conference room handle.

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

Return: the handle of conference room.

Host.GetConferenceRoomByName

Get conference room handle.

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

Return: the class of conference room.

Host.SetUserObj

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

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

Return: if succeed.

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

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

Return: the object

Host.GetChanUserObj

Get channel's object

Format: object GetChanUserObj(int ch, int idx); ch: channel index based on 0. idx: index of the object

Return: the object

Host.SetChanUserObj

Set channel's object

Format: bool SetChanUserObj(int ch, int idx, object obj); ch: channel index based on 0. idx: index of the object obj: object

Return: if succeed.

Host.StartPBX Start PBX

Format: void StartPBX();

Return: none.

Host.StopPBX Stop PBX

Format: void StopPBX();

Return: none

7.2 Manager Client Application (V3 only)

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



Currently manage client can receive those events:

Call events on each channel.

Registration events of extensions.

Agent login and logout event.

Manage client can also do those actions:

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

Methods and Events

ServerConnected

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

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

Return: none

Channel related methods and events:

SIP PBX v3 Server			
Server IP: 127.0.0.1	Server Port: 9232	Disconnect	
Basic ACD Agent Call Contro	ol Extension Supervisor Channel	ls Conference	
Based Channels 0	alaan indoo (kaan ()). 1 - all		
	cnan index(from U), -1 = all		

ResetChannel

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

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

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

Agent related methods:

🖳 PCBest SIP PBX v3 Manager Client Sample	
SIP PBX v3 Server Server IP: 127.0.0.1 Server Port: 9232 Disconnect	
Basic ACD Agent Call Control Extension Supervisor Channels Conference	_
Agent login and logout	
Agent Login Agent Code: Agent Logout On Extension:	
Reset ACD Group Name:	
node nob	
Events From SIP PBX v3 Server:	
[2013-06-11T21:02:49] Server Connected! [2013-06-11T21:02:49] OnCallIdle 0 [2013-06-11T21:02:49] OnCallIdle 1	
[2013-06-11T21:02:49] OnCallIdle 2 [2013-06-11T21:02:49] OnCallIdle 3	
[2013-06-11T21:02:49] OnCallIdle 4 [2013-06-11T21:02:49] OnCallIdle 5 [2013-06-11T21:02:49] OnCallIdle 6	

AgentLogin

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

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

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

OnAgentLog

The event when an agent login or logout.

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

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

Return: none.

GetAgentStatus

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

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

Return: none, it will trigger the event following.

OnAgentStatus

The event for agent status.

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

Return: none

ResetACD Reset a ACD group

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

Return: none

Call Control Related Methods and Events:

Server IP: 127.0.0.1	Serve
asic ACD Agent Call Control	Extension
Call Control	
Make Answer	Hang Up
Make	
Channel ID: Sta	rt from 0
Channel ID: Sta	rt from 0
Channel ID: Sta Caller: Sample: <sip:123@at< td=""><td>rt from 0 oc.com></td></sip:123@at<>	rt from 0 oc.com>
Channel ID: Sta Caller: Sample: <sip:123@at Callee:</sip:123@at 	rt from 0 bc.com>

MakeCall

make a call out

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

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

MakeCall

make a call out

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

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

MakeCall

make a call out

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

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

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

AnswerCall

answer an incoming call on a channel

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

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

HangupCall

disconnect call on a channel

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

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

HangupCall

disconnect call on a channel

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

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

HoldCall hold call on a channel

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

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

TransferCall

blind transfer call on a channel

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

Return: none

TransferCall consult transfer call on a channel

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

Return: none

OnCallConnected

This event is triggered whenever there is a call connected

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

Return: none

OnCallIdle

This event is triggered whenever a call got disconnected

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

Return: none

OnCallRinging

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

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

Return: none

OnCallDialing

This event is triggered whenever a outbound call is dialing.

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

Return: none

OnCallOffered

This event is triggered whenever there is a new incoming call

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

Return: none

Channel related methods and events:

Serve	IP: 127.0.0).1	Server Port: 9	232	Disco
Basic	ACD Agent	Call Control Exte	ension Supervis	or Channels	
-0.4	T C				
Char	ge I wo Lalis nnel 1 ID:				
Cha	nnol 2 ID:	Conne	ct		
Cha	iner 2.10.		12		
1					
Rur	Dialplan on (Channel			
Rur	n Dialplan on (annel:	Channel Dialplan Nan	ne:		Run
Rur	n Dialplan on (annel:	Channel Dialplan Nan	ne:		Run
Rur Cha	n Dialplan on (annel:	Channel Dialplan Nan fer Code on	ne:		Run

MagicTransfer

magic transfer call

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

Return: none

BridgeTwoCalls

brige the calls on two channels

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

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

RunDialPlan

run a dialplan on the channel

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

Return: none

Server IP: 127.0.0.1	Server Port: 9232	Disconnect
Basic ACD Agent Call C	ontrol Extension Supervisor Chann	els Conference
- Call Extension		
1. COLL ALELIND ICL.	2	
Call Extension:		
Call Extension On Channe	el:	
Call Extension: On Channe	el:	
Call Extension On Channe Extension Call Out	ek	

Extension related methods and events:

MakeExtensionCall

make a call to extension on specific channel

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

Return: the command id

ExtenCallOut

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

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

Return: none

ExtenCallOutEx

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

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

Return: none

OnExtenStatus

The event to reflect extension status

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

Return: none

Supervisor feature to monitor extension's call.

SIP PBX v3 Serv	/er				1.55
Server IP: 127.0.0.1		Server Port:	9232	Disconnect	
				o (_]	
Basic I ALLIAR	ent I Lall Control	Extension : SUDEIY	SOF I Channels I	Lonference I	
Basic ALD Ag	ent Call Control		sor Channels	Conference	
Basic ALD Ag	all Monitoring		sor Channels	Lonference	
Basic ALD Ag Supervisor C Supervisor I	ent Call Control all Monitoring Extension:	A supervisor e	sor Channels xtension that alre	conterence	iumb
Basic ALD Ag Supervisor C Supervisor I Monitoring E	ent Lall Lontrol all Monitoring Extension:	A supervisor e	xtension that alre	ady called to pbx ACD n	iumb
Basic ALD Ag Supervisor C Supervisor I Monitoring B	ent Call Control all Monitoring Extension: Extension:	A supervisor e An agent exte	sor Channels xtension that alre	ady called to pbx ACD n	iumb

MonitorCall

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

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

Return: none

OnCallMonitoring Monitoring call event

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

Return: none

Server IP: 127.0.0.1	Server Port: 9232		Disconnect	
asic ACD Agent Call Control	Extension Supervisor	Channels Confe	erence	
Conference Room Name: Te: Channel Conference Operatio Channel ID(from 0): Add this channel to	stConf1 n above conference room (Ful	Create it	Destroy it	Get Status
Add this chan	nel to above conference roo	m (Listening only	model)	
				-
Remo	ve this channel from above o	conference room		

Conference related methods and events

CreateConferenceRoom

Create a conference room on PBX dynamically.

Format: void CreateConferenceRoom(string conf_name)
conf name: the name of conference room

Return: none

DestroyConferenceRoom

Destroy a conference room on PBX dynamically.

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

Return: none

SetChanInConferenceRoom

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

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

Return: none

SetChanConferenceBitMask

Set channel's bitmask in conference room. Set channel's output when in conference room.
This function is used to disable the chan's output voice to other channels in the same
conference.
Default channel mask is always 0xFFFFFFFF, which means output to all other
channels in the conference room.
Every bit marks a channel. If the bit is 1, its voice can output to the channel.
The First channel in the conference room is 0x01.
The second channel in the conference room is $0x02$.
The third channel in the conference room is 0x04.
So if you want the channel's output goes to the first channel, and the third channel,
you can set this for this channel:
SetChanConfMaskch, $0x05$); //which $0x05 = 0x01 + 0x04$
Another example,
1st channel is connected with Agent. (Channel Index is 0, and it is the first channel set to the conference room)
2nd channel is connected with Customer. (Channel Index is 1, and it is the second
channel set to the conference room)
3rd channel is supervisor. (Channel Index is 2, and it is the third channel set to the
conference room)
They are all in the same conference room. Regularly if don't set anything, they can
hear each other.

If supervisor only wants the agent hear his voice, not the customer, you can do so: SetChanConfMask(2, 0x01);

It means that only the first channel get his voice.

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

Return: none

GetConferenceRoomStatus

Trigger the conference room event to get the status

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

Return: none, but the event OnConferenceRoomStatus will triggered.

OnConferenceRoomStatus

The event to receive current conference status

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

Return: none.

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

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

command parameter1|parameter2|parameter3.....

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

For call dialing out, you will receive command like this: **CallDialing** channel-id|unique-id|caller|callee|dialplan|recording-audio-filename If remote ringed for outbound call, you will receive: **CallRinging** channel-id|unique-id|dialplan|recording-audio-filename

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

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

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

7.3 Database Development (V3)

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

Tables:

cdr_xxx are CDR tables.

auto_dialer_xxx are auto dialer tables.

cfg_xxx are PBX configuration tables.

If you want to develop your own user interfaces, like web interface, to work with PBX, cfg_xxx tables are the tables you mostly need to deal with. Each cfg table has a field **ModTag**, which makes this record's status.

If you add or change a record, you need to set ModTag to 1. PBX service will later refresh its memory and set this tag back to 0.

If you want to remove(delete) the record, you need to set ModTag to 2. PBX service will later delete it from table.

When ModTag is 0, then it means there is no change on this record.

log_xxx are PBX real-time log table.

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

status_xxx are PBX real-time status table.

voice_mailbox is voice mailbox table.

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

lere is the full list of database ta
🖃 🚞 Tables
표 🚞 System Tables
표 🔲 dbo.auto_dialer_done
표 🧾 dbo.auto_dialer_jobs
표 🥅 dbo.cdr_acd
🗉 🥅 dbo.cdr_agent
표 🔲 dbo.cdr_exten
🗉 🔲 dbo.cdr_pbx
표 🥅 dbo.cfg_agents
표 🥅 dbo.cfg_autodialertasks
표 🧾 dbo.cfg_conferencerooms
표 🥅 dbo.cfg_dests
표 🥅 dbo.cfg_dialplans
표 🧾 dbo.cfg_extensions
표 🥅 dbo.cfg_huntgroups
표 🥅 dbo.cfg_ivrs
표 🧾 dbo.cfg_ivrsubitems
표 🧾 dbo.cfg_monitorgroups
표 🧾 dbo.cfg_parkingslots
표 🧾 dbo.cfg_pickupgroups
표 🧾 dbo.cfg_ringgroups
표 🧾 dbo.cfg_sipaccounts
표 🥅 dbo.cfg_sys
표 🥅 dbo.log_sys
표 🥅 dbo.opt_cmd
표 🔲 dbo.status_acd
표 🔲 dbo.status_agent
표 🔲 dbo.status_channel
표 🧾 dbo.status_conferenceroom
표 🔲 dbo.status_exten
표 🥅 dbo.status_parkingslot
표 📰 dbo.status_sipaccount
🕀 💷 dbo.voice_mailbox

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