

E d g e - c o r e E

Powered by Accton

VST3300 Series
IP PBX

Operation Manual

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VST3300 Series

All-In-One IP PBX

Operation Manual

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1. Safety Instructions

Warning

1. Do not attempt to service the product yourself. Any servicing of this product should be referred to qualified service personal.
2. To avoid electric shock, do not put your finger, pin, wire, or any other metal objects into vents and gaps.
3. To avoid accidental fire or electric shock, do not twist the power cord or place it under heavy objects.
4. The product should be connected to a power supply of the type described in the operating instructions or as marked on the product.
5. To avoid hazard to children, carefully dispose of the product's plastic packaging.
6. The phone line from Telecom Company or traditional PBX should always be connected to the LINE or FXO connector. Incorrect connection to the PHONE/FAX or FXS connector may damage the product.

Read all the instructions before using this product to avoid damage to the product. First read the Quick Installation Guide for installation followed by this manual for adjustments and advanced settings. If this document is in digital format, print out the whole manual for easy installation.

2. Preface

VST3300 Series was designed with the latest Voice Over IP (VoIP) technology. It acts as a traditional telecom PBX, IP-PBX, SIP Server and SIP Client, but integrated together in one device. It achieves high quality voice communication and FAX service by using low cost Internet connection and so saves money while providing additional value added services. This product features easy installation, high mobility and versatile applications.

VST3300 Series models are listed below:

Model Name	Analog Channels		SIP Channels (Default)	
			SIP Lines	SIP Trunks
VST3305	3	1 FXO + 2 FXS	8	2
VST3306A	4	2 FXO + 2 FXS	12	2
VST3306B	4	4 FXS	12	2
VST3306C	4	4 FXO	12	2
VST3306D	4	1 FXO + 3 FXS	12	2
VST3310	16 Port Base Unit		24	2

Note: Volume of SIP Channel can be increased with purchase of expansion license keys. Please contact your distributor for more information.

Optional accessories for the VST3318 model:

Modules	Description
MP3208+	4FXS + 4FXO
MP3008+	8FXS
MP3108+	8FXO

VST3300 Series is equipped with commercial PBX, IP-PBX, SIP Server and SIP Client functions.

SIP devices (Gateway, IP Phone, Softphone, Wi-Fi phone, etc.) that comply with RFC3261 standards can register to VST3300 Series devices and becoming part of the whole IP-PBX system. Furthermore, making SIP call is the same as dialing between extensions.

VST3300 Series also acts as a SIP Client by registering to ITSPs (Internet Telephony Service Provider). Thus, users can also benefit from additional services provided by ITSPs, such as free calls inside ITSP's network and ITSP's cost savings on calls to PSTN.

3. Introduction of Features

VST3300 Series is designed to be the most advanced fully distributed, stackable and networking structure. It not only provides traditional PBX functions, but also the latest IP switching technology for users around the world to share voice resources from the IP-PBX. Its Lines (extensions) can connect an analog phone set, a fax machine or SIP device complying with RFC3261. Its Trunk can connect to traditional CO line (PSTN) and new generation Soft Switch. Below is a brief feature description.

3.1 Trunk Features

- **Trunk Groups**

This function divides trunk into 2 groups, a local trunk and a SIP IP Trunk. Users can make calls via a trunk group according to what call type they want.

- **Auto Trunk Selection**

When an extension wants to make a trunk (PSTN or SIP) call, the user need only dial the Trunk Group access code such as "9". Then, the system will select a free trunk port in the selected trunk group automatically depending on the needed use.

This function works for one machine or all other machine that are cascaded together.

- **Specific Trunk Seizure**

When an extension wants to make a trunk (PSTN) call from a remote site, the user can seize a specific remote trunk in advance. When the dial tone of remote trunk is heard, they can then dial the PSTN number. When this function is used, extension barring is still under effect.

- **Trunk Class**

Trunks can be grouped by detail for different trunk connection types, special numbers or devices such as a door phone or for paging (speaker). Users can select a specific trunk according to their purpose.

- **Call Barring**

There are 6 barring classes for each extension. Each barring class can apply to each extension to limit calls.

- **Transit Call**

PSTN incoming calls can be transited out to another PSTN via the device under password control. Users who aren't at the office can save on PSTN calls via the VoIP network.

- **Speed Dial**

This machine provides 100 sets of Abbr. Dial numbers. Numbers that are too long can be defined as a speed dial number. User can then quickly dial without remembering long numbers.

- **Direct Outward Dial**

When the extension is seldom to make call to other extension, VST3300 Series can make this extension to seize trunk automatically when phone set is pickup. User can make trunk call directly without selecting or seizing different trunk.

3.2 Line Features

- **Call Pickup Group**

Users can be grouped for Call Pickup purposes, such as Sales, R.D. with a total of 10 groups maximum. Users can pickup others' extension belonging to the same group but not those belonging to others.

- **Group Hunting**

A series of extensions can be grouped together and assigned an access code. Incoming calls with that access code will be directed to the extension in that group that is idle.

- **Hot Line**

Extension can hotline to specified extension or PSTN number. When this hot lined extension is picked up, the call route to specified extension or PSTN number is built.

- **Zone Paging**

The (trunk) FXO port of VST3300 Series can connect with a paging (speaker) system. Users can then make remote speaker broadcasts.

3.3 Phone Features

- **Call Transfer**

VST3300 Series can make consulted transfers. An extension can transfer an incoming call to any extension in the system, either at a local or remote site.

- **Call Park**

When incoming calls are not answered by the right person, or the person answering doesn't want to talk, the person answering can activate Call Park. The incoming call is held and the caller will hear music. The person who answered can then give the message of the caller to the right person by using the broadcast system. The right person can retrieve the incoming call with his phone set using the park ID.

- **Music on Hold**

When an incoming call is held or transferred, the system plays music for the caller indicating his waiting status.

- **Automatic Call Release**

VST3300 Series provides a busy tone detection function. If activated, this function ensures a call release will be successful when (1) incoming trunk (PSTN) call is answered by auto attendant (2) incoming trunk (PSTN) call make PSTN-IP-PSTN

call. Busy tone detection can decide the trunk state of (PSTN) call to activate Call Release.

- **Conference Call (optional)**

When work with the Conference Bridge (optional), conference calls are available.

Each Conference Bridge provides one meeting room with 4 maximum lines or trunks to join a conference. The volume of meeting rooms is unlimited.

3.4 Extension Features

- **Call Forward**

Extensions can be forwarded to any extension (local or remote site). Forwarding calls for All Calls, Busy, No Answer, Busy or No Answer.

- **Offnet (Mobile Phone, PSTN) Call Forward**

This is a unique design of the VST3300 IP-PBX. When a user is unable to answer their extension on the desktop, the Call Forward function can be configured so that the incoming call is sent to the next device, such as mobile phone in the PSTN.

Never miss incoming calls. This rule for call forward also includes: All Calls, Busy, No Answer, Busy or No Answer.

- **Secretarial Intercept**

In manager/secretary situations, you can set all incoming call to manager to be filtered by the secretary. The secretary can then decide to forward calls to the manager.

- **Timed Alarm**

An extension can set an alarm via keypad on phone set. The phone set will ring notifying you when time is reach; like Morning Call.

- **Do Not Disturb (DND)**

When set, an extension can make outgoing calls, but incoming call are restricted.

3.5 System Features

- **Flexible Extension Numbering Plan**

Extension numbers can be flexible configured according to needs. Users can consider organization and original numbering plan of the company, user friendliness and easy of memorization.

- **Selectable Tone/Ring Specification**

VST3300 IP-PBX can select suitable Ring Back Tones, Dial Tones and Busy Tones to fit local telecom specifications.

- **Call Detail Recording**

The system provides trunk detail recording of (PSTN) calls that can be used for billing purposes.

- **Emergency Telephone**

For models that have symmetric module (VST3305, VST3306A, VST3306D, VST3318) they can connect PSTN and an extension directly for emergency call should there be a loss of power.

■ **Built-in DISA**

DISA is built into this product for each incoming trunk and IP call. The greeting section can be recorded by phone set.

■ **Built-in Dialer**

An ITSP dialer is built into this product. The system administrator can specify a suitable ITSP (Internet Telephony Service Provider) anytime for international or long distance calls. End users (extensions) of the system need not change their dialing behavior.

■ **Behind PBX Operation**

The device is also able of connecting to traditional analog PBXs. Trunk ports and Line ports of a traditional PBX can integrate with the VST3300 IP-PBX system to fulfill the complete VoIP switch system and so share voice resource in different places.

■ **Voice Mail Integration**

This product can connect to other commercial brand Voice Messaging devices with extension ports, then it is able to support function like voice guidance and voice messaging.

■ **Networking & Stacking Service**

This product is designed with distributed architecture, making reliability higher than normal Client-Server structure IP-PBXs and increasing the ability to grow and fit network requirements. Multiple units can create a single IP-PBX in a single location of a LAN or, similarly, add multiple units in different locations to work together as a single PBX system in a WAN.

■ **Private IP Supporting**

This device can work with Private IP under NAT, and with a Router using dynamic IP in a WAN. This is very important for countries where few IP address are available.

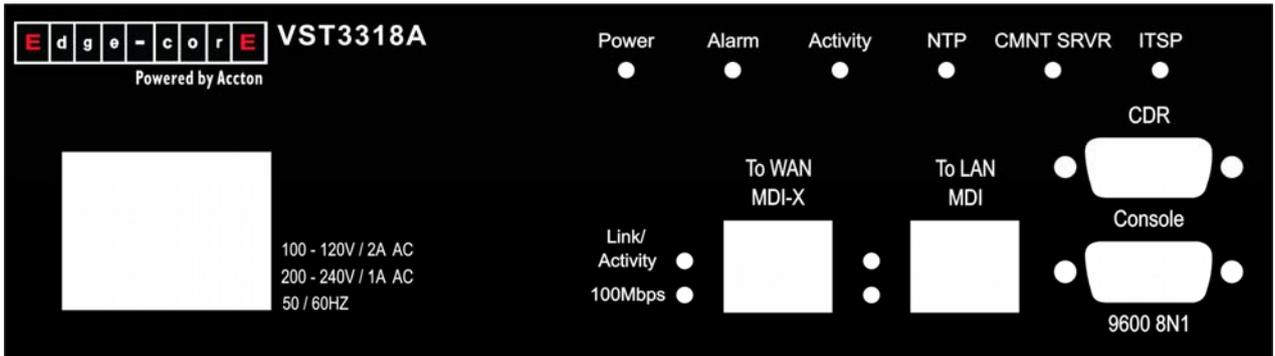
4. Appearance

This device can work together for stacking/networking. Below are the figures and LED indicators of the units.

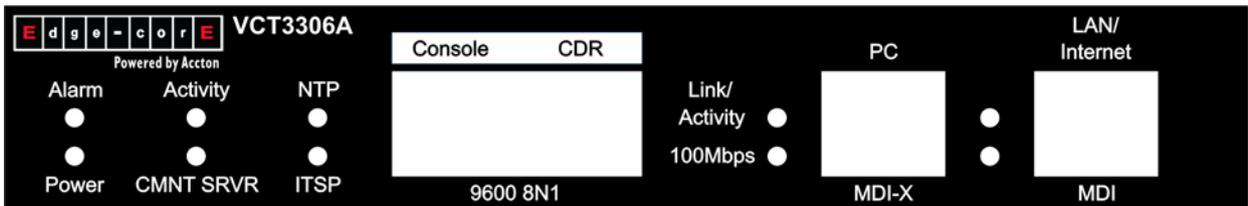
4.1 Panels

Front Panels

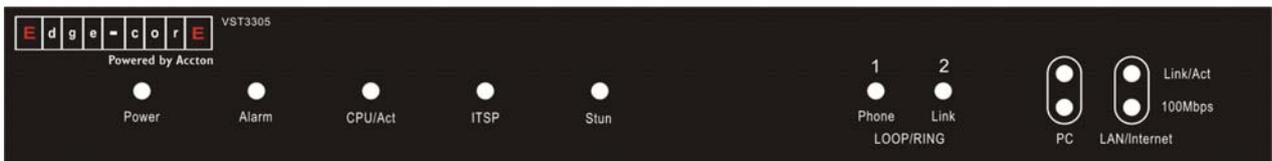
VST3318



VST3306 A/B/C/D

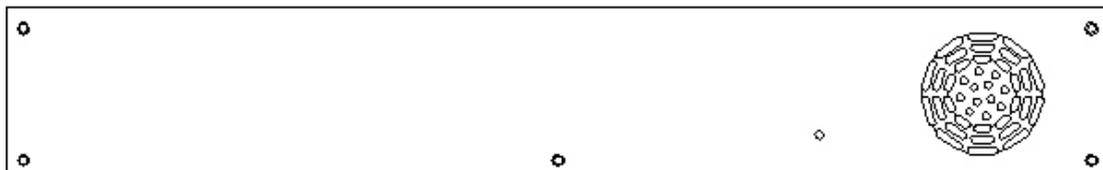


VST3305

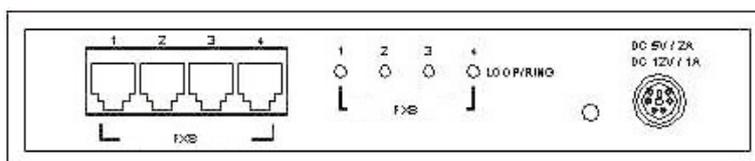


Rear Panel

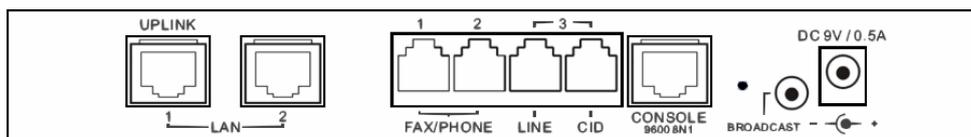
VST3318



VST3306A (FXS/FXO combinations vary depending on model)

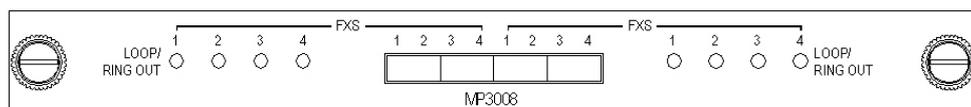


VST3305

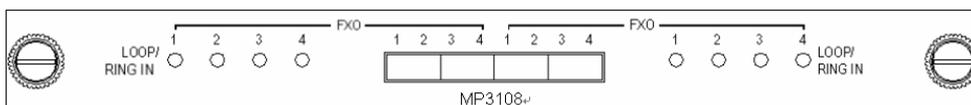


4.2 Module Appearances

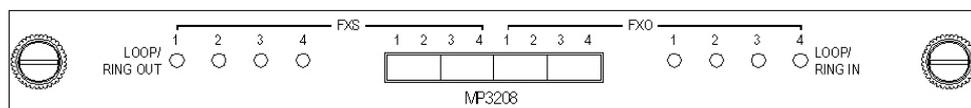
There are 3 modules (MP3008+, MP3108+, MP3208+) for the 8 and 16 port models. For the installation of these modules, please refer to quick installation guide.



MP3008+ (8FXS)



MP3108+ (8FXO)



MP3208+ (4FXS + 4FXO)

4.3 LED indicators

Type	Label	LED	Description
10/100 Ethernet	Link/ACT	ON	Network Linked Up
		FLASH	Sending/Receiving data packets
	100Mbps	ON	Transmission Rate is 100Mbps
		OFF	Transmission Rate is 10Mbps
Port Information	LOOP/ RING OUT (FXS)	ON	Phone set is off the Hook, loop current detected
		FLASH	Ring signal sending
	LOOP/ RING IN (FXO)	ON	Answered, loop current detected
		FLASH	Ringing detected
Device	PWR	ON	Power supply is normal

	ALARM	ON	Errors detected when auto HW diagnostics running
	Activity	ON	CPU is operating normal
		FLASH	CPU is Running
	NTP	ON	Able to access NTP Server (Time Server)
		FLASH	System is not connected or is not currently connected to NTP Server
		OFF	Not able to access NTP Server
	CMNT SRVR	ON	Registered to Community Server
		OFF	Not registered to Community Server
	ITSP	ON	All SIP Entities have registered to another Proxy Server (ITSP) successfully
		FLASH	SIP Entities have partially registered to another Proxy Server (ITSP) successfully
		OFF	Not registered to Proxy Server or SIP Entities failed to register to another SIP Proxy Server

4.4 Connection Ports

Terminal	Label	Description
Voice	FXS	For analog phone sets or FAX machines
	FXO	For PSTN or trunk connections
	BROADCAST (Only available on VST3305)	For an amplifier or speakers (earphone)
Network	To WAN (MDI-X)	RJ-45 MDI-X terminal, for WAN
	To LAN (MDI)	RJ-45 MDI terminal, for LAN
RS-232	CDR	For Call Detail Recording
	CONSOLE	For system console management

5. Planning of Extension

There are two kinds of extensions for this machine. One type is for phone sets that connect to this machine physically, and the other is for SIP extensions. A SIP extension does not have to physically connect to the VST3300. The SIP extension terminal device (SIP Phone, Softphone, wifi phone, etc) need only connect to the IP network and thus to the VST3300 device.

The extensions here in this chapter are the physical extensions of the PBX. SIP extensions, will be described in section 13 Create the Connection of SIP Device

5.1 Extension Numbers

The default physical extension numbers are 1011~1026 (physical ports available depend on model type)

5.1.1 How to Find the Default Extension No. from Web

Web Path: **2.Channel\2.1.Summary**

Pick the Channel you wish to configure that is of type FXS, note its extension number.

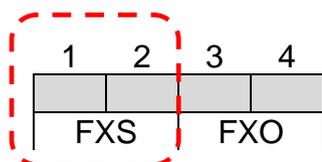
Ch	St	Suffix	Type	DND	Fwd	Barring Class	T.38	Statistics In/Out	PickUp/Group	Ext. No.	Voice Mail	Gain In/Out
1		11/OP*	FXS	-	-	1	-	0/0	✓ / 1	1011	-	0/0
2		12/OP	FXS	-	N*	0	-	0/0	✓ / 1	1012	-	0/0

1011 means its extension number is 1011, and so on.

To configure that channel, click its St (Status) icon to enter the sub-configuration menu.

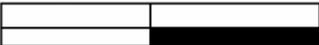
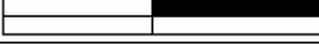
5.1.2 How to Find the Default Extension No. from Box

For 3 or 4 port models, check the machine label. Example below is of the 4 port model.



Only FXS ports can acts as extensions. 1 means the extension is 1011. 2 means the extension is 1012, and so on.

For the 16 port model, please check the matching table below.

Model	Group	Location	No. printed on panel			
			Extension number, only FXS are extensions			
VST3318	Group 1	Lower module (S1), 4 ports of left side 	1	2	3	4
			1011	1012	1013	1014
	Group 2	Lower module (S1), 4 ports of right side 	1	2	3	4
			1015	1016	1017	1018
	Group 3	Upper module (S2), 4 ports of left side 	1	2	3	4
			1019	1020	1021	1022
	Group 4	Upper module (S2), 4 ports of right side 	1	2	3	4
			1023	1024	1025	1026

5.2 Configuration of Pickup Group

Pickup groups can be configured for FXS ports and users in the same group can pick up each other's extensions.

5.2.1 Pickup Group for an Extension of This Machine

Web Path: **2.Channel\2.1.Summary**

Select the Channel with a type of FXS and click its St (Status) icon.

CH	St.	Suffix	Type	DND	Fwd	Barring Class	T.38	Statistics In/Out	PickUp/Group	Ext. No.	Voice Mail	Gain In/Out
1		11/OP*	FXS	-	-	0	-	0/0	✓ / 1	1011	-	0/0
2		12	FXS	-	B/C*	0	-	0/0	✓ / 1	1012	-	0/0

A window will pop up. Select a Group ID number from the My Pick Up Group drop-down menu. Select Yes in the Picked up By Others Control area to allow this function.

My Pick Up Group

Group ID:

Picked up By Others

Control:

PickUp/Group

✓ / 1

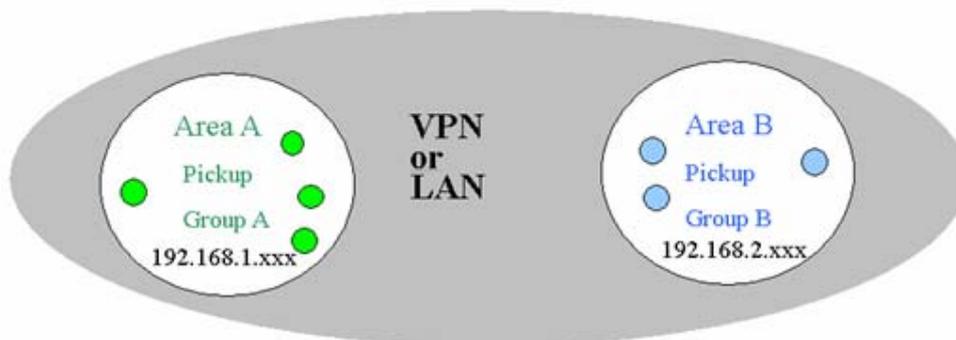
✓ / 1

Change is reflected in the summary table.

5.2.2 Call Pickup from another Machine

Call Pickup can be used for extensions belonging to the same pickup group, but different machines. However, please consider the following issues before configuration:

- The extension lines defined in a Call Pickup Group should be within hearing range of rings. This reduces missed calls if extensions are far apart or accidental call pickup of calls not in the same Pickup Group.
- The extension lines defined in the same Call Pickup Group must be under the same Subnet. Extensions configured in the same Pickup Group ID cannot pick up a call not in the same Subnet.
- Maximum 9 groups can be defined for Call Pickup.
- Using VPN applications, it is possible to have the same Subnet for two branch offices in different locations. Should this occur, separate them into different Subnets, such as 192.168.1.xxx and 192.168.2.xxx, to avoid accidental pickup from a different office.



- Note: extension lines can be configured individually if call pickup is set or not.

5.3 System Configuration for Extension Permission

There are two kinds of permission for extensions. One is to limit use of the auto routing function that makes PSTN call via this machine, called Outgoing Transit Permission. The other is to limit users making PSTN calls with certain prefix number when user had seized trunk (PSTN), called barring.

5.3.1 Permission of Outgoing Transit Calls

The feature Outgoing Transit Call enables remote devices to make or transfer a call to PSTN line via this machine.

Because charges for lines calling to PSTN are paid by the owner of this machine, only the calls from the remote gateway with permission are allowed to make outgoing calls through trunk of this machine. VST3300 Series defines three Route types to the different equipments. Each remote e will be assigned a Route Type for Outgoing Transit Call via this machine.

- Local : Only local calls are allowed
- Toll Call : Only local and toll calls are allowed

There is a "Permission List of Outgoing Transit" in this machine; the devices that have permission to make Outgoing Transit call are listed here. The list records information of:

- MAC Address
- Phone Number
- Route Type

Outgoing Transit permission of a machine itself is already entered by default when shipped. Users of the machine can make PSTN calls directly or make PSTN calls when PSTN trunk is seized. For one unit no additional Outgoing permission configuration is required. If another unit wishes to make a trunk (PSTN) call via this machine, see the following instructions.

1. Enter the MAC address and the telephone number of the remote device to be permitted. MAC Address can be found on the bottom label of the machine. The example below is 00-03-62-80-27-37

MODEL NAME:	
PRODUCT S/N:	TP350481B0423000006 8
MAC ADDRESS:	000362802737
IP PHONE NO:	+28-6-201-1006
Input Rating: 5V / 1.5A DC 12V / 0.5A DC	
檢磁 3902C635	
This unit complies with Part 15 & 68 of FCC Rules.	
FCC REG.NO.:US:VTLMF06BPF3504	
REN: 0.6B	
USOC JACK: RJ11C	
Operation is subject to following two conditions:	
(1) This device may not cause harmful interference	
(2) This device must accept any interference received including interference that may cause undesired operation.	
Region: Taiwan	
MADE IN TAIWAN	00460R05

MAC Address of this machine

2. Choose the Route Type from Local, Toll
3. Click Button

Web Path : **1.System\1.3.Outgoing Permission**

Outgoing Permission

Capacity: 64
Current Entry: 1

Selection List... Page 1 / 1 Show << >>

MAC Address	Phone Number	Attempts	Duration	Route Type
00-03-62-80-77-90+	886218419216	0000	0000	Toll

MAC Address Phone Number Route Type

Add/Modify: Local

Delete:

Delete All: No

Key In MAC:

List of devices already Permitted to use the device's trunks

Entry Blanks for steps 1 and 2

The " + " sign followed by the MAC Address in the **Outgoing Permission** List means that device is allowed to make Outgoing Transit calls and to seize remote trunks from the FXO ports of this machine.

5.3.2 Barring Class

The main purpose of Barring is to allow or disallow extensions to make PSTN trunk calls that start from certain digits (prefix). For the example, in Taiwan, 0 is used for long distance calls and 002 for international call.

For VST3300 Series, there are a maximum of six Barring Classes of any pre-defined barring rule that can apply to each individual extension. For example, the ability to dial a phone number can be defined in the Barring Class table. For each extension line, only one Barring Class can be selected.

Attention: Barring class only control phone calls that seize a trunk first, then dial

outgoing numbers later. Barring class can not control phone calls that use the auto routing function.

5.3.2.1 Create Barring Class

To configure Barring Classes follow the Web Path **1.System\1.7.Barring Class**. Barring Classes can be defined as "Accept" or "Deny". Each attribute can be defined in Barring List and Exception List. Only one Attribute can be defined for each Barring Class. The default values for the six classes are 0 and no data are defined.

Barring Class Configuration						
	Analog Trunk			SIP Trunk		
Class:	Class 1	▼	Select	Class 1	▼	Select
Name:						
Attribute:	N/A			N/A		
	Barring Class Setting			Barring Class Setting		
Class Attribute:	Deny	▼	New Delete	Deny	▼	New Delete
Class Name:			Add			Add
Barring:			Add Delete			Add Delete
Exception:			Add Delete			Add Delete
	Barring List			Barring List		
	Exception List			Exception List		

Group	Field	Description	Default Value
Class	Class	Selection of Barring Classes, Choose from 1 to 6	
	Name	Displays the name of barring class (Display Only)	Blank
	Attribute	Displays the attribute of Barring Class (Display Only)	N/A

Barring Class Setting	Class Attribute	<p>The attribute of the class, Deny or Accept. There are Barring Table and Except Table may be defined for each attribute. Only one attribute can apply to one barring class, can be Deny or Accept</p> <p>DENY: (Positive list, configure prefix number for phone call allowed only)</p> <p>All numbers are denied except the numbers listed in the Exception List. When DENY is selected, it is not necessary to define Barring List, because DENY is to reject all numbers.</p> <p>ACCEPT: (Negative list, configure prefix number that is not allowed for phone call)</p> <p>Accept all numbers except number in the Barring List. The number in the Exception List are exceptions.</p> <p>New: When Attribute has not yet to apply to barring class (shown N/A), click New to add the Attribute to selected Barring Class</p> <p>Delete: Delete the Attribute to selected Barring Class. Delete the attribute of certain barring class will deactivate that barring class.</p>	
	Class Name	<p>Defines a name for the selected Barring Class; any name less than 15 characters can be defined by the system Manager.</p> <p>Add: Add class name for Barring Class.</p>	
	Barring	<p>The phone number (less than 18 characters) that is limited to be dialed</p> <p>Add: Adds phone number to Baring List for selected Barring Class</p> <p>Delete: Deletes phone number from Barring List for selected Barring Class</p>	
	Exception	<p>The phone number (less than 18 characters) for exception</p> <p>Add: Adds phone number to Exception List for selected Barring Class</p> <p>Delete: Deletes phone number from Exception List for selected Barring Class</p>	
	Barring List	<p>Displays all numbers to be (Display Only) barred that is related to the attribute</p>	
	Exception List	<p>Displays all exception in Except (Display Only) Table that is related to the attribute</p>	

5.3.2.2 Steps to create the Barring Classes:

1. Select a Barring Class (1~6) from the field of "Class" under Barring Class Configuration, then click **Select** button.
2. Select an Attribute for the selected Barring Class, ACCEPT or DENY, and click **New** button.
3. Enter a name for Barring Class in the field of "Class Name" and click **Add** button.
4. Now you may define the details based on the attribute selected. Create the Baring List by clicking the **Add** button, and create the Exception List by clicking **Add** button.

After the Barring Classes are created, you may select a Barring Class for the IP-PBX extension.

(1) Example-1

If this machine is located in Shanghai, and only the calls to Beijing (Area Code = 010) and the calls to mobile phone (Area Code = 013) in China are allowed. For this Barring configuration, define the Area Code 010 and 013 in Exception List, other numbers will be denied.

Attribute	DENY
Barring Table	(No configuration is required when attribute is "Deny")
Exception Table	010 013

(2) Example-2

If this machine is located in Shanghai and only local calls to Shanghai are allowed, then all numbers starting with 0 (including toll calls and international calls) are not allowed except Beijing (Area Code = 010) and Shenzhen (Area Code = 0755).

Attribute	ACCEPT
Barring Table	0
Exception Table	010 0755

(3) Example-3

If this machine is located in St. Jose, USA, and only local calls in St. Jose are allowed, then Toll calls (starting with 1) and the international calls (starting with 011) are not allowed except calls to Beijing (011-86-10).

Attribute ACCEPT
Barring Table 011 1
Exception Table 0118610

(4) Example-4

If this machine is located in Tokyo, Japan, and only local calls to Tokyo are allowed. The toll call (starting with 0) and international call (starting with 001) are not allowed except calls to Shanghai (001-86-21).

Attribute ACCEPT
Barring Table 001 0
Exception Table 0018621

5.3.2.3 Modify the Attribute of Baring Classes

If the attribute of a Barring Class is defined, a new attribute (New) to the same Barring Class is not allowed to be defined. To modify to the attribute of a Barring Class has, delete the attribute (Delete), and then define a new one.

5.3.2.4 Apply Barring Class to Extension Line

Each extension line may select a Barring Class from the six classes when Barring Class is configured.

Web Path : **2.Channel\2.1.Summary**

Select the Channel with FXS Type and click its St (Status) icon.

CH	St.	Suffix	Type	DND	Fwd	Barring Class	T.38	Statistics In/Out	PickUp/Group	Ext. No.	Voice Mail	Gain In/Out
1		11/OP*	FXS	-	-	0	-	0/0	✓ /1	1011	-	0/0
2		12	FXS	-	B/C*	0	-	0/0	✓ /1	1012	-	0/0

Select a Barring Class, 1~6 (default is 0, meaning no barring) in the status table and then click **Apply**

Barring Class

ID:

5.4 Configuration to Join Operator Group

VST3300 series IP-PBX supports several types of Operator:

- DISA: is an Auto Attendant system that can answer incoming calls and play the greeting instructions. It can be disabled.
- Operator for this machine: Operator (personnel) takes incoming calls manually.
- Network Operator: Operator (personnel) at network site takes incoming calls manually.

When a call is coming from a trunk (i.e. FXO port) or from IP, DISA answers the call by default. If incoming call user presses 0, it will be picked up by the Operator of this machine. If company selects to use an Operator, at least one extension has to join the operator group (default value)

5.4.1 Operator on This Machine

Many extensions of this machine can join an operator group. If the specified operator is busy, the system will ring the idle extension at Operator Group from the first port (Channel 1) upward. Therefore, pay attention to the ports when wiring extension lines.

5.4.1.1 Configure Operator Group

When the default operator is busy, the system rings the next idle extension in operator group (from next extension upward to the end). By default, all physical extensions are part of the Operator group. For extensions to join the Operator Group, see below.

Web Path : **2.Channel\2.1.Summary**

Select the Channel with FXS Type and click its St (Status) icon.

CH	St.	Suffix	Type	DND	Fwd	Barring Class	T.38	Statistics In/Out	PickUp/Group	Ext. No.	Voice Mail	Gain In/Out
1		11/OP*	FXS	-	-	0	-	0/0	✓ / 1	1011	-	0/0
2		12	FXS	-	B/C*	0	-	0/0	✓ / 1	1012	-	0/0

In Status table, select to join group (Join Operator Group) or not. Default value is Yes, and then click the **Apply** button.

Join Operator Group

Yes/No: Yes

Note: If the extension with T.38 (for FAX only) enabled had already join to Operator Group, the system don't ring this extension when extension of operator is busy.

5.4.2 Network Operator

Network operator means the operator (personnel) is not configured at this machine and it is configured at other VST3300 Series.

It allows an operator to be located at another VST3300 Series via IP network. When incoming call dials the Operator, the system will search for the Operator in this machine. If no extensions are part of Operator, the system will assume that the Operator is defined on another machine. The system rings the Prefix/extension number of another machine defined in the **Network Operator Extension Number**. (The Prefix/extension number of should be already configured in Extension Table in advance.)

In the following example, the Operator is configured on the device with Prefix/Ext.: 81, which is a VST3300 Series with phone No. 886-2-8226-8881, as a Network Operator.

Configuration Steps:

1. Configure FXS ports to not be part of the Operator Group:
Web Path : **2.Channel\2.1.Summary**

Select the Channel with FXS Type and click its St (Status) icon

CH	St.	Suffix	Type	DND	Fwd	Barring Class	T.38	Statistics In/Out	PickUp/Group	Ext. No.	Voice Mail	Gain In/Out
1		11/OP*	FXS	-	-	0	-	0/0	✓ / 1	1011	-	0/0
2		12	FXS	-	B/C*	0	-	0/0	✓ / 1	1012	-	0/0

Select not to join Operator Group in Status Table, and then click **Apply** button. If there is no operator for this machine, configure each FXS port not to join Operator Group.

Join Operator Group

Yes/No:

2. Input the Prefix/Ext. of network operator into the Extension Number table of this machine. Follow Web Path: **1.System\1.2.Extension Number** Input the Prefix/Ext.=81, Phone_No=886282268881, Type=iPBX, and then click **Apply**.

	Prefix/Ext. (Max.6 digits)	Phone Number	Type
Add/Modify:	81	886282268881	iPBX

3. Configure the Prefix/Ext. of the network operator to the Network Operator Extension Number field.
Web Path : **1.System\1.2.Extension Number**

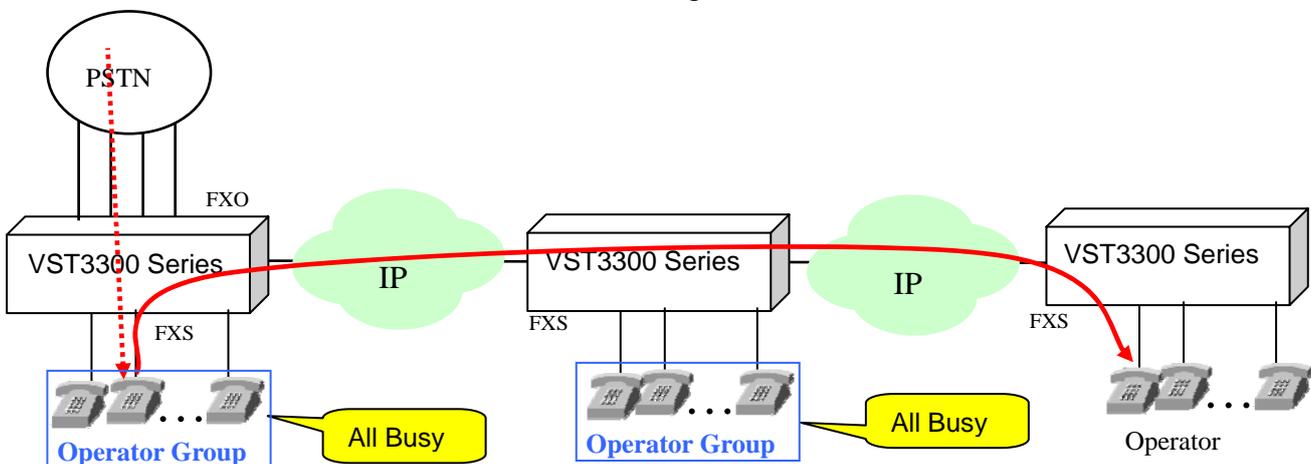
Network Operator		
Extension Number:	<input type="text" value="81"/>	
Extension Number List		
Capacity:	100	
Current Entry:	3	
Page <input type="text" value="1"/> / 1 <input type="button" value="Show"/> <input type="button" value="<<"/> <input type="button" value=">>"/>		
Prefix/Ext.	Phone Number	Type
10	0	iPBX
6600	886282263368	Phone
81	886282268881	iPBX

No matter what type (Phone/iPBX) of Prefix/Ext. is assigned to the network operator, connection is made without dialing an extra 0 or extension number.

5.4.2.1 Backup Network Operator (Cascade Device)

Even if an operator in this machine is configured, the other machine's network operator is automatically seized when all operators of the original machine are busy.

When a call is coming and the Operator Code is dialed, this machine will find the Operators of this machine for answering the call first. If the extensions in Operator Group of this machine are busy, and another machine is assigned as backup Operator, then system will find an available Operator from the second machine. If the second machine has also assigned a third machine as the backup Operator, and all Operators of the first and the second machine are all busy; then the system will find an available Operator from the third IP-PBX. Backup network operators can be cascaded up to 16 times. A maximum of 17 machines can be cascaded together.



Configuration:

Web path: **1.System\Telephony Tuning**, Input Phone Number (Web Path : **1.System\1.1.Basic Information**, My Phone Number) of other machine in Cascade Device field. This has to be full number, Country Code + Area Code + Phone Number, or a Community IP Phone number.

Cascade Device	
Phone Number:	<input type="text"/>

Attention: Make calls between the operator extensions in different machine in advance.

5.5 Changing Extension Numbers

The default extension number of physical extension (FXS port) is 1011~1026 (extension amount available depends on model). In 5.1 Extension Number, we know the rules of extensions. By these rule, you can specify your own extension number.

5.5.1 Change Prefix of Extension

The default physical extension number are 1011~1026, where the default prefix of extensions is "10"

Web Path: **1.System\1.2.Extension Number**

Prefix/Ext.	Phone Number	Type
10	0	iPBX

These values are changeable. Example: add a new Prefix/Ext.: 8. Then the extension numbers are changed to 811~826. To do this, see the diagram below.

	Prefix/Ext. (Max.6 digits)	Phone Number	Type
Add/Modify:	<input type="text" value="8"/>	<input type="text" value="0"/>	<input type="text" value="iPBX"/>

(For simplicity leave the Prefix/Ext.: 10, unless there is a numbering conflict.)

Note: Setting the Phone Number as 0 means this is phone number of the machine itself. It can sync with **My Phone Number** in Web Path: **1.System\1.1.Basic Information**

5.5.2 Configure Dedicate Extension Number

◆ Example 1

The Full number of **This Machine** is 886-2-82263397, configure two extensions as below.

From Web Path: **1.System\1.2.Extension Number**

Prefix/Ext.: 8888, Phone Number: 886-2-82263397-11, Type: Phone

Prefix/Ext.: 6666, Phone Number: 886-2-82263397-12, Type: Phone

When finished,
Dial 8888, the original default extension 1011 will ring.
Dial 6666, the original default extension 1012 will ring.

◆ Example 2

The full number of **Other Machine** is 886-2-82263380, configure two extensions as below.

From Web Path: **1.System\1.2.Extension Number**

Prefix/Ext.: 8801, Phone Number: 886-2-82263380-11, Type: Phone

Prefix/Ext.: 6601, Phone Number: 886-2-82263380-12, Type: Phone

When finished,
Dial 8801, the original default extension 1011 at other machine will ring.
Dial 6601, the original default extension 1012 at other machine will ring.

Attention: creating a prefix or changing extension numbers will be written into a .MEM file. This file can be backed up and/or modified in Windows Notepad. Such .MEM files can be uploaded into a VST300 Series device (requires a restart). This can be used for organization, management and to maintain dialing behavior.

6. Configuration of Trunk

6.1 Permission of Incoming Call Route to Trunk

VST3300 Series provides a feature that can transit a call to a trunk (FXO or SIP) port, then route to FXS port of another machine; or to PSTN via other machine. In other words, a transit calls is a PSTN-IP-PSTN call. This chapter will tell you how to configure the permission of incoming PSTN calls to transit to a PSTN call.

If the transit phone number is a Community IP Phone No. or FXS extension number of another machine, it does not have this type of limitation as it is a VoIP call with no extra charge. If the transit call is transit to PSTN (PSTN-IP-PSTN), then a PIN Number will limit the transit calls. Different PIN Numbers can have different permissions. See below.

- **Disable** : Call can not be transited to outside the IP network
- **Local** : Call can be transited to a local PSTN call
- **Toll** : Call can be transited to a local or toll PSTN call

- International : Call can be transit to a local, toll, or international call

This is relative to the "Region ID" of the device that is called to. See the example below.

Example: User in Taipei makes a call to a VST3300 device in Taipei, phone number is 82268888. This call is transited to PSTN in Shanghai via remote VST3300 device in Shanghai. A PIN Number for Transit Calls with privilege for international call, e.g. 2222 is required. Follow Web Path : **1.System\1.5.PIN Number** to make configurations.

	PIN Number	Class
Add/Modify:	2222	International

A PIN Number is required for the VST3300 Series to provide this function.

The steps to place the call are:

- (1) Make a PSTN call to Taipei's VST3300 device (82268888) from a public phone.
- (2) After the greeting, dial **2222#002862164452222# (002 is the International Access Code of Taiwan; 86 is the Country Code of PRC; 21 is the Area Code of Shanghai)

Attention:

1. Note that the user is trying to make a transit call from Taipei's VST3300 Series to a public line in Shanghai; therefore this user needs the PIN Number for Transit Call with privilege for international permission.
2. The remote device that provides routing to the PSTN needs to have configured the Outgoing Permission Table. Follow Web Path: **1.System\1.3.Outgoing Permission** to check Outgoing Permission. Refer to 9.1.4 Add Permission of Outgoing Transit to PSTN

6.2 Allow Incoming Calls Only (Incoming Only Trunk)

For office telephone systems, a trunk or trunks should be reserved for incoming calls only. Users should not be allowed to seize these trunks to make calls, thus protecting against incoming calls always receiving busy signals.

The Trunk Class of each Analog Trunk has default setting "0". All extension can seize all Analog Trunks by default. To enable this function, change the Trunk Class ID of FXO port(s) so incoming calls only are accepted.

1. Web Path : **2.Channel\2.1.Summary**

Select the FXO channel to be set for incoming call only and click its St (Status) icon

Ch	St	Access	Type	Error Count	Fwd	Trunk Class	T.38	Statistics In/Out	Last Number Dialed	Gain In/Out
3		9	PSTN	0	-	0	✓	0/0	-	0/0
4		9	PSTN	0	-	2	✓	0/0	-	0/0

2. Configure Trunk Class as "2":Incoming Only" for example, and then Click **Apply**

Trunk Class

ID:

3. If more than one trunks will be set as an incoming only FXO (PSTN) port, please select the other FXO port and repeat steps 1 and 2.

7. Configuration of Abbreviated Dial

The feature of Abbreviated Dial is to provide a simple and short dialing behavior for complex and long telephone numbers, eliminating the need to dial the full telephone number. There are 100 entries for Abbreviated Dial. The Abbreviated Dial Index allows every extension line to make a call by taking the phone off hook and dialing:

< "*"2" > + <Abbr. Dial Index (00 ~ 99) >

Attention: When making a call by Abbreviated Dial, restrictions are overridden if the Barring Table index is from 00-69, and will be restricted if the Barring table index is 70-79.

0 ~ 9, *, and # may be used to define the Abbr. Dial Index in Abbr. Dial Table. The number defined is the actual digits to be dialed out for making the phone call. For example, if you need to dial "9" before the telephone number " 0921888666 " can be made, then you should configure the telephone number in the Abbr. Dial Table as " 90921888666 ". Another example, if you would like to make an IP call **82263368#, then you can configure the telephone number as "***82263368#" in the Abbr. Dial.

Web Path: **1.System\1.6.Abr. Dial** A Web example is provided below.

	Index	Abbr. Dial Number
Add/Modify:	<input type="text" value="00"/>	<input type="text" value="**88628226339012#"/>

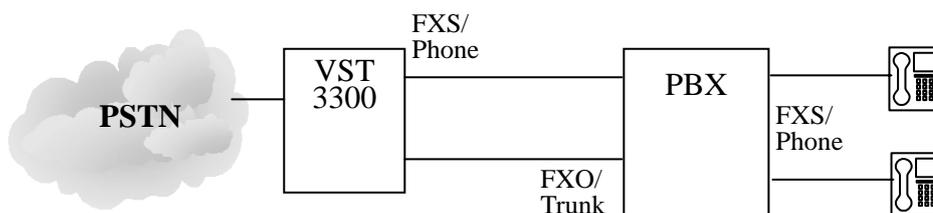
8. Connect a Traditional Analog PBX

8.1 Connect FXS to PBX Analog Trunk

There are 2 and 4 wires type phone cables available. Use 2 wire type cables with a RJ-11 jack.

- ◆ FXS or Phone ports of this machine can connect to phone sets or Trunk card/FXO ports of a tradition Analog PBX only.

Example Structure



8.1.1 Configuration on VST3300 Series

8.1.1.1 DISA Configurations (Auto Attendant)

By default, the machine will answer trunk (PSTN) calls automatically and play a greeting. If the DISA of a traditional analog PBX will be used, the DISA of VST3300 Series needs to be disabled. Incoming calls will bypass the VST3300 device go to the first FXS (Channel 1) and to the traditional analog PBX

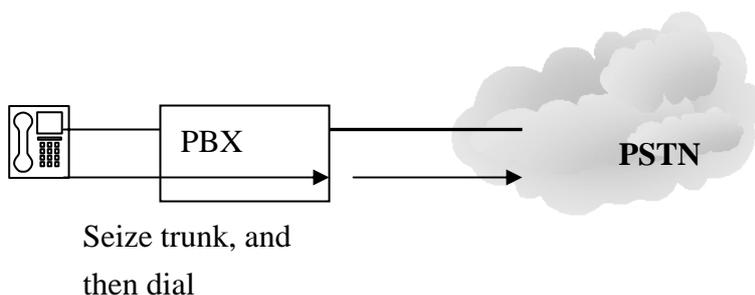
Web Path : **1.System\1.1.Basic Information**

Automated Attendant	
PSTN Call:	Disable
ITSP Call:	Enable
IP Call:	Enable

8.1.2 Analog PBX Configuration

8.1.2.1 Test DTMF sending on PBX trunk

Before the doing the above, first test to see if the trunk of the PBX is able to send the correct DTMF tone. Connect the PBX and phone set below to do this test.



◆ **Seize PBX Trunk to Send DTMF**

Test to see if the PBX extension is able to send */# DTMF when seizing a trunk. If it does not work, please contact your PBX service personnel to change the PBX's settings.

Testing Method

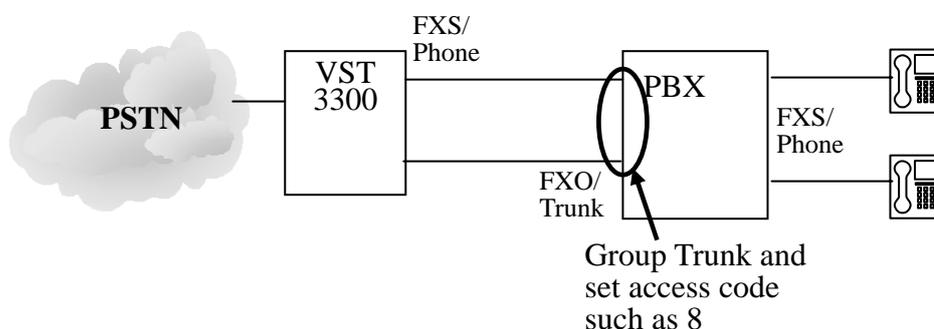
Seize the PBX Trunk to PSTN by extension first. If seized okay, you will hear a dial tone. Dial * or # to hear the tone reaction from the phone-set. If a dial tone stops immediately, i.e., goes silent, it means that the PBX is able to send the DTMF signal correctly.

8.1.2.2 Configuration of PBX Trunk Group and Tuning of Wire

Configure the trunk group on the line connection from the VST3300 Series device and also assign an access code of that PBX Trunk group

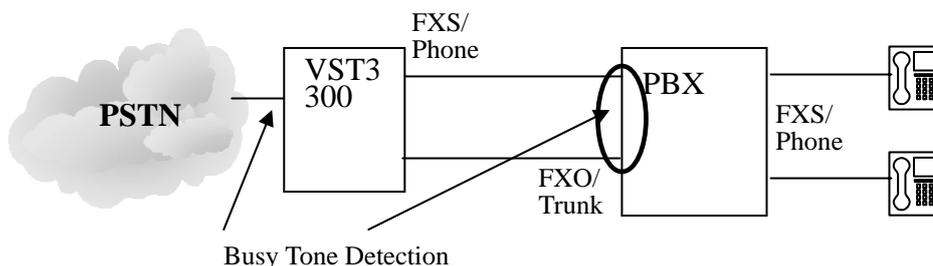
◆ **Connection of PBX trunk port**

Group PBX FXO ports that connect to two lines (or more lines, depends on ports available) of VST3300 Series. Specify trunk group access code, such as "8" to this trunk group. When "8" of phone set is dialed, dial tone form VST3300 Series is heard. User then can make IP call or seize trunk (PSTN) to make call.



8.1.2.3 Busy Tone Detection

Busy tone detection is for line termination when conversation is finished and it only applies to FXO Trunk only. For FXS port, it use loop current detection and there is no Busy Tone Detection issue.



◆ **Busy Tone Detection of VST3300 Series**

VST3300 Series had built-in busy tone detection system of PSTN for many countries. With a correct Region ID configured, the system can correctly detect a busy tone.

To see the current Region ID setting, please refer to

Web Path : **1.System\1.1.Basic Information**

Information	
Region ID:	0 (Taiwan)

To change Region ID, use Console or Telnet. Do a Factory Reset when it is done. Please refer to Quick Installation Guide, section 5.2 Configuration of Telecom Region ID and also refer to its Region ID table.

If the location of this machine is not listed in the table, you can refer to

<http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf>, “**Various Tones Used in National Networks**” to find a busy tone specification of the location (country) and configure the Region ID of the device so it can be used in an area with similar busy tone specifications.

For example, configure Macao’s Region ID to France or Germany, since their busy tone is similar to Macao’s.

◆ **Busy Tone Detection of PBX**

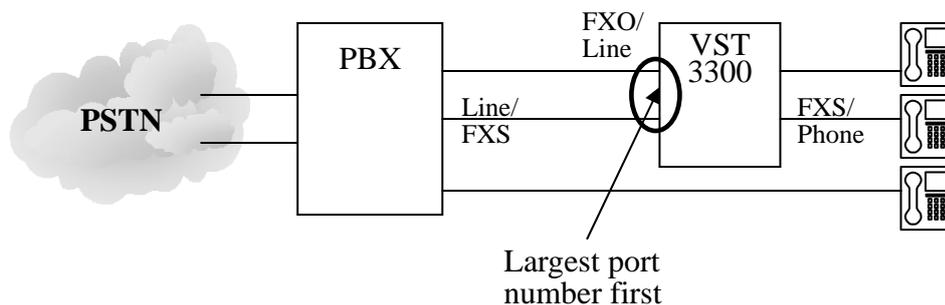
The VST3300 Series sends a busy tone determined on the specification from the selected Region ID, so configure the busy tone detection of PBX to be the same country as VST3300 Series.

8.2 Connect FXO to PBX Analog Line

There are 2 and 4 wire type phone cable available. The 2 wire type cable is OK, It uses a RJ-11 jack. Note:

- ◆ FXO or Line ports of this device can connect to PSTN from telecom company or Line card/FXS port of tradition Analog PBX only.

Example Structure



Note: By default, trunk (FXO) ports with the highest number will be seized by extension first. Connect the FXO/Line ports of VST3300 Series from the highest number downward. For the example, in the 16 port model, connect it by the sequence 16→15→14→13

Attention: Misconnecting the FXS/phone ports of this device to a PSTN trunk for a Telecom Company of Line Card of an analog PBX will cause damage to the device, Telecom company and analog PBX.

8.2.1 VST3300 Series Configurations

8.2.1.1 Configuration of PBX Trunk Access Code

By default, incoming trunk (PSTN) calls are answered by DISA that will play a greeting. If FXO port(s) of VST3300 Series are connected to extension lines of a PBX, the PBX Co Line Access Code must be defined. When the VST3300 Series makes an Outgoing transit call, users need not know that the VST3300 device is connected behind a PBX and he can dial a PSTN phone number directly. An access code should be configured here to seize the real PSTN Trunk and then make a PSTN call.

Web Path: **4.Dialing Plan**



In the example above, "9" means PSTN trunk access code of analog PBX and "P" means pause 1 second. If the speed that PBX seize trunk is a little slower, "P" can be repeated, such as 9PP

Note: If the FXO port is seized by a remote extension, the dial tone heard by the extension is sent from the PBX. This procedure will not pass the PBX to get a dial tone from a Telecom Company.

8.2.1.2 Gain Control

Calls to the PSTN or an extension of another VST3300, must go through PBX and the VST3300. So adjust Line/Trunk volume (Gain) of PBX and VST3300 Series as required if necessary. The tuning of volume (Gain) is important for DTMF sending.

In normal conditions, tune volume (Gain) is not required for both the PBX and VST3300 when connected together. If the volume (Gain) is too large or too small, simply adjust it.

◆ Tune volume (Gain) of VST3300 Series

Web Path : **2.Channel\2.1.Summary**

Select the PSTN (FXO) Channel that is connected to PBX and click its St (Status) icon.

Ch	St	Access	Type	Error Count	Fwd	Trunk Class	T.38	Statistics In/Out	Last Number Dialed	Gain In/Out
3		9	PSTN	0	-	0	✓	0/0	-	0/0
4		9	PSTN	0	-	0	✓	0/0	-	0/0

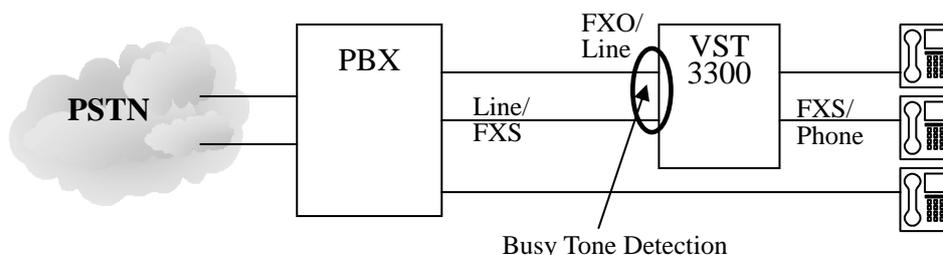
In the Voice section of the Status table

Voice	
Input Gain:	<input type="text" value="0"/> dB
Output Gain:	<input type="text" value="0"/> dB
Silence Suppression:	<input type="text" value="Disable"/>

Input Gain and Output Gain can be adjusted individually.

8.2.1.3 Busy Tone Detection

Busy tone detection is for line termination when conversation is finished. This only applies to FXO Trunks only. For FXS ports, loop current detection is used and there is no Busy Tone Detection issue.



◆ Busy Tone Detection of VST3300 Series

The busy tone that is sent from PBX should be the same as local telecom specifications.

If the PBX specification is the same as local telecom's, configure the Region ID, see the Quick Installation Guide Section 5.2. The VST3300 can then correctly detect busy tone. However, the PBX busy tone specification may be defined by its manufacture individually, and not by the specification of a country. Special busy tone specifications can be configured by yourself.

Web Path : **1.System\1.8 Telephony Tuning**

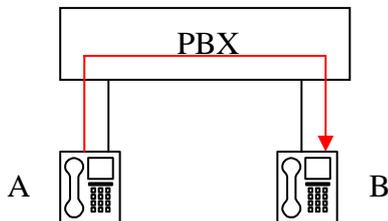
Busy Tone spec.			
Frequency (300~3000Hz):	f1:	<input type="text" value="480"/>	f2: <input type="text" value="620"/>
Cadence (100~5000ms):	on:	<input type="text" value="500"/>	off: <input type="text" value="500"/>

Tune it to have the same busy tone specifications of the PBX

8.2.2 Busy Tone Detection of PBX

8.2.2.1 Test DTMF Sending Between PBX Extension

Before the construction of the example above, check DTMF sending (including */#) between PBX extensions. Connect the PBX and phone set shown as below first.

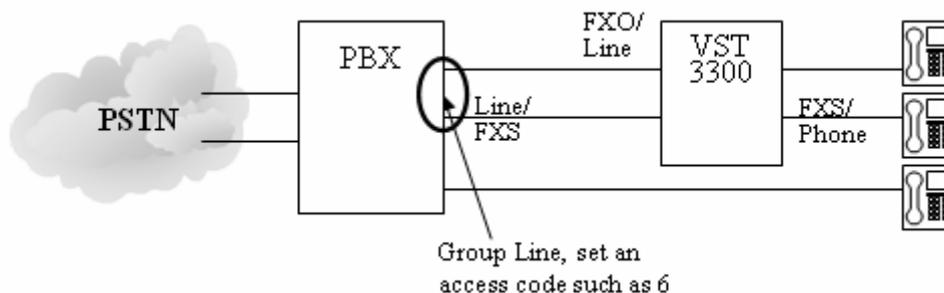


Testing Method

A that is under PBX makes call to B and B answers the call. Then A dial */# from phone set. If B can hear the DTMF Du tone which A dialed, then it means the PBX can send the DTMF correctly. If not correct, then some function will not be workable.

8.2.2.2 Configuration of PBX Line Group and Wiring

For a Group of PBX FXS ports that are connected to the VST3300, specify a line group access code, such as "6". When digit "6" from an extension or PSTN is dialed, a DISA greeting from the VST3300 or a second dial tone is heard. IP Calls can be dialed now.



9. Working with Other VST3300 Products

9.1 Stacking/Networking Configuration

Here is the definition of Stacking/Networking:

- ◆ Stacking: Means more than one machines stack together in one single location. Stacking can increase available port (Channel)
- ◆ Networking: Means devices are used in different place and connected with Internet. Networking can expand useable location

VST3300 Series uses fully distributed architecture and offers a variety of devices (2/4/8/16 models) for stacking/networking and expansion purposes.

Although intended for stacking/network environments, when starting, finish the installation and configuration of a single device first. Refer to Quick Installation Guide. When finished, follow the section below to configure for stacking/networking.

9.1.1 Configure Different Port Numbers When Stacking

For devices using public IP (Internet), simultaneous port use (2000, 5060, 10000) is okay without adjustment to port number. However, VST3300 devices stacked together under NAT (Router, Firewall) require different Signaling Port Numbers.

On the NAT (Router, Firewall) side, it is required to open Server Port for NAT penetration. Refer to Quick Installation Guide section 6.2 Open Server Port to Penetrate NAT

On the VST3300 Series side:

The default port number is 2000. If another device is using 2000, then this machine has to change.

See the following configuration example:

	Call Control	SIP Message	RTP Base
VST3300 1	2000	5060	10000
VST3300 2	2010	5070	11000
VST3300 3	2020	5080	12000

And so on...

Web Path : **1.System\1.1.Basic Information**

UDP Port Setting	
PBX Signaling:	2000
SIP Signaling:	5060
Voice:	10000

1. Follow Web Path: **1.System\1.1.Basic Information**
2. Input 2010 (or other value) in PBX Signaling field of UDP Port Setting
3. Click **【Apply】**
4. Repeat steps 1 - 3 as necessary for other stacking/networking VST3300 devices.
5. Warm Restart is required to apply the changes.

9.1.2 Add Prefix / Extension Number

When machines are stacked/networked, information of other VST3300 Series needs to be inputted into Prefix/Ext. The extension numbers of this machine and other machines or the Prefix Extension Numbers of this machine or other machines should be entered/defined into Extension Number table. The device already has default data for this machine when shipped.

Web Path: **1.System \1.2.Extension Number**

Example: Prefix/Ext. **【10】**、Phone Number **【My Phone No. Full No.】**、Type **【iPBX】**

Prefix/Ext.	Phone Number	Type
10	0	iPBX

Means the extension number of this machine is 10xx, the physical extension numbers that connect to phone set are 1011~1026 (amount of extension numbers depends on models and modules)

For machines that are used in stacking/networking, only one device can retain Prefix/Ext. 10. Others should be different to avoid conflict. **All devices have same default values saved upon shipping.**

◆ The example of stacking:

Prefix/Ext. **【10】**、Phone Number **【886282263368】**、Type **【iPBX】** (Ext. 10xx)

Prefix/Ext. **【20】**、Phone Number **【886282263369】**、Type **【iPBX】** (Ext. 20xx)

Prefix/Ext. **【30】**、Phone Number **【886282263370】**、Type **【iPBX】** (Ext. 30xx)

...and so on

◆ The configuration below is example of networking:

Input Data	Description
Prefix/Ext. 【10】、Phone Number 【886282263368】、Type 【iPBX】	VST3318 at Taipei, Extensions are 10xx
Prefix/Ext. 【20】、Phone Number 【862177453276】、Type 【iPBX】	VST3318 at Shanghai, Extensions are 20xx
Prefix/Ext. 【30】、Phone Number 【85299673356】、Type 【iPBX】	VST3318 at Hong Kong, Extensions are 30xx
...and so on	...and so on

Web Path : **1.System \1.2.Extension Number**

1. Click Web Path on left side 【1.System \1.2.Extension Number】。
2. Input Prefix of extension of other machine into 【Prefix/Ext.】 field, such as 20. Then extensions of other machine are 20xx
3. Input the phone number that comes from 【My Phone Number】 (Web Path : 1.System\1.1.Basic Information) of other machine, into the 【Phone Number】 field. This phone number has to be full number (Country Code+ Area Code + Phone Number)
4. Select iPBX in 【Type】 field
5. Click 【Apply】。
6. Repeat the steps above if there are more machines to be stacked/networked

9.1.3 Input Information of Other Machine into Phonebook

The machines for stacking/networking have to input others' information into the Phonebook of own machine. Assumes there is 3 sets of VST3300 Series devices.

Machine ID	Info of other machine that should added into Phonebook of this machine
1 st set	2 nd set, 3 rd set
2 nd set	1 st set, 3 rd set
3 rd set	1 st set, 2 nd set
	...and so on

◆ The example of **stacking**:

Below is information for the 3 sets. In most of case, they use fixed Private IP of LAN when they are stacking together.

1st set : Phone Number 【886282263368】、IP/Port 【10.13.6.107/2000】

2nd set : Phone Number 【886282263369】、IP/Port 【10.13.6.15/2001】

3rd set : Phone Number 【886282263370】、IP/Port 【10.13.6.34/2002】

...and so on

Input the others' information into each machine according to the table above.

◆ The example of **networking**:

Below is information for 3 sets. In most of case, they use fixed Public IP of Internet when they are networking together.

1st set : Phone Number 【886282263368】、IP/Port 【61.222.217.39/2000】 (at Taipei)

2nd set : Phone Number 【862177453276】、IP/Port 【61.220.219.84/2000】 (at Shanghai)

3rd set : Phone Number 【85299673356】、IP/Port 【203.204.89.38/2000】 (at Hong Kong)

...and so on

Input the others' information into each machine according to the table above.

According to the service of different ISP, PPPoE can be used on Router. If the Router that connect to this machine use dynamic IP (PPPoE), then input IP/Port information 0.0.0.0/0 of other machine into the Phonebook of this machine and do auto-learning of Phone Book procedure. Please refer to next section 9.1.3.1 Sync of Phonebook IP or contact with distributor.

Example:

1st set : Phone Number 【886282263368】、use dynamic IP (PPPoE) (at Taipei)

2nd set : Phone Number 【862177453276】、use dynamic IP (PPPoE) (at Shanghai)

3rd set : Phone Number 【85299673356】、IP/Port 【203.204.89.38/2000】 (at Hong Kong)

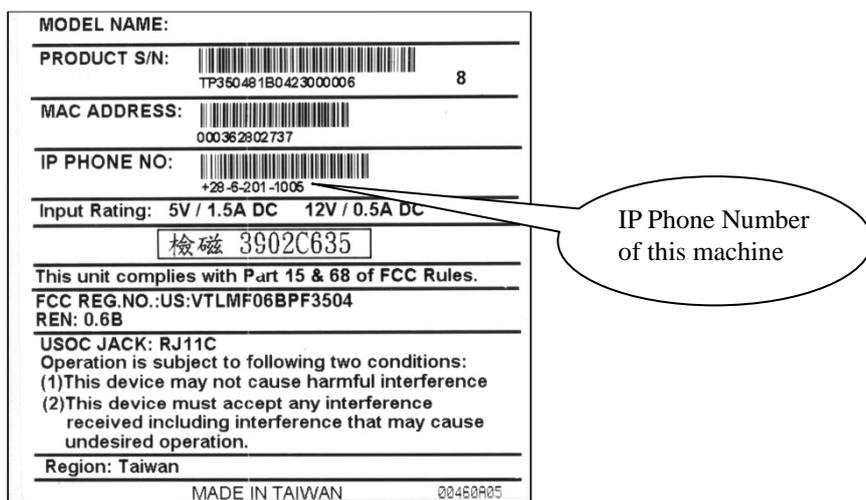
Machine ID	Phone Book info that need to learn from	Information of other machine that need to add into Phonebook
1 st set	2 nd set	Input Phone Number 【862177453276】、IP/Port 【0.0.0.0/0】 in PHONEBOOK field, and do auto-learning: 1 st set dial to 2 nd set, then 2 nd set dial to 1 st set
	3 rd set	Input Phone Number 【85299673356】、IP/Port 【203.204.89.38/2000】 in PHONEBOOK field, and do auto-learning: 1 st set dial to 3 rd set
2 nd set	1 st set	Input Phone Number 【886282263368】、IP/Port 【0.0.0.0/0】 in PHONEBOOK field, and do auto-learning: 2 nd set dial to 1 st set, then 1 st set dial to 2 nd set
	3 rd set	Input Phone Number 【85299673356】、IP/Port 【203.204.89.38/2000】 in PHONEBOOK field, and do auto-learning: 2 nd set dial to 3 rd set
3 rd set	1 st set	Input Phone Number 【886282263369】、IP/Port 【0.0.0.0/0】 in PHONEBOOK field, and do auto-learning: 3 rd set dial to 1 st set, then 1 st set dial to 3 rd set
	2 nd set	Input Phone Number 【862177453276】、IP/Port 【0.0.0.0/0】 in PHONEBOOK field, and do auto-learning: 3 rd set dial to 2 nd set

Web Path : **6.Phonebook**

1. Click Web Path on left side 【6.Phonebook\】。
2. Input phone number that come from 【My Phone Number】 (Web Path : 1.System\1.1.Basic Information) of other machine, into the 【Phone Number】 field. This phone number has to be full number (Country Code+ Area Code + Phone Number)
3. Input IP and Port Number of other machine to 【IP Address】 and 【Port】 fields
4. Click 【Apply】。
5. Repeat the steps above if there are more machines have to be stacked/networked

9.1.3.1 Sync of Phonebook IP

The IP address of Phonebook can be learned by making Community IP Phone call. (If fix IP is not used, do learning again every time IP address is changed). The Community call IP number can be found on the bottom label of the device. Be sure that the status is OK before making call. (The LED of NTP and CMNT SRVR is ON)



Dialing Method:

The number structure and dialing method of Community IP calls are the same as traditional telecom calls.

Find the Community IP Phone number to be dialed and compare it with your own number:

- If the country code and area code are the same, you only need to dial **+office code+ subscriber number+#
- If the country code is the same, but area code is different, you need to dial **+long distance access code+area code +office code+ subscriber number+#
- If the country code is different, you need to dial **+International access code+country code+area code +office code+ subscriber number+#

Example: (Assume Long Distance access code is "0" and International Access code is 002.)

Calling side IP Phone number	Called side IP Phone number	Dialing Method
28-6-201-1000	28-6-201-7799	**201 7799 #
28-6-201-1000	28-7-200-6600	**07 200 6600 #
28-6-201-1000	18-7-8226-3368	**002 187 8226 3368 #

If the number you dial is not busy (a second dial tone or instruction greeting is heard), and then auto-learning of phone book is done.

9.1.4 Add Permission of Outgoing Transit to PSTN

The same as a traditional analog PBX, extensions can seize a trunk to make PSTN calls if this machine or other VST3300 Series is connected to PSTN trunk. By default, the extension of this machine can seize the PSTN trunk of this machine to make PSTN calls. If you input the information of the other machine into this machine, then the other machine can make PSTN calls via this machine. PSTN calls are not free, so configure permission to avoid extra charges. Do not input the information of other machine into this machine if you don't want that device to make call via this machine.

In the previous section, after the other machine has inputted the information of this machine into its **Phone Book**. This machine can list info of other machine in **Selection List**. Then this machine can tick the machine that can make PSTN call via this machine.

Web Path : 1.System\1.3.Outgoing Permission

Apply Cancel

Outgoing Permission

Capacity: 64

Select other machines that will make trunk calls via this machine

MAC Address, Phone Number and Route Type of the machine itself (default)

Selection List...

MAC Address	Phone Number	Attempts	Duration	Route Type
00-03-62-80-77-90+	886218419216	0000	0000	Toll

Page 1 / 1 Show << >>

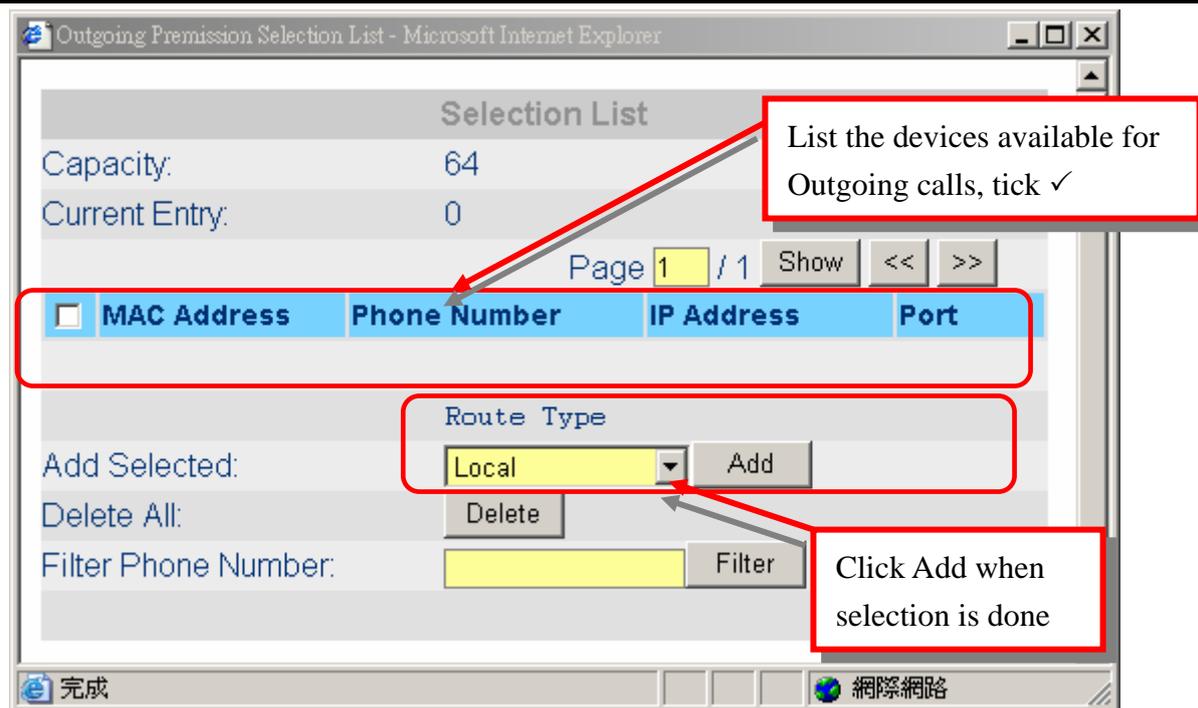
Add/Modify: Local

Delete:

Delete All:

Key In MAC: Search

Selection List



Example of Configuration

◆ The example of stacking:

MAC Address【00-03-62-80-25-37】、Phone Number【886282263368】、Route Type【Toll】
 MAC Address【00-03-62-80-25-48】、Phone Number【886282263369】、Route Type【Toll】
 MAC Address【00-03-62-80-25-56】、Phone Number【886282263370】、Route Type【Toll】

◆ The example of networking:

VST3318 at Taipei and Shanghai:

MAC Address【00-03-62-80-25-37】、Phone Number【886282263368】、Route Type【Toll】
 MAC Address【00-03-62-80-25-48】、Phone Number【862177453276】、Route Type【Toll】
 MAC Address【00-03-62-80-25-56】、Phone Number【85299673356】、Route Type【Toll】

VST3318 at Hong Kong:

MAC Address【00-03-62-80-25-37】、Phone Number【886282263368】、Route Type【Local】
 MAC Address【00-03-62-80-25-48】、Phone Number【862177453276】、Route Type【Local】
 MAC Address【00-03-62-80-25-56】、Phone Number【85299673356】、Route Type【Local】

If you don't want to use Selection List, you can also input it manually

Web Path : **1.System\1.3.Outgoing Permission**

Outgoing Permission

Capacity: 64
Current Entry: 1

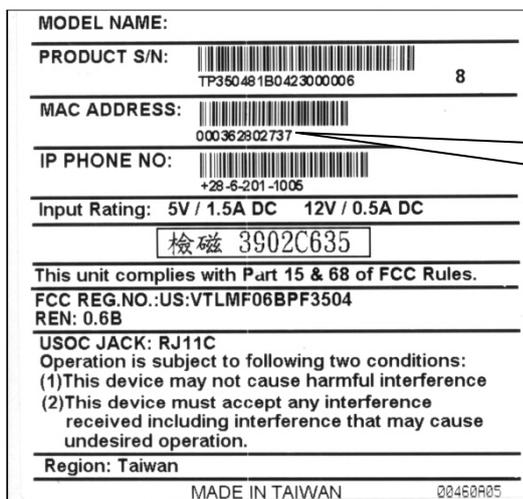
Selection List... Page 1 / 1 Show << >>

MAC Address	Phone Number	Attempts	Duration	Route Type
00-03-62-80-77-90+	886218419216	0000	0000	Toll

	MAC Address	Phone Number	Route Type
Add/Modify:	<input type="text"/>	<input type="text"/>	Local
Delete:	<input type="text"/>		
Delete All:	No		
Key In MAC:	<input type="text"/>	<input type="button" value="Search"/>	

Input MAC Address, Phone Number and Route Type manually.

1. Input MAC address of other device. The MAC can be found on the bottom label of the device. It is 00-03-62-80-27-37 in the figure below.

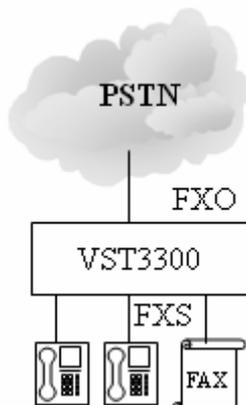


MAC Address of this machine

2. Input the phone number that came from **【 My Phone Number 】** (Web Path : 1.System\1.1.Basic Information) of other machine, into the**【 Phone Number 】**field. This phone number has to be full number (Country Code+ Area Code + Phone Number), such as 886282263368
3. Select Route Type as Toll (For areas like Hong Kong, with no long distance calls, select Local). Toll means local and long distance calls are usable. Local means only local calls are usable.
4. Click **【 Apply 】**
5. Repeat the steps above if there are more machines to be stacked/networked.

10. Connecting a FAX Machine

There is no special connection for a Fax machine. Connect a physical extension port (FXS) to a FAX machine directly. FXS ports are originally intended for analog phone sets



Each physical FXO port has the ability to receive FAX with no configuration required.

Each physical FXS port can be configured to have T.38 FAX function. Ports that with the FAX function not activated can not receive or send a FAX.

Note: If the FAX function is activated for a certain FXS extension, that extension can not be pick up by other extension.

Web Path : **2.Channel\2.1.Summary**

Select channel of Type FXS.

Analog Channel												
Ch	St	Suffix	Type	DND	Fwd	Barring Class	T.38	Statistics In/Out	PickUp/Group	Ext. No.	Voice Mail	Gain In/Out
1		11/OP*	FXS	-	-	0	-	0/0	✓ / 1	1011	-	0/0
2		12/OP	FXS	-	-	0	-	0/0	✓ / 1	1012	-	0/0
Ch	St	Access	Type	Error Count	Fwd	Trunk Class	T.38	Statistics In/Out	Last Number Dialed		Gain In/Out	
3		9	PSTN	0	-	0	✓	0/0	-		0/0	
4		9	PSTN	0	-	0	✓	0/0	-		0/0	

Click St (Status) icon. In Status table, select FAX in Connect Device Type and click **Apply**. VST3300 Series uses T.38 protocol to send a FAX in an IP network.

Connect Device Type

Type: Fax

The T.38 FAX status of each port can be seen in Summary table of all ports.

Web Path : **2.Channel\2.1.Summary**

Analog Channel													
Ch	St	Suffix	Type	DND	Fwd	Barring Class	T.38	Statistics In/Out	PickUp/Group	Ext. No.	Voice Mail	Gain In/Out	
1		11/OP*	FXS	-	-	0	-	0/0	✓ /1	1011	-	0/0	
2		12/OP	FXS	-	-	0	-	0/0	✓ /1	1012	-	0/0	
3		13	FXS	-	-	0	✓	0/0	✓ /1	1013	-	0/0	
4		14	FXS	-	-	0	-	0/0	✓ /1	1014	-	0/0	

11. DISA System Greeting (Auto Attendant)

The default greeting of the VST3300 Series is English. For other languages, recording of greetings can be done by yourself.

No special tools are required and any extension line can record the messages. There are in total, 8 greetings with one minute maximum for each.

Example:

Type of Greetings	Description of Greetings	Example of Messages
Greeting (1)	The Greetings for office hour	Good day, this is XX XXX, please dial an extension number or 0 for Operator
Greeting (2)	The message when line is busy	Line is busy, please dial another extension number or 0 for operator
Greeting (3)	The message when the number is wrong or can not be recognized	The number you dialed can not be recognized, please dial again
Greeting (4)	The message for waiting, the call is transferring	Thank you, please wait a moment
Greeting (5)	The greetings for company off duty or holiday	This is off duty time, please dial the extension number directly or call again during office hours
Greeting (6)	The message for no answer	There is no answer, please dial another extension number or 0 for Operator
Greeting (7)	The message for unable to answer the call, may be network problem or line problem	unable to answer, please dial another extension number or 0 for operator
Greeting (8)	When another IP-PBX activates Consult Transfer. The call side of this machine will hear this greeting section	Default is music. Change is not required. However, if desired it can record as: " Please wait a moment, the extension you dialed is making a consult transfer "

If the extension line is assigned as Operator, this line can activate the greetings for office hours and the greetings for off duty time.

- (1) To activate the greetings for office hour, take the phone set off hook, dial *0, then 071#
- (2) To activate the greetings for off duty time, take the phone set off hook, dial *0, then 070#

If the extension line is not assigned as Operator, this line should enter management mode to activate the greetings for office hours and off duty time.

- (1) To activate the greetings for office hour, take the phone set off hook, dial *0 , dial 09 9999# to enter the management mode, then dial 071#
- (2) To activate the greetings for off duty hour, take the phone set off hook, dial *0 , dial 09 9999# to enter the management mode, then dial 070#

11.1 Record System Greeting

- (1) Entering the Management Mode

Take the phone set off hook, when you hear the dial tone, dial *0, → then 09 9999# to enter the management mode → you will hear the tone of “DuDu.....”

- (2) Recording the 1st section

Dial 99 1 → * → start to record → # (end the record)

- (3) Storing the 1st section

Dial 9# → hear the tone of “DuDu...” → #

- (4) Recording the 2nd section

Dial 99 2 → * → start to record → # (end the record)

- (5) Storing the 2nd section

Dial 9# → hear the tone of “DuDu...” → #

- (6) Recording the 3rd section

Dial 99 3 → * → start to record → # (end the record)

- (7) Storing the 3rd section

Dial 9# → hear the tone of “DuDu...” → #

(8) Recording the 4th section

Dial 99 4 → * → start to record → # (end the record)

(9) Storing the 4th section

Dial 9# → hear the tone of “DuDu...” → #

(10) Recording the 5th section

Dial 99 5 → * → start to record → # (end the record)

(11) Storing the 5th section

Dial 9# → hear the tone of “DuDu...” → #

(12) Recording the 6th section

Dial 99 6 → * → start to record → # (end the record)

(13) Storing the 6th section

Dial 9# → hear the tone of “DuDu...” → #

(14) Recording the 7th section

Dial 99 7 → * → start to record → # (end the record)

(15) Storing the 7th section

Dial 9# → hear the tone of “DuDu...” → #

(16) Recording the 8th section

Dial 99 8 → * → start to record → # (end the record)

(17) Storing the 8th section

Dial 9# → hear the tone of “DuDu...” → #

Attention: Don't forget to dial “#” to end the last recording, then start the next section.

11.2 Listening to the Recorded Greetings

(1) Enter management mode

Take the phone set off hook, when you hear the dial tone, dial *0, → then 09 9999# to enter the management mode → you will hear the tone of “DuDu.....”

- (2) Listening the 1st message : Dial 961 → To stop, just dial #
- (3) Listening the 2nd message : Dial 962 → To stop, just dial #
- (4) Listening the 3rd message : Dial 963 → To stop, just dial #
- (5) Listening the 4th message : Dial 964 → To stop, just dial #
- (6) Listening the 5th message : Dial 965 → To stop, just dial #
- (7) Listening the 6th message : Dial 966 → To stop, just dial #
- (8) Listening the 7th message : Dial 967 → To stop, just dial #
- (9) Listening the 8th message : Dial 968 → To stop, just dial #

If you have more VST3300 devices or related products that use system greeting that don't have FXS port, you can download the recorded greeting files. These files are .GT1~GT8 and can be uploaded by WEB or FTP. No additional recording is required.

To download/upload files to a PC or VST3300, refer to chapter 17 Firmware Update and Backup

12. Work with Voice Mail Product

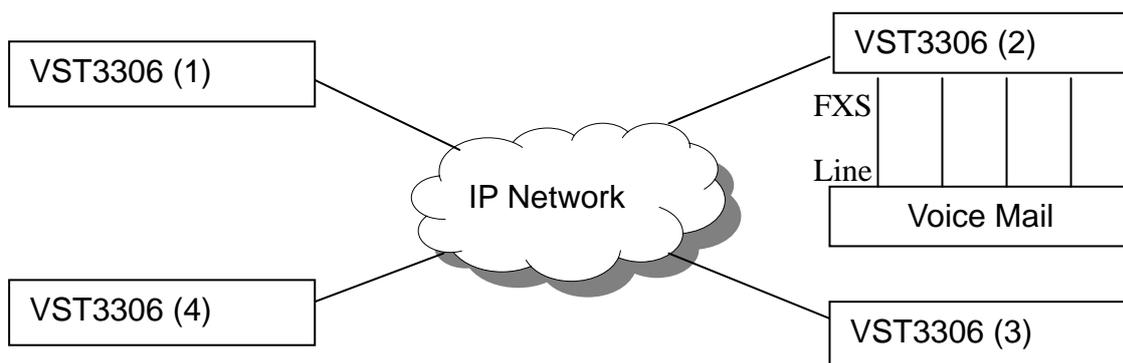
VST3300 Series can work with other brand's analog Voice Mail Product (Voice Mail will be abbreviated as "VM"). The function of VM includes Voice Mail Box, Retrieve (Listen) Voice Mail. After proper configuration, VST3300 Series not only has complete original features, but also has VM functions. (VM function depends on brand and model).

Suitable Hardware and Software

- One unit of (or several units for cascade) VST3300 Series
- Voice Mail product able to work with commercial PBX

Hardware Connection

Reserve a FXS port of the VST3300 device to connect with the line of VM.



12.1 Basic Configuration of VM and VST3300 Series

12.1.1 Configuration on VST3300 Series

- Enable a VM connection on an FXS port of VST3300 device

Web Path : **2.Channel\2.1.Summary**

Select the Channel with Type FXS and click its St (Status) icon.

CH	St.	Suffix	Type	DND	Fwd	Barring Class	T.38	Statistics In/Out	PickUp/Group	Ext. No.	Voice Mail	Gain In/Out
1		11/OP*	FXS	-	-	0	-	0/0	✓ / 1	1011	-	0/0
2		12	FXS	-	B/C*	0	-	0/0	✓ / 1	1012	-	0/0

Select Voice Mail in Status table

Connect Device Type

Type:

- Configure this FXS port to join the operator group in Join Operator Group table.

Join Operator Group

Yes/No:

- Disable DISA of the VST3300 and incoming call is answered by the DISA of VM. The DISA of VST3300 can deal with different kinds of incoming call, so disable only the incoming call type that is answered by DISA of VM

Web Path: **1.System /1.1.Basic Information**

Automated Attendant	
PSTN Call:	Disable
ITSP Call:	Enable
IP Call:	Enable

- Configure the Flash Time of both the VST3300 and VM to be the same value. For example, 200ms

Web Path : **1.System\1.8.Telephony Tuning**

Flash Button	
Flash Time:	200 msec.

12.1.2 Configuration on VM

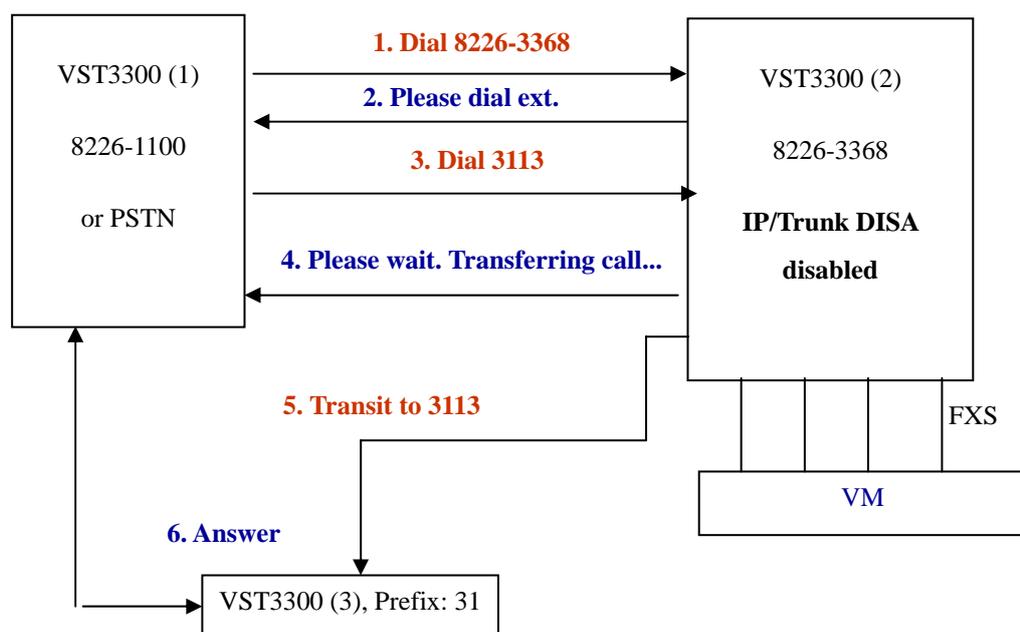
- Dial into the Voice Mail product from FXS port of VST3300 Series and configure Password, Enter code...etc. Don't forget to save its configuration
- Record the VM greeting. It may include a general greeting, International language greeting, holiday greeting...etc.
- Configure extension numbers. Some VMs may be able to group extension numbers. If so, extension numbers should be the same as the VST3300's. The default extension numbers of the VST3300 is 1011~1026, but can also be customized in the Extension Number Table. Since the VST3300DISA is disabled, call transit is processed by VM, so mailbox is setup according to extensions
- Configure ring back tone times. Configure how many rings before the VM DISA answers an incoming call.
- Configure the longest numbers of the extension. Please configure it according to the current VST3300's extension number. If extension numbers were created by VST3300 Extension Number Table, the longest length is 6 digits.
- Configure call transit action. When the DISA of VM answers the incoming call, it will play the greeting, prompt calling side to dial an extension number. These actions pass through the VST3300. So the call transit action needs to setup to work with the VST3300. For call transit, please setup it as below.
"Flash + Pause + Extension No. + 4 times of Pause"
 There are different VM brands and models, so configure according to real conditions. For configuration of Dial Ending Time, please refer to Web Path

1.System\1.8.Telephony Tuning

- Configure the extensions of main operator. Configure the VM's main operator to the operator's extension line of the VST3300 Series device.
- Configure the assistant operator. When the extension line of the main operator is busy, the call is transited to an assistant operator.
- Configure Operator Access Code for the Calling side to reach the operator.
- Configure transit greeting as "system transit greeting" or "personal transit greeting"

Operation Work Flow

1. VST3300 (2) accepts incoming calls from VST3300 (1) or PSTN. Since the DISA of VST3300 (2) is disabled, the call is answered by the DISA of VM
2. VM prompts calling side to dial extension number.
3. Calling side dials extension 3113
4. VM plays greeting "Please wait. Transferring call to 3113" and then sends **"Flash + Pause + Extension No. + 4 times of Pause"** to VST3300 (2)
5. VST3300 (2) transits the call to 3113. Extension 3113 of VST3300 (3) is ringing.
6. Called side, extension 3113 of VST3300 (3) answers the phone. VM releases the line, VST3300 (1) and VST3300 (3) route is built directly.



12.2 Message Configuration of VM and VST3300 Series

12.2.1 Use Call Forward Function of VST3300 Series to Connect VM

12.2.1.1 Configuration on VST3300 Series

- Configure the Prefix, Suffix of VM in VST3300 (2)

The design of Prefix and Suffix is for numbers using VM. Generally speaking, it needs to dial certain numbers as below to VM to activate voice mailbox.

Prefix + Extension Number (that use mailbox function) + Suffix

For example: *3122#

Different VM have different Prefixes, Suffixes or may not have Prefix, Suffix codes. If there is no such code, please leave it blank. Configure it for real conditions.

Web Path: **1.System\1.8.Telephony Tuning**

Voice Mail Access	
Message Waiting On:	<input type="text"/>
Message Waiting Off:	<input type="text"/>
VM Prefix:	<input type="text"/>
VM Suffix:	<input type="text"/>

- Configure Call forward of VST3300 (3)
Configure related FXS port of VST3300 (3) to Forward To "VM connected VST3300 (2)" when the line is busy or no answer (Busy/No Answer function).

Web Path: **2.Channel\2.1.Summary**

Select Channel with Analog FXS type or SIP Line and click its St (Status) icon.

CH	St.	Suffix	Type	DND	Fwd	Barring Class	T.38	Statistics In/Out	PickUp/Group	Ext. No.	Voice Mail	Gain In/Out
1		11/OP*	FXS	-	-	0	-	0/0	✓ /1	1011	-	0/0
2		12	FXS	-	B/C*	0	-	0/0	✓ /1	1012	-	0/0

SIP Line												
Ch	St	Suffix	Add User	DND	Fwd	Barring Class	2833 Status	Statistics In/Out	PickUp/Group	Ext. No.	User Name	
5		27	Set	-	-	0	-	0/0	✓ /2	1027	Christine1027	
6		28	Set	-	-	0	-	0/0	✓ /2	1028	Edward1028	

Configure Call Forward in Status table

Call Forward	
Control:	Busy/No Answer
Forward To:	886218400610
Mobile Number:	

- The default setting for VST3300 (3) is No answer, and sends greeting after 25 seconds. When the DISA of VST3300 (2) is enabled, the DISA will answer the incoming call with instruction greeting if the incoming call is not answered by called person in 25 seconds. If Busy/No Answer Forward is configured on the extension of VST3300 (3), then the incoming call is transferred to VST3300 (2) that will connect to VM in 20 seconds if no one answers the call.

Note: No Answer, send greeting (default is 20 seconds) minus 5 seconds is the Busy/No Answer Forward time

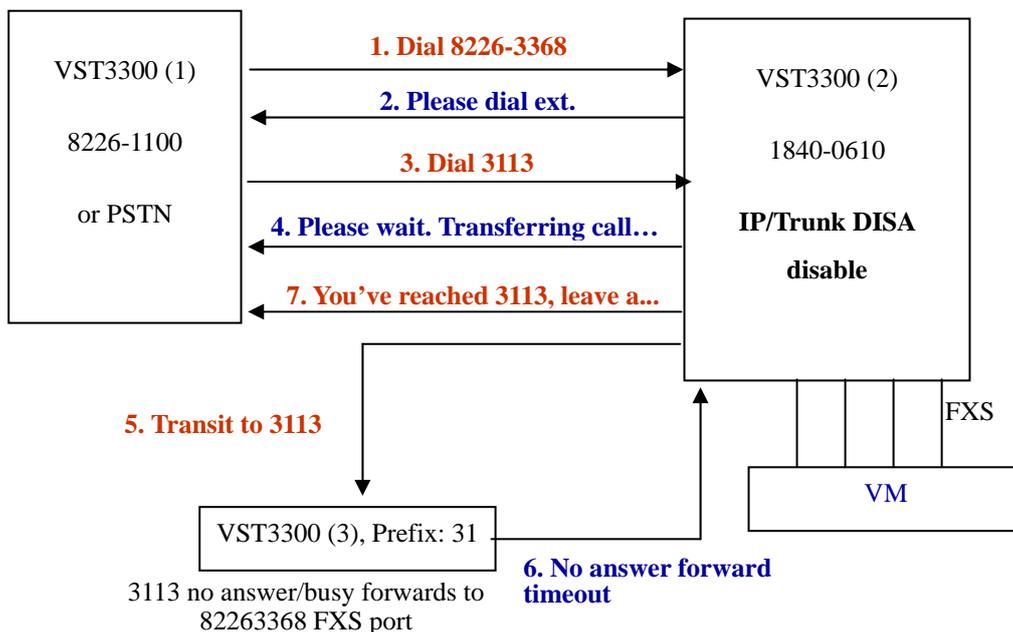
12.2.1.2 Configuration on VM

- Configure the extension number according to the section above Basic Configuration of VM and VST3300 Series
- Configure the password for extension line to retrieve Voice Mailbox.
- Configure the Voice Mail access code for incoming call. (For example: *)
- Configure Delete Mailbox, Maximum No. of Messages, Delete Message, Maximum Message Time, Min. Message Time...and other settings.

Operation Work Flow

1. VST3300 (2) accepts the incoming call from VST3300 (1) or PSTN. The DISA of VST3300 (2) is disabled, so the call is answered by the DISA of VM
2. VM prompts calling side to dial extension number.
3. Calling side dial 3113 extension. VM gets extension 3113 via VST3300 (2)
4. VM plays greeting "Please wait. Transferring call to 3113" and then sends "**Flash + Pause + Extension No. + 4 times of Pause**" to VST3300 (2)
5. VST3300 (2) transits the call to 3113. Extension 3113 of VST3300 (3) is rings.
6. Called side, extension 3113 of VST3300 (3) is busy or doesn't answer, so the call is forwarded to VST3300 (2) and then enters the VM.

7. VM plays the mailbox system greeting "You've reached extension 3113, leave a message". Calling side starts to leave a voice message.



12.2.2 Use VM Function of VST3300 Series and Busy Tone / Ring Back Detection Function of VM

Configuration on VST3300 Series

Setting is the same as the configuration of VST3300 Series in last section 12.2.1 Use Call Forward Function of VST3300 Series to Connect VM

Configuration on VM

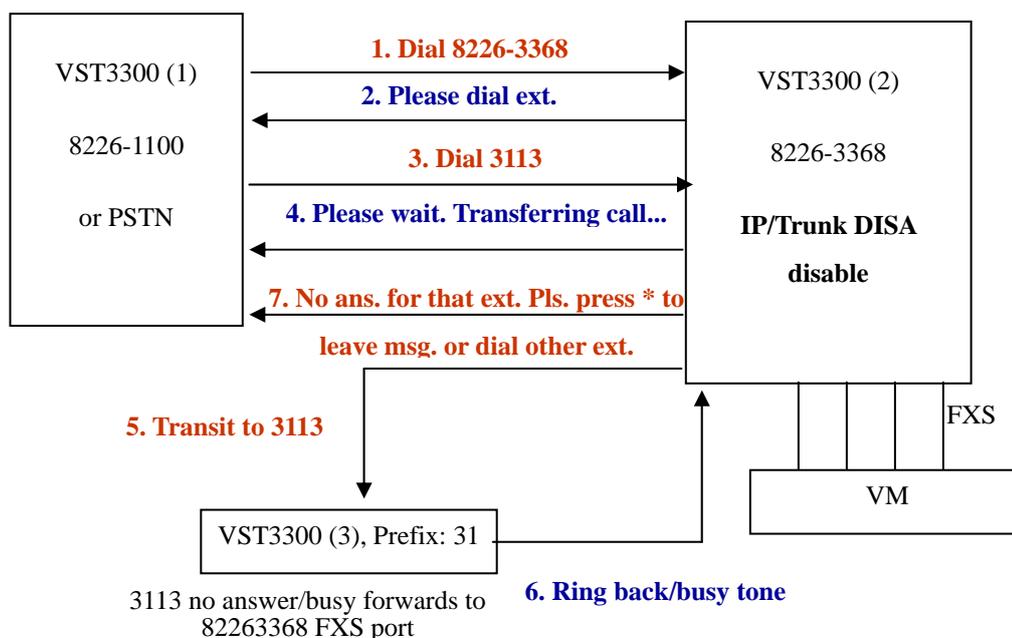
- Setting is the same as the configuration of VM in last section 12.2.1 Use Call Forward Function of VST3300 Series to Connect VM
- Enable detective mode of VM. The tone types for VM monitoring include:
 - Ring back tone detective times
 - Ring back tone cadence
 - Monitor action (Enable, Disable ring back tone detective mode)
 - Busy tone detective times
 - Busy tone cadence
 - Monitor action (Enable, Disable busy tone detective mode)
- Use system greeting to prompt calling side to leave a message
 - Disable mailbox personal greeting and enable mailbox system greeting

- Configure monitor action or cadence/times of tone that VM is going to monitor. In this example, we configure **No Answer** Forward time of VST3300 (3) to be larger than VM ring back/busy tone detective time. In this situation, VM ring back/busy tone detective time times out before **No Answer** Forward time, so VM retrieves the incoming call and plays system greeting to prompt caller to leave a message.

■ Use mailbox personal greeting to prompt caller to leave a message

- Enable mailbox personal greeting and disable mailbox system greeting

- Configure **No Answer** Forward time of VST3300 (3) to be smaller than VM ring back/busy tone detective time. In this situation, **No Answer** Forward time of VST3300 (3) times out before VM ring back/busy tone detective time; so incoming call is forwarded to FXS port of VST3300 (2) and then enters VM. VM identifies the caller and plays mailbox personal greeting prompting caller to leave a message.



No answer forward time > VM ring back/busy tone detective time

12.3 Retrieve (listen) Message / New Message Notification Related Configuration

When there is new message for an extension, VST3300 Series will signal this by changing the user of that extension's dial tone.

Most VM provide a new message prompt (light) function, however, most phone-sets do not have a message prompt light. To overcome this problem, VST3300 Series send short

Du, Du, Du tones and then a normal dial tone when the user takes the phone-set off hook, if the VST3300 receives the a message prompt code from the VM.

Upon a new message, general VM products send to a commercial PBX the following:

New message prompt code + Extension number

Upon retrieval of all new messages, general VM products send to a commercial PBX the following:

All new messages are retrieved code + Extension number

Configuration on VST3300 (2)

Please check the "new message prompt code" and "all new messages are retrieved code" of the VM you have. Input it in VST3300 Series. Configure as follows:

- Input "Message Waiting On code". This code must be the same as VM's new message prompt code.

When VST3300 receives "New Message Prompt code + Extension number", it will send Du, Du, Du tone and then normal dial tone to prompt the user there is a new voice message when the user takes the phone-set off hook.

Web Path : **1.System /1.8.Telephony Tuning**

Voice Mail Access	
Message Waiting On:	<input type="text"/>
Message Waiting Off:	<input type="text"/>

- Input "Message Waiting Off code", this code must be the same as VM's "All new messages are retrieved code".

When VST3300 has got "All new messages are retrieved code + Extension number", the dial tone of that extension will return to normal status.

Web Path : **1.System /1.8.Telephony Tuning**

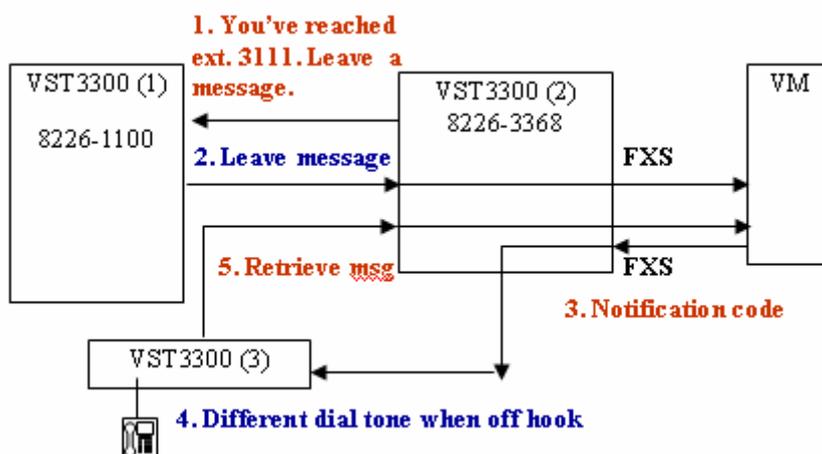
Voice Mail Access	
Message Waiting On:	<input type="text"/>
Message Waiting Off:	<input type="text"/>

Configuration on VM

- Enable message notification (light) function
- Enable new message prompt function
- Enable extension line message notification (light) function
- Enable oral notification function (Extra VM function)

Operation Work Flow

1. According to previous example, VM will prompt caller to leave a message
2. Caller leaves a message. Voice message pass through VST3300 (2), enters VM and leaves a message successfully.
3. VM sends code "New message prompt code + Extension number" to the FXS port of VST3300 (3)
4. VST3300 (3) receives the code and then changes the dial tone. When user of that extension take the phone set off hook, he/she will hear Du, Du, Du before normal dialing tone. He/she knows that a new message is waiting for retrieval.
5. When the extension of VST3300 (3) needs to retrieve (listen) the voice message, they call to VST3300 (2) and it are answered by the DISA of VM. They retrieve the message by the instruction of VM.



12.4 Backup on VM System

Most VM products can connect with a PC for online or offline editing and system backup. Please backup the data to PC for system update or restoration.

13. Create the Connection of SIP Device

13.1 Concept:

A SIP telephony network is a kind of Public VoIP specification. SIP devices (IP Phone, Softphone, Wi-Fi Phone, Gateways, etc) complying with SIP standards can work with this device.

SIP devices can register to VST3300 Series and become an extension of the whole IP-PBX. SIP devices can work together with an analog phone set to make calls with each other like an extension.

The VST3300 Series can also acts as SIP Client to register to ITSP (Internet Telephony Service Provider). Then, user can receive services from ITSP, such as free call inside network, cost saving call to PSTN, etc.

IP Phone Community networks provides better telephony service than SIP VoIP networks. IP Phone Community network means that Edge-Core VoIP products have an IP Phone Community Number that allow products to communicate with each other via that number. It includes the VoIP network and extra value services built from VST3300 devices.

13.2 SIP Line (SIP Proxy Server)

VST3300 Series is also a SIP Proxy Server. SIP devices can register to this machine and become an extension of the IP-PBX.

13.2.1 Open a New Account

When a new SIP device intends to register to this machine as an extension, it can use a default pre-configured account and extension number, or the administrator can create a new account for this device.

13.2.1.1 Default Pre-configured Account and Extension Number

- All SIP channels have default User Name (account) from Ext27 upward when this machine is shipped. You can purchase extra available accounts. Accounts can go upward to Ext** (** means 2 or 3 digits)
- All SIP channels have default Extension Number from 1027 upward upon shipping. The number can go upward to ExtXX (XX means 2 or 3 digits) depends on account available.

Web Path : **2.Channel\2.1.Summary, SIP Line Table**

SIP Line												
Ch	St	Suffix	Add User	DND	Fwd	Barring Class	2833 Status	Statistics In/Out	PickUp/Group	Ext. No.	User Name	
5		27	Set	-	-	0	-	0/0	✓/2	1027	Christine1027	
6		28	Set	-	-	0	-	0/0	✓/2	1028	Edward1028	
7		29	Set	-	-	0	-	0/0	✓/2	1029	Lyd1029	
8		30	Set	-	-	0	-	0/0	✓/2	1030	Lyd1030	

13.2.1.2 Create a New Account

Click Set in Add User column according to Channel.

Web Path: **2.Channel\2.1.Summary, SIP Line Table**

SIP Line												
Ch	St	Suffix	Add User	DND	Fwd	Barring Class	2833 Status	Statistics In/Out	PickUp/Group	Ext. No.	User Name	
5		27	Set	-	-	0	-	0/0	✓/2	1027	Christine1027	
6		28	Set	-	-	0	-	0/0	✓/2	1028	Edward1028	
7		29	Set	-	-	0	-	0/0	✓/2	1029	Lyd1029	
8		30	Set	-	-	0	-	0/0	✓/2	1030	Lyd1030	

Create a new account that is the same as the User Name that is pre-configured by default for extension number, such as Ext27, and also input password. Finally, click Apply. This can simplify the configuration procedure later.

You can also configure customized accounts, such as

SIP Line User Account			
	Username	Password	Confirm Password
Add/Modify:	<input type="text" value="Lester"/>	<input type="password" value="****"/>	<input type="password" value="****"/>
Delete:	<input type="text"/>		

13.2.2 Specify a Channel to the New Created Account

When configuration of account is done, specify this account to a certain SIP Channel. SIP device that register to this machine can use the specify account to register the certain SIP Channel. Later, it can make call or answer call with this channel.

Web Path: **Channel\2.1.Summary, SIP Line Table**

SIP Line												
Ch	St	Suffix	Add User	DND	Fwd	Barring Class	2833 Status	Statistics In/Out	PickUp/Group	Ext. No.	User Name	
5		27	Set	-	-	0	-	0/0	✓ /2	1027	Christine1027	
6		28	Set	-	-	0	-	0/0	✓ /2	1028	Edward1028	
7		29	Set	-	-	0	-	0/0	✓ /2	1029	Lyd1029	
8		30	Set	-	-	0	-	0/0	✓ /2	1030	Lyd1030	

Click channel St (status) icon marked with X (X means the channel is not registered from other SIP device. Note: Channel that is specified with account also marks X if the SIP device does not register to this machine)

Confirm in the Status Table that Channel State is enabled and User Name is the account configured in the last section.

SIP Line Information		SIP Device	
Channel:	7	User Name:	Ext29
Extension Suffix:	29	Outgoing Call	
Admin. State:	Enable	Privilege:	International
Operation State:	Enable	Join Operator Group	

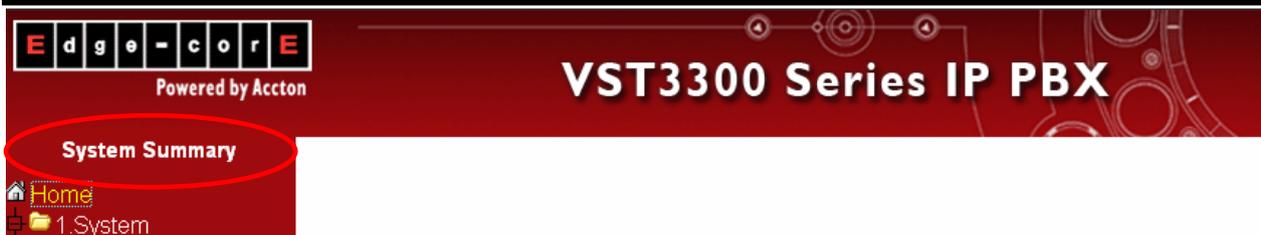
User Name is selectable to avoid input error. Double Click User Name and it shows the sub-menu. Select the User Name that was configured in the last section.

SIP Device	
User Name:	Ext29
Privilege:	International

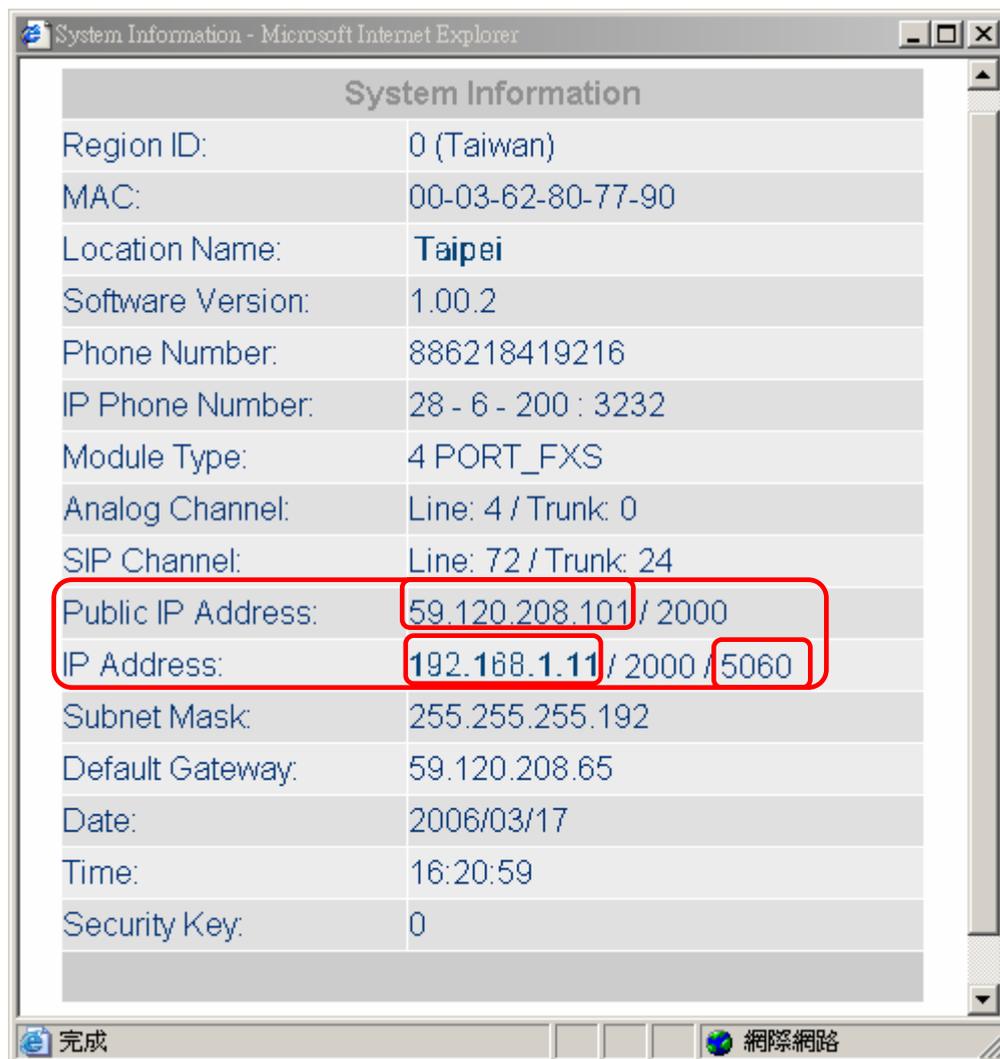
13.2.3 Registration of SIP Device (Client)

To register this machine, configure the parameter of SIP device (client) is required. The parameters of this machine are shown below. (Please configure it according to your real environment; the example below is for reference)

Please click System Summary at the left top corner of Web of this machine to find the information



System shows the information of this machine as seen below.



◆ **Outbound Proxy:**

If SIP device and VST3300 Series is under the same Subnet, please use IP address in LAN. For the example above, use 192.168.1.11, otherwise, use Public IP 59.120.208.101

◆ **Registrar:**

If SIP device and VST3300 Series is under the same Subnet, please use IP address in LAN. For the example above, use 192.168.1.11, otherwise, use Public IP 59.120.208.101

- ◆ Port No: Default is 5060. If different, change it according to your own setting.
- ◆ Public Address (Account): Account@Proxy Server

Check account name from Web Path: **Channel Config\2.1.Summary, SIP Line Table**

This depends on the Channel that the SIP device uses.

SIP Line											
Ch	St	Suffix	Add User	DND	Fwd	Barring Class	2833 Status	Statistics In/Out	PickUp/Group	Ext. No.	User Name
5		27	Set	-	-	0	-	0/0	✓ / 2	1027	Christine1027
6		28	Set	-	-	0	-	0/0	✓ / 2	1028	Edward1028
7		29	Set	-	-	0	-	0/0	✓ / 2	1029	Lyd1029
8		30	Set	-	-	0	-	0/0	✓ / 2	1030	Lyd1030

Example of Public Address (Account): Ext30@59.120.208.101 , Ext28@192.168.1.11

- ◆ User Name (Account: as the example above, such as Ext30, Christine1027)
- ◆ Password: It depends on the Password of User Name for this Channel. Please refer to the Password in section 13.2.1.2 Create a New Account

Use the parameters found in this chapter, SIP device (client) configured with these parameters can now register to this machine.

13.2.4 Check Registration Status

Use the new account, password and related information, SIP device is now able to register this machine. The registration status can be checked by the VST3300 and includes if Softphone registration to this machine successfully.

Web Path: **3.SIP \3.1.Online Users**

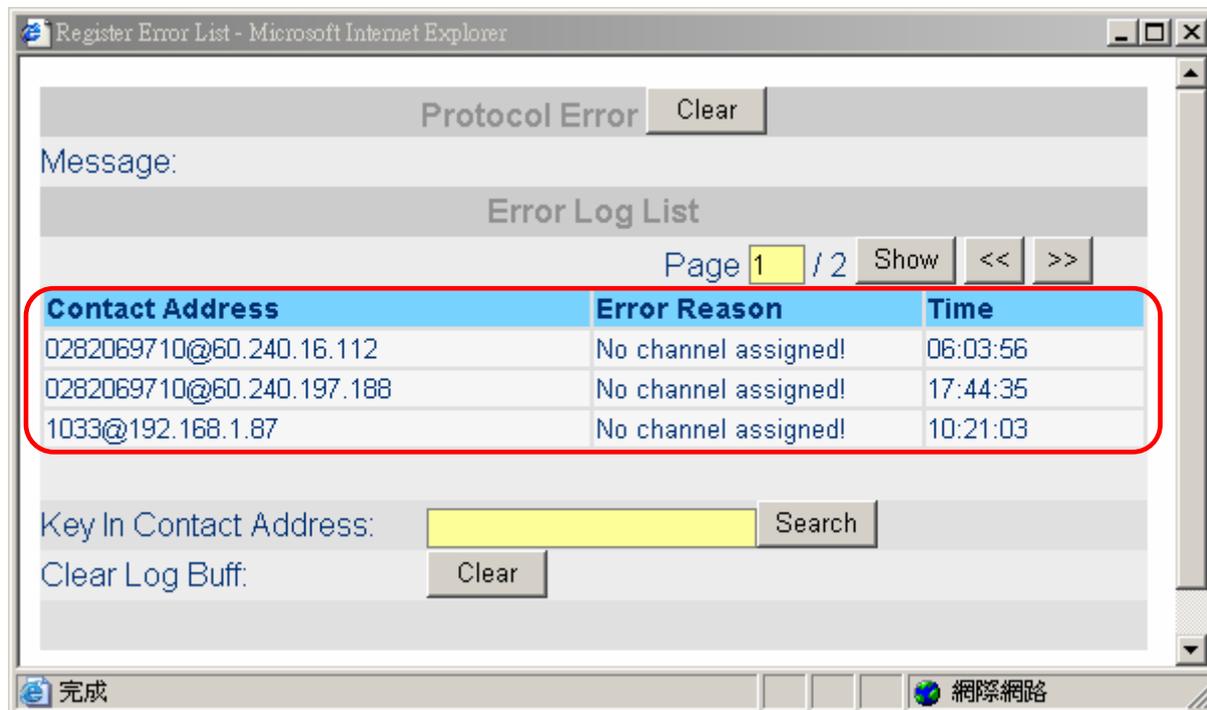
Contacts List: Page 1 / 1 Show << >> Error Log...			
User	Contact URL	Expires (sec)	Control
Jason1067	SIP:Jason1067@220.130.191.134:3099	2740	DeRegister
steven1039	SIP:steven1039@60.240.197.188:5060	500	DeRegister

SIP device no longer used, can be removed from registration. Click Deregister to terminate the registration of that SIP device. If registration has failed, check Log to know for the reason of registration is failure.

Contacts List: Page 1 / 1 Show << >> Error Log...			
User	Contact URL	Expires (sec)	Control
Jason1067	SIP:Jason1067@220.130.191.134:3099	2740	DeRegister
steven1039	SIP:steven1039@60.240.197.188:5060	500	DeRegister

Contact Address, Error Reason and Time shows the Contact Address, fail reason and time of registration attempt for the registration fail SIP device.

Common error are due to Password/Account error or No channel assigned.



13.3 SIP Trunk Server

In addition to FXO (analog trunk) ports, this machine also has SIP Trunks that can register to SIP Service Provider (ITSP). The extension of this machine can seize trunk or use auto routing to enter ITSP. The SIP connection of ITSP may offer free calls inside the network and cost savings on call to PSTN.

13.3.1 Register to SIP Telephony Server Provider

Assume the following registration information of an ITSP:

SIP Outbound Proxy: fwd.pulver.com

Port Number: 5060

Registrar: fwd.pulver.com

Port Number: 5060

Your number: 211.

So, Public Address of SIP Phone Number is "211@fwd.pulver.com"

Password is 1234

Input the information above to

Web Path: **3.SIP \ 3.2.ITSP**

(Need Warm-Restart)

Apply Cancel

Registrar Setting

Domain Name (IP:Port): fwd.pulver.com : 5060 Enable

Outbound Proxy Setting

Domain Name: fwd.pulver.com Enable

Port: 5060

Register Expiration

Time Interval (30~86400 sec.): 3600 sec.

Incoming Call Screening

Accept Calls From Proxy Only: No

Later, be sure to check the registration status.

Web Path: **2.Channel\2.1.Summary, SIP Trunk Table**

SIP Trunk									
Ch	St	Access	Contact Address	Fwd	Trunk Class	2833 Status	Statistics In/Out	Reg.	Last Number Dialed
173		8	13	-	12	-	0/0	-	
174		8	14	-	12	-	0/0	-	Not Registered
175		8	15	-	12	-	0/0	-	-
176		8	16	-	12	-	0/0	-	-

13.3.2 Configuration of SIP Trunk Dialing Behavior

A call from SIP Trunk is routed to the ITSP (Internet Telephony Service Provider) and connected to another SIP phone by that ITSP. Dial 8 to seize a SIP Trunk, you will hear the dial tone from ITSP, and can then make a SIP Phone call (Suggested method).

If auto-routing function is used, dialing behavior here needs to be configure to make dialing number correct for different ITSPs, because the installation location of this machine may be different from the ITSP that this machine registers to.

The real dialing number of SIP Trunk is composed according to the rule here (add or subtract Country Code, Area Code, Access Code)

Web Path: **3.SIP\3.3.Trunk Numbering**

Setting		
Country Code:	<input style="width: 100%;" type="text"/>	
Area Code:	<input style="width: 100%;" type="text"/>	
Add/Remove Digits		
Call Type	Remove Leading N Digit(s)	Add Leading Digit String
International Call	<input style="width: 50%;" type="text" value="0"/> ▼	<input style="width: 100%;" type="text"/>
Long Distance Call	<input style="width: 50%;" type="text" value="0"/> ▼	<input style="width: 100%;" type="text"/>
Local Call	<input style="width: 50%;" type="text" value="0"/> ▼	<input style="width: 100%;" type="text"/>

[Simulator](#)

Group	Item	Description	Default Value
Setting	Country Code	Country Code of ITSP	Blank
	Area Code	Area code of SIP Telecom service of the ITSP country above	Blank
Add/Remove Digits	Call Type	The prefix type that add or subtract International Call Long Distance Call Local Call	
	Remove Leading N Digit(s)	Remove the first N digits of access code (prefix) of the types below. It can be 0 (default) to 10 International Call Long Distance Call Local Call	0
	Add Leading Digit String	After N digits is removed (described in the paragraph above), add access code (prefix) digits of the types below. It can be a digit string from 1~9. International Call Long Distance Call Local Call	Blank
	Simulator	Test the real number dial that is simulated from the add/subtract rule of this system. It has to use full number to simulate, including Country Code + Area Code + Local phone number.	

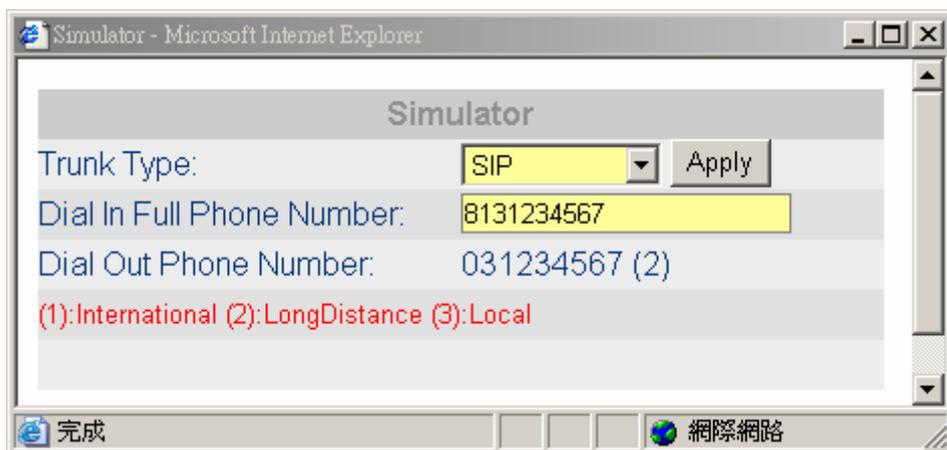
Note: If no rules are applied on SIP Trunk Web page, the system follows the standard

telecom rule (E.164) to process the number. The system exams the full number (Country Code + Area Code + Local phone number) of destination and decides the real number and dials according to the rules below.

- If the Country Code and Area Code of destination are the same as the configuration of SIP Trunk. System dials Local Phone Number only. (Country Code and Area Code of destination number is removed)
- If the destination Country Code is the same, but area code is different, the number dials by the system is: the long distance Access Code + Area Code + Local Phone Number. (destination Country Code is removed and long distance Access Code is added)
- If the destination Country Code is different, the number the system dials is: the International Access Code of the machine's country + Country Code + Area Code. (Add Internation Access Code in front of destination number)

Note: The International Access Code and Long Distance Access Code are configured when the configuration of Region ID is done. To configure Region ID, please refer to VST3300 Series Quick Installation Guide, Section 5.2 Telecom Region ID Configuration. To find the International Access Code and Long Distance Access Code, please refer to Web page: **4.PBX Advanced\4.1.Trunk Access Code**

Simulator Figure



Group	Item	Description	Default Value
Simulator	Trunk Type	Select the Trunk Type. There are Analog and SIP trunk. Please select SIP and click Apply	
	Dial In Full Phone Number	Input the full destination number (including Country Code + Area Code + Local phone number and International Access Code is excluded) for simulation.	Blank
	Dial Out Phone Number	The real number dials from the system. The suffix (1), (2), (3) means the call types below.	
	(1):International (2):Long Distance (3):Local	The call type that is judged by the system. There are: (1):International (2): Long Distance (3): Local	

13.3.2.1 Dialing Example

■ Standard dialing method:

General SIP ITSPs use the same network phone number rule as traditional PSTNs and the dialing behavior is the same as PSTN. You should know the network phone number of destination and compare it with the location of your SIP ITSP (calling side).

- If the country code and area code are the same, you only need to dial Local Phone Number of called side.
- If the country code is the same, but area code is different, you need to dial Long Distance Access Code + Area Code + Local Phone Number (Long Distance Access Code needs to comply with the telecom rules of the location of SIP ITSP)
- If the country code is different, you need to dial International Access Code + Country Code + Area Code + Local Phone Number (International Access Code needs to comply with the telecom rule of the location of SIP ITSP)

If dialing behavior (standard) of SIP ITSP is the same as traditional PSTN, and ITSP use the same Country Code, Area Code as the configuration of this machine, then no configuration of Add/Subtract rule is required.

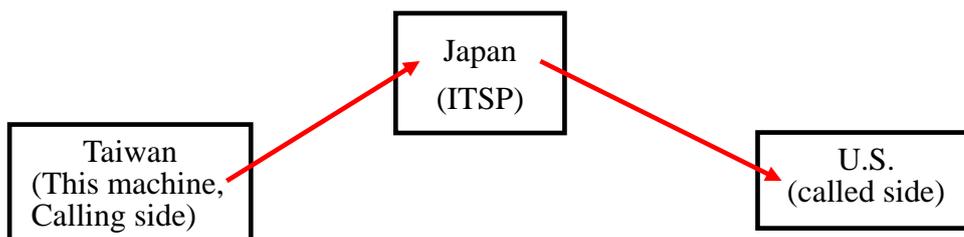
◆ **Example (1)** of Add/Subtract rule

When the location of this machine and the registration of ITSP and the destination place that user make call are different, then Add/Subtract rule should be applied to make real number (Access Code, Country Code, Area Code, Local Phone Number) correct

For example: (Codes may not be same as yours, adjust according to real condition)

Role	Location	International Access Code	Country Code	Long Distance Access Code	Area Code
Location of this machine (calling side)	Taiwan	002	886 (3 digits)	0	2 (1 digit)
Registration of ITSP	Japan	010	81 (2 digits)	0	50 (2 digits)
Called side	U.S.	011	1 (1 digit)	1	408 (digits)

Assumes the Local Phone Number of U.S. called side is 2345678,
Since the location of calling side is Taiwan, the number dialed is 002 1 408 2345678
(Don't add International Access Code 002 when you do simulation)



In this example, calling side make a call to US, however, it routed through Japan's ITSP, so the number of called side has to comply with Japan's telecom standard. The number dialed from Japan's ITSP should be 010 1 408 2345678. The setting of Add/Subtract rule should subtract the first 3 digits (002) of original dialing 002 1 408 2345678, and then add Japan's International Access Code: 010, and become 010 1 408 2345678 finally.

◆ **Configuration of This Machine**

Configure value below

Web Path: **3.SIP Environment\3.4.SIP Trunk Numbering**

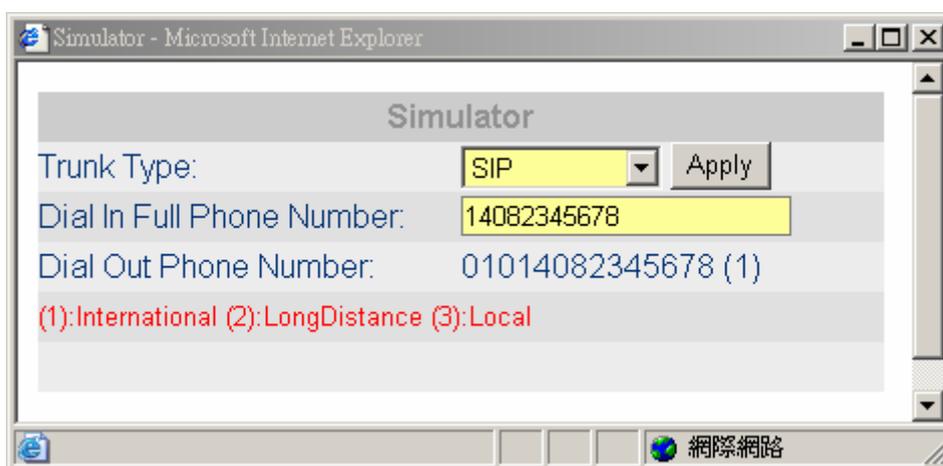
Japan: Country Code: 81, Area Code: 50 (Country Code and Area Code of ITSP)

Setting		
Country Code:	<input type="text" value="81"/>	
Area Code:	<input type="text" value="50"/>	
Add/Remove Digits		
Call Type	Remove Leading N Digit(s)	Add Leading Digit String
International Call	<input type="text" value="3"/>	<input type="text" value="010"/>
Long Distance Call	<input type="text" value="1"/>	<input type="text" value="0"/>
Local Call	<input type="text" value="0"/>	<input type="text"/>

[Simulator](#)

Group	Item	Description	Default Value
Setting	Country Code	Country Code of Japan	81
	Area Code	Area Code of the location of Japan	50
Add/Remove Digits	Remove Leading N Digit(s)	International Call: 3, in this example, the digits actually removed are 002 and it is the International Access Code of Taiwan, 3 digits.	3
		Long Distance Call: 1, the Long Distance Access Code of Taiwan is 0, 1 digit. (In this example, Distance Access Code is not required for making international call, so it doesn't affect the result). Since the Long Distance Access Code of Japan is the same as Taiwan's 0, so configure the field to 0 is also OK. If 0 is configured, the Add Leading Digit String at Long Distance Call field below is not required.	1→Set 0 at below field 0→Keep blank at below filed
		Local Call: 0, number is not changed, set 0	0

Group	Item	Description	Default Value
	Add Leading Digit String	International Call: 010, International Access Code of Japan Long Distance Call: 0, Long Distance Access Call of Japan Local Call: Blank. Needs not to add and don't change	010 0→When 1 at field above is configured Blank→ When 0 at field above is configured
	Simulator	Input 14082345678 to simulator (International Access Code of Taiwan, 002 is excluded), then output 01014082345678 (1)	Blank



◆ **Other Example**

1. The same as the Country Code and Area Code of Japan (Example 2)

The destination Local Phone Number is 22345678 and Area Code is 50
Calling side (Taiwan) dials: 002 81 50 22345678, (002 is excluded when you do simulation), then system simulator dials 22345678 (3), because the ITSP use the same Country Code and Area Code and it is regarded as Local call.

2. Country Code is the same, but Area Code is different (Example 3)

The destination Local Phone Number is 22345678 and Area Code is 3
Calling side (Taiwan) dials: 002 81 3 22345678, (002 is excluded when you do simulation), then system simulator dials 0 322345678 (2), because the ITSP use the same Country Code, but Area Code is different and it is regarded as Long Distance call.

3. The same as the Country Code and Area Code of Japan, however, add digits are required (VoIP telecom standard that define 050 Access Code even it is in the same area). Here is the add/subtract rule: (Example 4)

Setting		
Country Code:	<input type="text" value="81"/>	
Area Code:	<input type="text" value="50"/>	
Add/Remove Digits		
Call Type	Remove Leading N Digit(s)	Add Leading Digit String
International Call	<input type="text" value="3"/>	<input type="text" value="010"/>
Long Distance Call	<input type="text" value="1"/>	<input type="text" value="0"/>
Local Call	<input type="text" value="0"/>	<input type="text" value="050"/>

[Simulator](#)

The destination Local Phone Number is 22345678 and Area Code is 50
 Calling side (Taiwan) dials: 002 81 50 22345678, (002 is excluded when you do simulation), then system simulator dials 05022345678 (3), because the ITSP use the same Country Code and Area Code and it is regarded as Local call.

13.3.2.2 Use Simulator to Simplify Configuration

After the instructions above, you can use Simulator first to achieve the result you want, without thinking adding/subtracting digits in which field first.

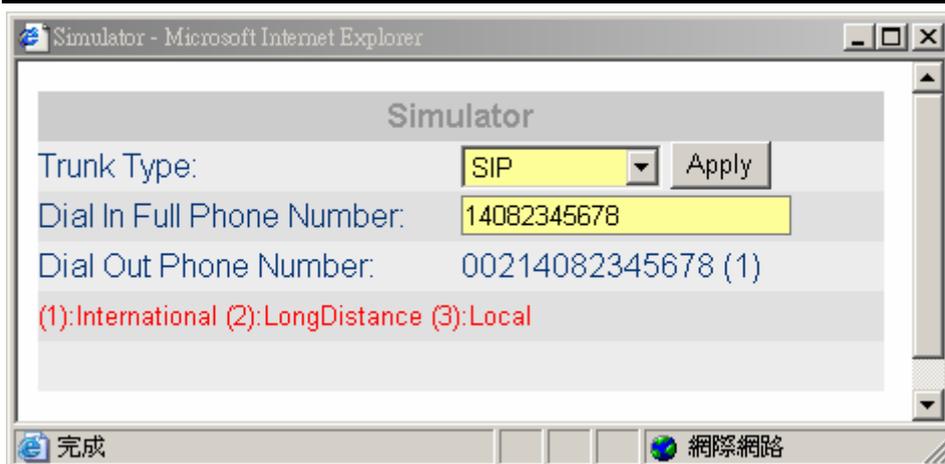
Input the Country Code, Area Code of ITSP field in Country Code, Area Code field, such as Japan, and then click Apply button.

Setting		
Country Code:	<input type="text" value="81"/>	
Area Code:	<input type="text" value="50"/>	
Add/Remove Digits		
Call Type	Remove Leading N Digit(s)	Add Leading Digit String
International Call	<input type="text" value="0"/>	<input type="text"/>
Long Distance Call	<input type="text" value="0"/>	<input type="text"/>
Local Call	<input type="text" value="0"/>	<input type="text"/>

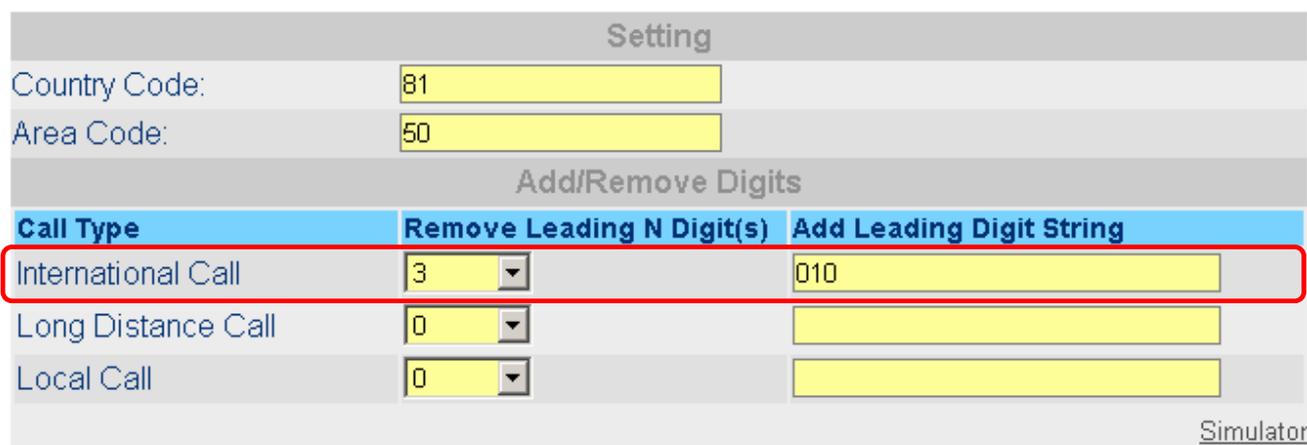
[Simulator](#)

Simulate the number you want to dial with default value (no add/subtract digits). Get the result to decide which fields needs to be added/subtracted.

For the example (1), input number and get number below.



To remove 002 (3 digits) and add the International Access Code of Japan, 010:



You can also apply these rules to tune the number in other examples.

This method is more complicated. A better method for dialing behavior control will be provided on the next version.

14. Configuration of Conference Call

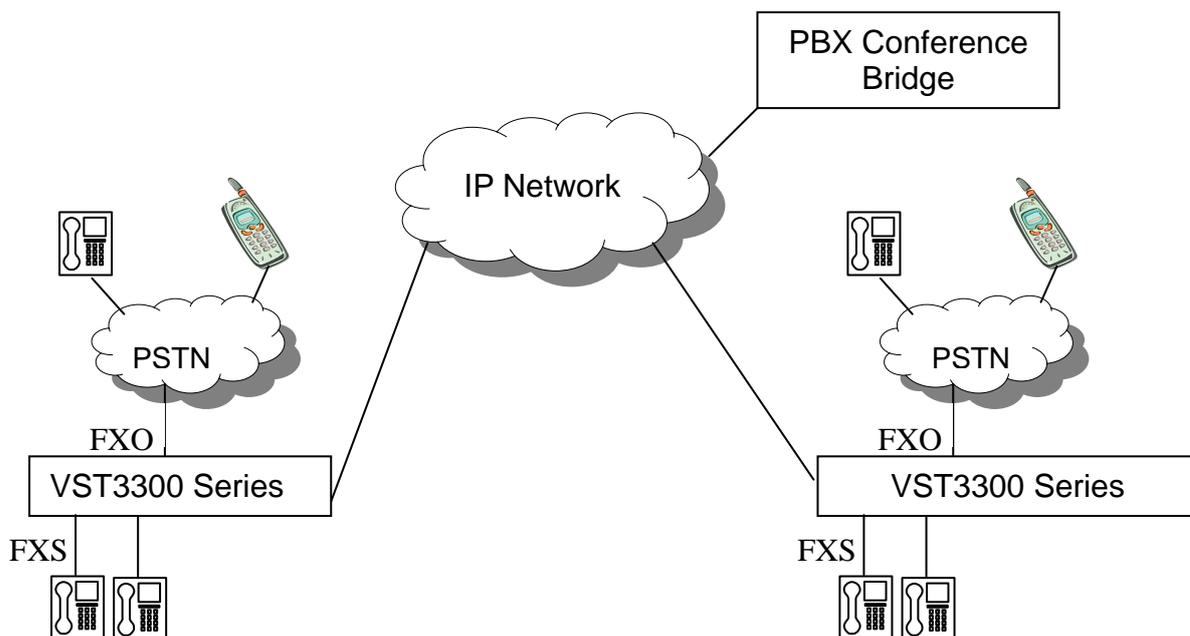
To use this function, a PBX Conference Bridge is required. This function is not included in the VST3300 Series and needs to be purchased additionally.

When working with a PBX Conference Bridge, a user (chairman) is able to call other parties to invite them to join a conference. Conference Bridges can be cascaded. You may access many Conference Bridges with only one conference ID assigned. The system will search for free conference when user requests.

For details of Conference Bridge, please refer to Conference Bridge User Manual

14.1 Configuration of a Single Conference Bridge

Please connect the devices as the example below.



- Configure a Prefix/Ext. to connect to Conference Bridge

Input a Prefix or Community IP Phone Number of Conference Bridge to Phone Number field.

Web Path : **1.System \1.2.Extension Number**

	Prefix/Ext. (Max.6 digits)	Phone Number	Type
Add/Modify:	3200	889944964496	Conference

As the example above, 3200 is the Prefix/Ext. of Conference Bridge (Phone Number: 889944964496)

- If this machine or Conference Bridge does not join IP Phone Community, input the information of Conference Bridge into the Phone Book

Web Path: **6.Phone Book**

	Phone Number	IP Address	Port
Add/Modify:	889944964496	10.13.6.130	2000

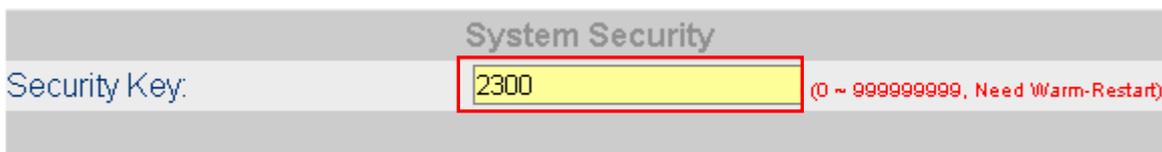
As the example above, phone number of Conference Bridge is 889944964496, IP is 10.13.6.130, and Port is 2000. Input all of them to web page Phone Book of this machine.

14.1.1 Chairman is Unspecified Member

Configuration of one or more Conference Bridge has to be verified with the built-in Security Key of this machine (VST3300 Series)

Authority Key of each Conference Bridge has to be the same as the Security Key of this machine.

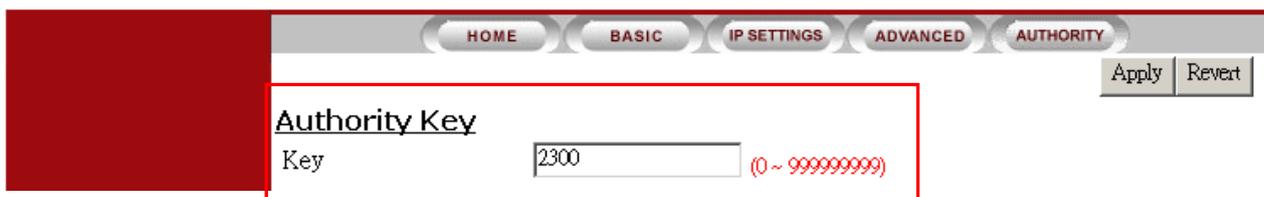
- VST3300 Series Web Path: **1.System\1.1.Basic Information**



As the example above, configure Security Key to 2300 and then click **Apply**.

- Configure the Authority Key of Conference Bridge to the same as Security Setting of Parking Server of this machine.

Conference Bridge Web Path: **AUTHORITY**



As the example above, configure Authority Key of Conference Bridge to the same as 2300

Follow the configuration above; each port of the VST3300 is able to use the Conference Bridge. When conversation starts, a party who activates Consult Transfer to Conference Bridge will become chairman. (For example: Flash+3200+#, 3200 is the Prefix/Ext. of Conference Bridge (phone number: 889944964496))

14.1.2 Specify Dedicated Member as Chairman

The chairmen of Conference Bridge can be specified as a dedicated person. Specified VST3300 Series ports can be configured in Conference Bridge. Only the party at these specified ports can become chairman. Users at other ports of the VST3300 can be invited by the chairman to join the conference only. Check the following configuration

- VST3300 Series Web Path: **1.System\1.1.Basic Information**

Configure Security Key of this system

System Security	
Security Key:	<input type="text" value="2300"/> (0 ~ 999999999, Need Warm-Restart)

As the example above, configure Security Key to 2300, and click **Apply**

- Authority Key of Conference Bridge has to be different from this machine

Conference Bridge Web Path: **AUTHORITY**

Authority Key	
Key	<input type="text" value="0"/> (0 ~ 999999999)

As the example above, configure it to 0

- Specify certain ports or all ports of VST3300 Series to be the chairman.

Conference Bridge Web Path: **AUTHORITY**

Authority Phone Number of Chairman	
Search	<input type="text"/>
Add	<input type="text" value="886272235511"/>
Delete	<input type="text"/>

As the example above, 886272235511 is the phone number of a VST3300 Series, then all parties of that machine can be a chairman.

Conference Bridge Web Path: **AUTHORITY**

Authority Phone Number of Chairman	
Search	<input type="text"/>
Add	<input type="text" value="88627223551112"/>
Delete	<input type="text"/>

If the phone number is specified to certain port, such as 88627223551112, then the last two digits, 12, are Channel Number. Only that Channel can be a chairman. Many Channels can be specified for chairman in the same IP-PBX.

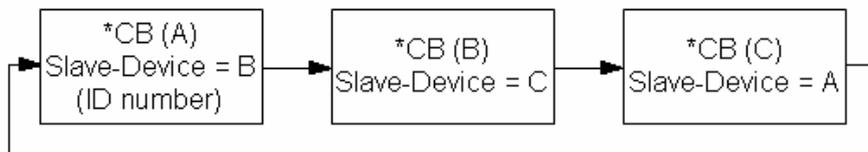
- For all changes, click **Apply** button to confirm

14.2 Configuration of Cascaded Conference Bridge

Conference Bridges can be cascaded as a round chain with max 16 devices. It is necessary to configure the IP information of next (slave) IP-PBX in the Web Page. Users

should select one IP-PBX number as the identification number of conference.

See the diagram below:



*CB= Conference Bridge

For Conference Bridge(A), Input **IP address** and **Port Number** of next (slave) Conference Bridge(B) in A's **Slave Conference Bridge Chain Address** field via web page. When Conference Bridge (A) is busy, the IP-PBX redirects the request to the next IP-PBX configured here, Conference Bridge(B). Conference Bridge(B) also can configure its next IP-PBX. When the last IP-PBX is configured, a **Slave Conference Bridge Chain Address** to the first Conference Bridge (A), a round chain structure, has been created. A maximum 16 IP-PBXs can be chained together. The chain information can be viewed in **Conference Bridge Chain Info** field.

Conference Bridge Web Path : **BASIC \ CONFERENCE**

Slave Conference Bridge Chain Address (Need Warm-Restart)

IP / Port

Conference Bridge Chain Info

Phone Number	IP1	Port1	IP2	Port2	Chairman Phone Number
88624496	10.13.6.4	2000	211.21.40.178	64999	-

As the example above, configure IP / Port field of next (slave) Conference Bridge

15. Configuration of Other Application

15.1 Call Forward

The feature Call Forward predefines a destination for an extension line automatically sending calls to the defined destination configured in the VST3300 Series are:

- The extension line in the same IP-PBX
- The extension line of another VST3300 Series at remote side
- SIP device (SIP Phone , Softphone , SIP Wi-Fi phone)
- The public line (PSTN) through the machine at remote side (Off-net Forward)

For the Call Forward feature, there are 3 parameters to be configured:

Parameter	Description
Control	<p>Disable: Disables Forward Feature</p> <p>All Calls: Forwards all calls</p> <p>Busy: Forwards only if this line is busy</p> <p>Busy (Cascade): Forwards only if this line is busy. And if the forwarded line is also busy and Busy (Cascade) is defined, this call will be forwarded to next destination as configured. Can be forward maximum of 16 times..</p> <p>No Answer: When there is no answer for this line, the call forwards to the specified destination</p> <p>Busy/No Answer: When there is no answer or line busy for this channel, the call forwards to the specified destination</p>
Forward To	<p>Phone number, extension number, or Community IP Phone Number of the remote device to be forwarded to.</p> <p>If the call needs to forwarded to a Voice Mail Product, configure this number as the phone number or Community IP Phone Number of VST3300 Series that connect to Voice Mail Product.</p> <p>The phone number must be a full number including country code and area code. If Mobile Number (Offnet to PSTN) below is also configured. The call is offnet to PSTN via the machine of the phone number configured here.</p>
Mobile Number	<p>The telephone number of PSTN or mobile phone that the call will be forwarded to. The telephone number is entered from the viewpoint of transfer IP-PBX (the remote equipment that the phone number had entered in the field of "Forward To").</p>

For general Call Forward, only the parameters of **Control** and **Forward To** need to be configured. There are two kinds of configuration tools, via Web or Phone set.

15.1.1 General Call Forward

Each extension can have their own call forward. Here is the configuration.

Configured Call Forward by Phone set

- (1) Take the phone set Off-hook and dial ; *0 0000 (Du, Du, Du)
- (2) Dial 011 (DuDuDu) ; Activates Forward

- 0 : Disable
- 1 : All Calls
- 2 : Busy
- 3 : No Answer
- 4 : Busy or No Answer
- 5 : Busy (Cascade)

- (3) Dial 0286216666111 ; Defines field of "Forward To"
- (4) Hang up the phone set

Disable Call Forward by Phone set

- (1) Off-hook the phone set and dial *0 0000 (Du, Du, Du)
- (2) Dial 010 (Du, Du, Du) ; Disable Call Forward
- (3) Hang up the phone set

Configure General Call Forward by Web

Web Path : **2.Channel\2.1.Summary**

Select Channel with FXS type and click its St (Status) icon

CH	St.	Suffix	Type	DND	Fwd	Barring Class	T.38	Statistics In/Out	PickUp/Group	Ext. No.	Voice Mail	Gain In/Out
1		11/OP*	FXS	-	-	0	-	0/0	✓ / 1	1011	-	0/0
2		12	FXS	-	B/C*	0	-	0/0	✓ / 1	1012	-	0/0

Configure Call Forward in Status Table

Call Forward

Control: All Calls

Forward To: 88628226339124

Mobile Number:

Fwd

-

A

It shows icon after configuration

Icon of other status is shown below

- A:** All Calls
- B:** Busy
- B/C:** Busy (cascade)
- N:** No Answer
- B/N:** Busy or No Answer

	Activate Call Forward	Disable Call Forward
Control	All Calls	Disable
Forward To	The phone number (full number, can add an extension number) of the destination or Community IP Phone Number	
Offnet To	Blank	

15.1.2 Secretarial Intercept Feature

In a company scenario, phone calls may be picked up by a secretary. Phone calls are then filtered by secretary and it is transferred to the correct destination.

1. Configure all calls to a boss to be forwarded to the extension line of the secretary, then any calls that for the boss will be forwarded to the secretary.

Configure Call Forward in Status Table

	Activate Call Forward
Control	All Calls
Forward To	The extension line of secretary

2. Secretary on the phone-set press:
Flash (or #) + extension of GM
Then the incoming calls is transfer to GM.
3. Only the secretary's extension is allowed to transfer or dial a call to the boss.

15.1.3 Offnet Forward to Mobile Phone (Offnet PSTN)

15.1.3.1 Offnet Forward to Remote Mobile Phone (PSTN) From Local Machine

User may forward a call from the extension line of this machine to a PSTN line in remote site via a transfer machine. Here is the configuration:

- (1) Configuration for the parameters of "Call Forward" of **This Machine (A)**

<u>Field</u>	<u>Description</u>
Control	All Calls

Forward To	The telephone number of the remote machine (B) that will forward to. If the transfer machine is the machine (A) itself, keep this field in blank
Mobile Number	The telephone number of Trunk (FXO (PSTN) or SIP) that the call will be forwarded to. Because this call will be dialed from the remote machine (B), the phone number must be entered from the point of view of the remote machine (B). The phone number of remote "Forward To" machine (B) needs to be entered for Mobile Number function to PSTN

(2) Configuration for the equipment of **Remote Transfer Machine (B)**

For remote transfer machine, the telephone number, same as the telephone number configured in "Mobile Number" of machine (A), need to be defined in Web Path:

1.System\1.4.Mobile Number.

Example

Own machine (A) VST3300 is located at Taipei and remote machine (B) as the transfer machine is located at Shanghai. If an extension line at Taipei will forward a call and offnet to a mobile phone 1360567888 in Beijing. The configuration for both parties is:

Parameters	Configuration of Machine (A) at Taipei	Configuration of Remote Machine (B) at Shanghai (8621-6445-1111)
Control	All Calls	
Forward to	862164451111	
Mobile Number	1360567888	
Web Path: 1.System\1.4.Mobile Number.		1360567888, can be selected via Selection List

In order to forward the call to remote PSTN line, the "Mobile Number" of machine (A) needs be configured; in addition, the parameters of remote VST3300 also need to be configured. The example is shown below:

Configuration of remote machine (B) (Web Path : **1.System\1.4.Mobile Number**

Apply Cancel

Mobile Numbers

Capacity: 128
Current Entry: 0

[Selection List...](#) Page 1 / 1 Show << >>

Phone Number	Trunk	Attempt	Duration
Add/Modify: <input type="text" value="1360567888"/> <input type="text" value="PSTN"/>			
Delete: <input type="text"/>			
Key In Phone Number: <input type="text"/> Search			

Selection List is useful for machine (B) to select the phone number that can offnet forward to PSTN. Machine (A)'s MAC Address, Channel, Phone Number and Mobile Number input in machine (A)'s channel are shown in the List.

Mobile Numbers Selection List - Microsoft Internet Explorer

Selection List

Capacity: 200
Current Entry: 1

Page 1 / 1 Show << >>

<input type="checkbox"/> MAC Address	Channel	Phone Number	Mobile Number
<input checked="" type="checkbox"/> 00-03-62-80-76-76	02	88628226331112	0921856123

Trunk

Add Selected: Add

Delete All: Delete

Filter Condition

MAC Address: and

Phone Number: and

Mobile Number: Filter

完成 網際網路

Attention:

- 1、The telephone number defined in the field of “Mobile Number” is the number actually dialed from the remote transfer machine. In this example, the call is forwarded to offnet mobile phone, therefore configure it to mobile number and a full number is not required.
- 2、Machine (A) has to be a **VST3300 Series model**, then it can be seen in Selection List

15.2 Hotline Extension

When an extension is configured for hotline, the hotline call route is built when the phone is taken off hook.

Example: Extension of 1012 makes a call, ****88628226336814#** is automatically input when a user picks up the phone set of this extension. (** is access code for IP Call, # is the ending code of IP Call)

Configuration of Example

Web Path : **2.Channel\2.1.Summary**

Select Channel with FXS type and click its St (Status) icon.

CH	St.	Suffix	Type	DND	Fwd	Barring Class	T.38	Statistics In/Out	PickUp/Group	Ext. No.	Voice Mail	Gain In/Out
1		11/OP*	FXS	-	-	0	-	0/0	✓ / 1	1011	-	0/0
2		12	FXS	-	B/C*	0	-	0/0	✓ / 1	1012	-	0/0

Configure Softkey in Status Table

Soft Key	
Soft Key String:	<input type="text" value="**88628226336814#"/>
Trigger Mode:	<input type="text" value="Auto"/>
Trigger Digits:	<input type="text" value=""/>
Append Trigger Digit:	<input type="text" value="Not Append"/>

Parameter	Value	Description
Soft key String	**88628226336814#	Contents of hotline number.
Trigger Mode	Auto	When the system detect a loop on the extension, it triggers the call
Trigger Digits	(Blank)	The digit to trigger the call. When Trigger Mode is Auto, it is blank here.
Append Trigger Digits	Not Append	

IP Call **88628226336814# is made automatically when user of extension 1012 pick up the phone set.

15.3 Line Group

Several extensions can be grouped as a Line Group. When one of the FXS (extension) in the group is busy, incoming call to this FXS can be forward to a pre-configured destination. If the destination FXS port is also busy and has also been configured as Forward (cascade), then this call will be forwarded to the pre-configured destination again. The maximum cascade is up to 16 extensions. The final extension is configured to forward back to the first extension of the Line Group loop created.

Web Path: **2.Channel\2.1.Summary**

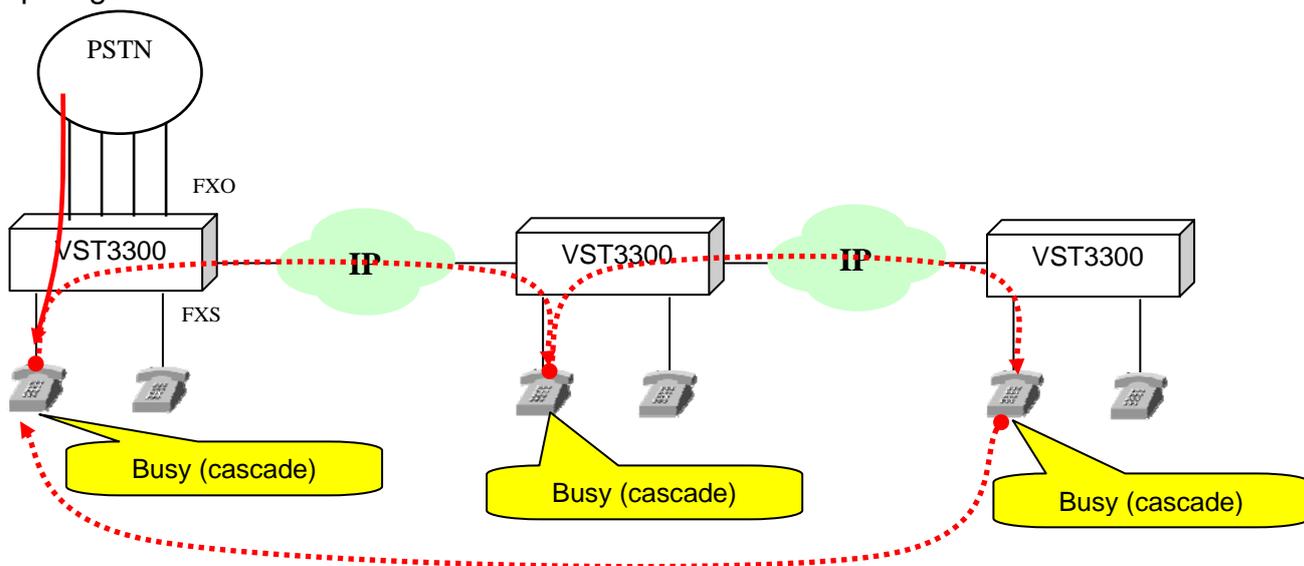
Select Channel with type FXS and click its St (Status) icon.

CH	St.	Suffix	Type	DND	Fwd	Barring Class	T.38	Statistics In/Out	PickUp/Group	Ext. No.	Voice Mail	Gain In/Out
1		11/OP*	FXS	-	-	0	-	0/0	✓ / 1	1011	-	0/0
2		12	FXS	-	B/C*	0	-	0/0	✓ / 1	1012	-	0/0

Configure Call Forward in Status Table. Configure **Forward To** number to the next extension.

Call Forward	
Control:	Busy(cascade) ▼
Forward To:	88628226445712
Mobile Number:	

Example figure:



Attention: Confirming calls between these extensions in the Line Group in advanced is required.

15.4 Call Park

Called side can park the incoming call and anyone using this system can pick up the call anywhere.

15.4.1 Operation Procedure

- A party is the one who triggers Call Park
- B party is the one who has conversation with party A. A will park B on the parking server of this machine.
- C is the third party who retrieves the phone call.

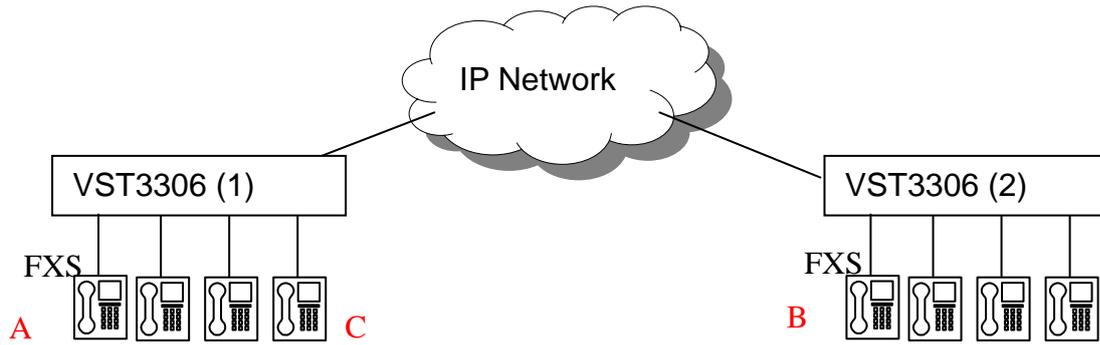
Operation method

- A and B are having a conversation (A calls B, or B calls A)
- A parks B on the Parking Server of this machine, and then any other C party can retrieve the phone call

15.4.2 Analysis of Example

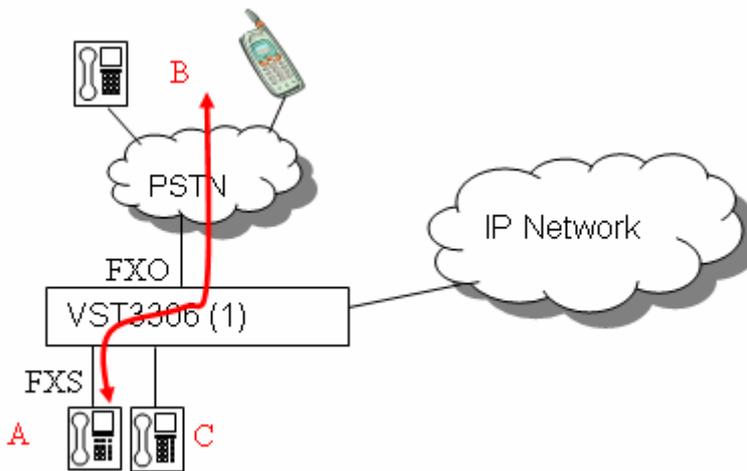
Example 1:

Parties A and C use the extension line of the same VST3300 device. B is the extension of another VST3300 device.



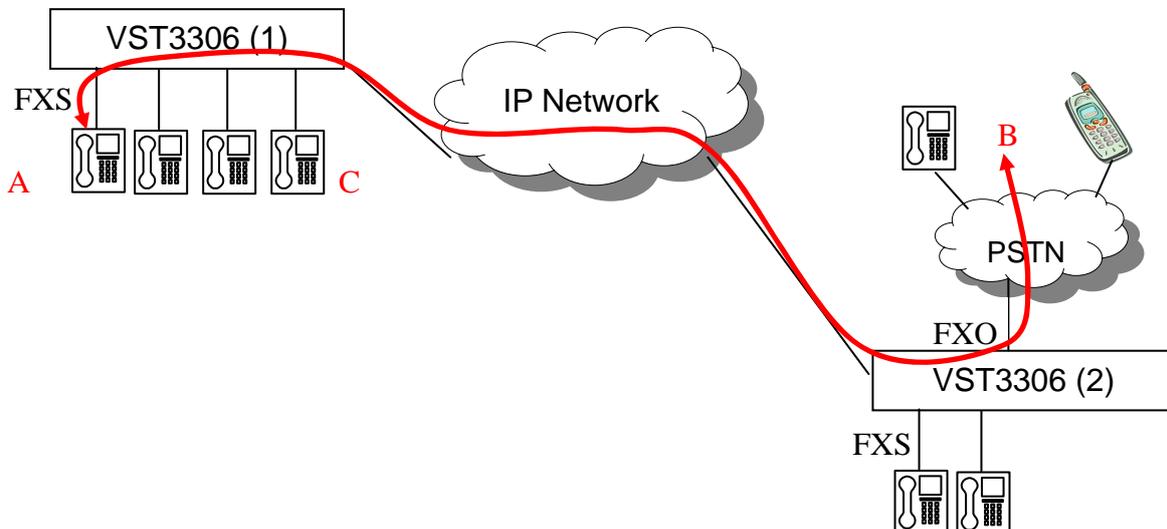
Example 2 :

Parties A, B and C use the same VST3300 device, A and C use extension lines (FXS), B is connected via the PSTN.



Example 3 :

Parties A and C use the same VST3300 device. B is connected via the PSTN (also can be Mobile Phone) of another VST3300.



15.4.3 Configuration Method

Configuration of A party's VST3306 (1):

- ◆ Configure a Prefix/Ext. No: 5555 in A party's VST3306 (1). Phone Number can be configured as the phone number (My Phone Number found in Web Path: 1.System\ Basic Information) of VST3306 (1) and Type as Phone. This Prefix/Ext. is identified as Park ID of VST3306 (1)'s Parking Server

Web Path : **1.System \1.2.Extension Number**

	Prefix/Ext. (Max.6 digits)	Phone Number	Type
Add/Modify:	5555	886218418934	Phone

Park ID can be divided into few types below

- The Prefix/Ext. of VST3300 can configure Prefix/Ext. No as the device number. This number is identified as Park ID. Prefix/Ext. No + Ext. suffix can also be a Park ID and its type is iPBX
- ◆ Configure related configuration that make A and B able to communicate.

Configuration of party B 's VST3306 (2):

- ◆ Configure related configuration that make A and B are able to communicate.

Configuration of party C's VST3306 (3):

- ◆ C must use the extension of VST3300 Series device that is connected to IP network. Lines from PSTN do not work. C and A must use the extension of the same box, so further configuration is not required.

15.4.4 Practice

Method 1 :

1. A calls B. Conversation starts.
2. A press Flash+5555*, B is parked on Parking Server of this machine.
3. Any extension line of this machine, such as C, presses *1 5555, and then it is able to talk with B.

Method 2 :

1. A calls B or B calls A, and conversation starts.
2. A press Flash+3102 13*, B is parked to Parking Server. (3102 is the Prefix/Ext.

of A's VST3306 Plus (1). It needs to be configured in advance. 13 is the extension suffix)

3. The extension on the same IP-PBX box, C press *1 3102 13, and then C is able to talk with B.

If multiple parties A all use the same Park ID, it is possible that many parties B are parked with the same Park ID. The retrieve sequence is the same as the parking sequence, no matter how many C parties they have.

16. Connect to Special Device (Seize Specific Trunk)

16.1 Connect Door Phone and Voice Broadcast System

You can connect this machine to some non-telecom analog devices. These devices supply analog current and it is similar to the trunk line of a PSTN. Open or close the circuit can open or close the device at remote place, such as

- Door Phone
- some kinds of voice broadcast systems

For these devices, specify a FXO port to connect to door phone or broadcasting system. A user of the VST3300 Series can Seize Specific Trunk, then door lock is open or voice broadcast system is seize for voice broadcast.

Description:

- Trunk Class ID (0~11) may be assigned to every FXO port; default value is 0.
- By defining Trunk Class, the specific FXO port can be accessed by the remote machine.
- Dial Method:
<* 8> + <Prefix/Ext.> +< Class ID (0~11)> +<*/#> + (PSTN number) Note: If no PSTN number is dialed, dial tone from PSTN is heard
- Note: If there are several FXO ports have the same Trunk Class ID, the access sequence is from the last port upward.

Configuration:

- Web Path : **2.Channel\2.1.Summary**

Select Channel with PSTN (FXO) Type and click its St (Status) icon.

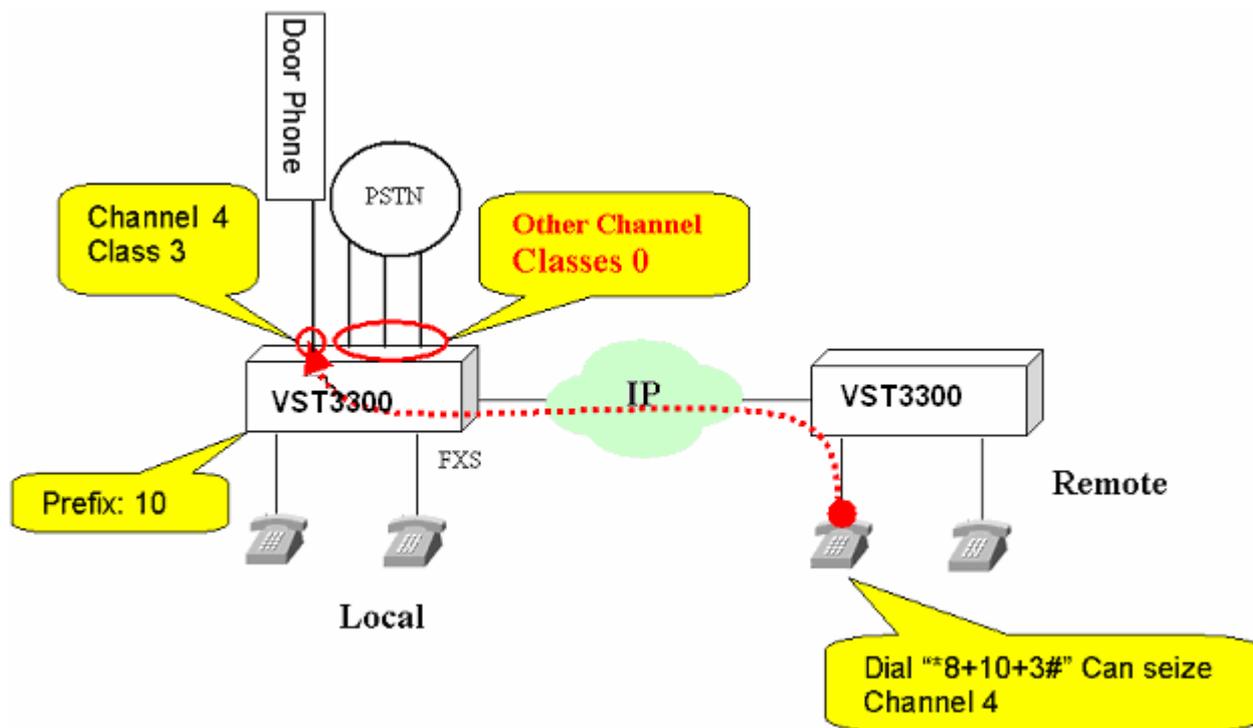
Ch	St	Access	Type	Error Count	Fwd	Trunk Class	T.38	Statistics In/Out	Last Number Dialed	Gain In/Out
3		9	PSTN	0	-	0	✓	0/0	-	0/0
4		9	PSTN	0	-	0	✓	0/0	-	0/0

Configure Trunk Class in Status Table, such as "3:Particular App.(1)". Click **Apply**.

Trunk Class

ID:

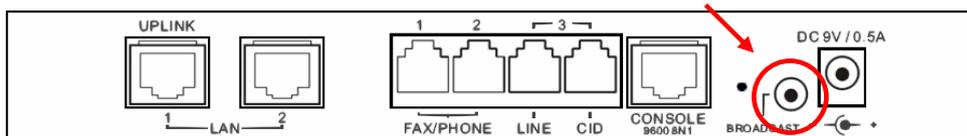
Example Figure:



Note: The default Prefix of Extension is 10 and access code of Seize Specific Trunk is *8

16.2 Connecting a Voice Broadcast System (VST3305 Only)

In addition to general channel, VST3305 has an extra Broadcast port.



This port can connect to voice amplifier for broadcasting to company

To use this function, seize specific trunk method is used. When the specific trunk is seized, personnel can talk into phone set directly for broadcasting.

Method: Pick up phone set and dial *8 10 11 (#). Start to broadcast when hint tone is heard. (*8: access code of Seize Specific Trunk, 10: Extension Prefix of this machine, 11: Trunk Class ID, #: dial ending code)

This function uses a fixed Channel and can not be changed.

Web Path: **2.Channel\2.1.Summary**

Ch	St	Access	Type	Error Count	Fwd	Trunk Class	T.38	Statistics In/Out	Last Number Dialed	Gain In/Out
3		9	FXO	0	-	2	✓	0/8	808	0/0
4		*810	SPK	0	-	11	-	0/0	-	0/0

17. Firmware Update and Backup

17.1 File Types

VST3300 Series files, each have their own meaning. Here lists its meaning:

File Name	File Type	Description
XP4421.CFG XP4423.CFG XP44XX.CFG	System configuration file	File of system configuration
XP44XX.GT1	1 st greeting file	File of voice greeting record
XP44XX.GT2	2 nd greeting file	File of voice greeting record
XP44XX.GT3	3 rd greeting file	File of voice greeting record
XP44XX.GT4	4 th greeting file	File of voice greeting record
XP44XX.GT5	5 th greeting file	File of voice greeting record
XP44XX.GT6	6 th greeting file	File of voice greeting record
XP44XX.GT7	7 th greeting file	File of voice greeting record
XP44XX.GT8	8 th greeting file	File of voice greeting record
XP44XX.VON	System voice file	Voice file for announcement
XP4421.RUN XP4423.RUN XP44XX.RUN	Executing file	System Software
XP4421.WEB XP4423.WEB XP44XX.WEB	Web page	Page for web browser

XP44XX.MEM	Text file	MEM setting file can be downloaded by FTP to PC; open file and modify the contents using NOTEPAD or other word processing tool; then uploaded the file to system.
COLDSTART	Cold start	It is a pseudo file. VST3300 Series will execute the cold start if this file is deleted via FTP. It is a convenient function if cold start is required after a FTP software update.
WARMSTART	Warm start	It is a pseudo file. VST3300 will execute the warm start if this file is deleted via FTP. It is a convenient function when using FTP.
CP_TONE.BIN	CP binary tone file	CP binary tone file
DSP_CORE.RUN	Run DSP file	DSP execution file

17.2 Update and Backup Firmware via Web

VST3300 Series can update or backup firmware via the Web management page.

17.2.1 Firmware Update

Web Path : **7.File Transfer**

1. Click "Browse"

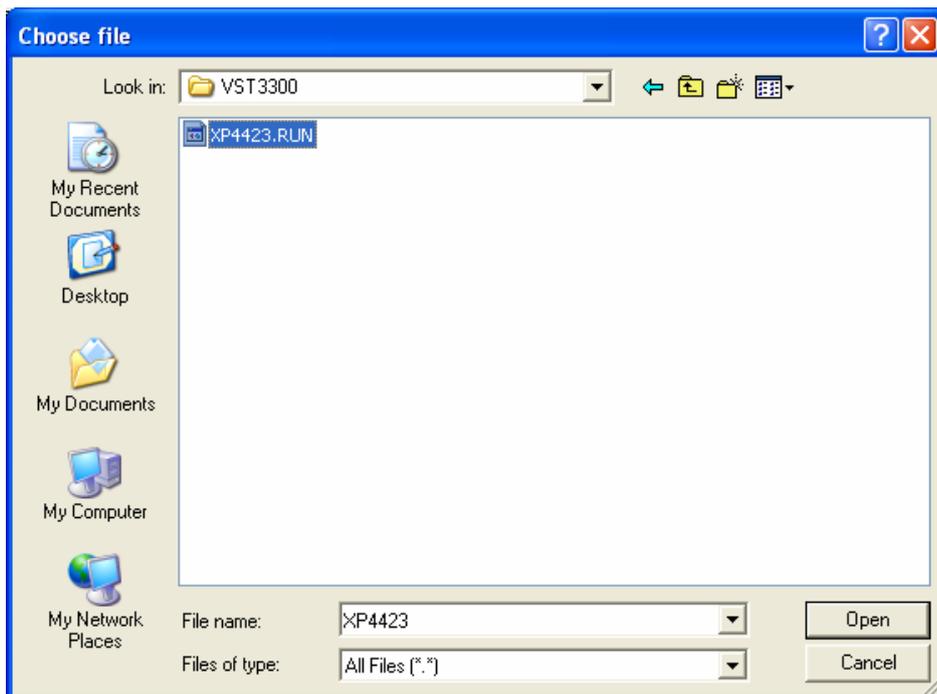
Put File from PC to this Device

Select File: Browse... Clear

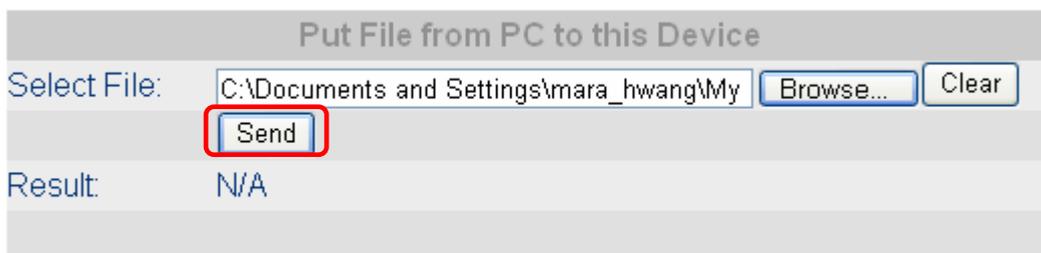
Send

Result: N/A Browse

2. Select "XP4423.RUN" in your PC.

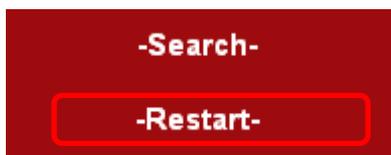


3. Click Open



4. Click **Send**. Wait for the Result to give a successful message and then do a restart to this machine

Web Path: Click -Restart- at left side of the Web page.



17.2.2 Backup Firmware

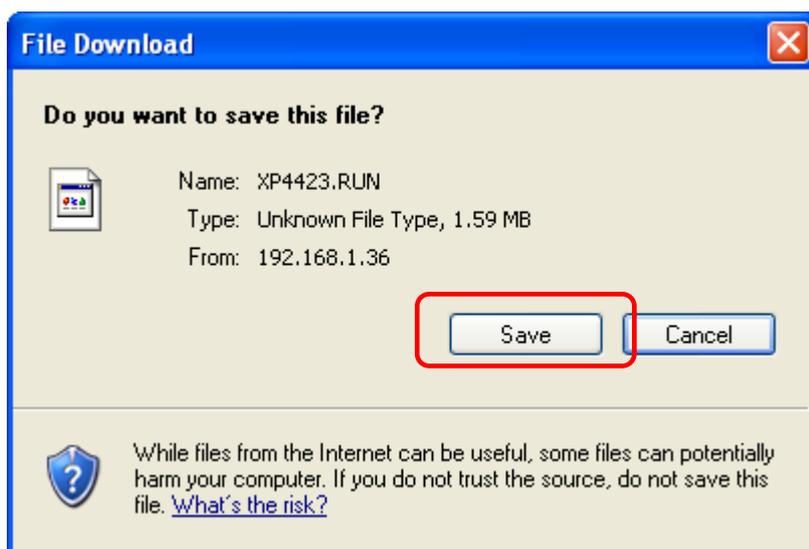
Web Path: **7.File Transfer**

1. Click the download icon "Get".

Get File from this Device to PC				
File Name	Size	Date	Time	Get
XP4423.RUN	1670812 Bytes	2006/03/14	17:31:00	
XP4423.CFG	131152 Bytes	2006/03/20	11:12:00	
XP44XX.GT1	22100 Bytes	2000/01/01	00:21:37	
XP44XX.GT2	7360 Bytes	2000/01/01	00:08:50	
XP44XX.GT3	5424 Bytes	2000/01/01	00:09:10	
XP44XX.GT4	3708 Bytes	2000/01/01	00:09:30	
XP44XX.GT5	8240 Bytes	2000/01/01	00:10:45	
XP44XX.GT6	8108 Bytes	2000/01/01	00:11:05	
XP44XX.GT7	8020 Bytes	2000/01/01	00:12:13	
XP44XX.GT8	65536 Bytes	2000/01/02	00:57:59	
XP44XX.MEM	164 Bytes	2006/03/20	11:12:00	
XP4423.WEB	67660 Bytes	2006/02/24	16:15:00	

[\(Sample MEM File\)](#)

2. Click "Save File"



3. Select the save file path of PC and then save the file

17.2.3 Application of .MEM file

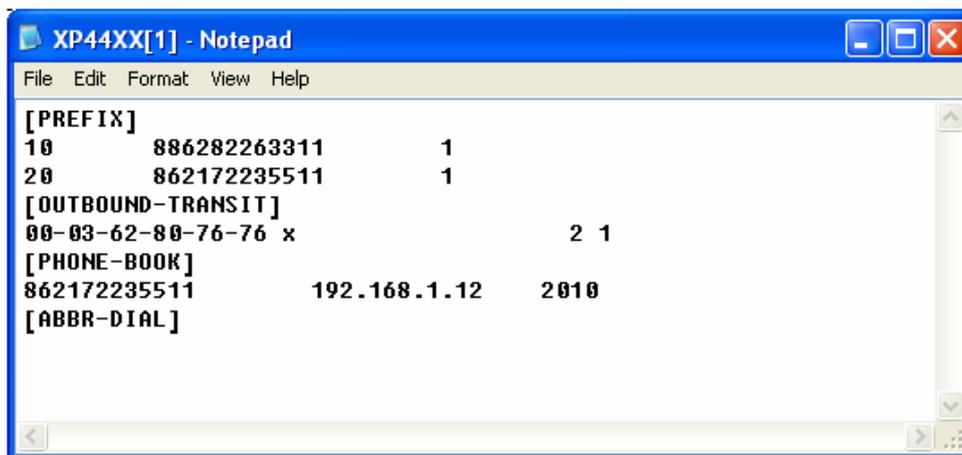
A .MEM file records lots of customized data which the user creates from the Web. MEM files can be downloaded from the devices. Open the file in Notepad or other word processing software, modify the data and then upload the file back to the device. If there is a lot of data to be created or modified this method saves a lot of time.

Find a data file .MEM in this machine. It is a data file of many items that can be created by Notepad:

Web Path : **7.File Transfer**

Get File from this Device to PC				
File Name	Size	Date	Time	Get
XP4423.RUN	1663024 Bytes	2006/01/23	11:07:00	
XP4423.CFG	131152 Bytes	2006/01/26	16:09:00	
XP44XX.GT1	22100 Bytes	2000/01/01	00:21:37	
XP44XX.GT2	10396 Bytes	2004/01/30	18:04:00	
XP44XX.GT3	8372 Bytes	2004/01/30	18:06:11	
XP44XX.GT4	4852 Bytes	2004/01/30	18:07:07	
XP44XX.GT5	6876 Bytes	2004/01/30	18:09:51	
XP44XX.GT6	10704 Bytes	2004/01/30	18:11:22	
XP44XX.GT7	11320 Bytes	2004/03/10	09:12:42	
XP44XX.GT8	45244 Bytes	2004/09/15	14:14:52	
XP44XX.MEM	175 Bytes	2006/01/26	16:09:00	
XP4423.WEB	73228 Bytes	2006/01/19	15:06:00	

Download the .MEM file to PC and open it in Notepad. See example below.



Data can then be created by your self. To see the format of data entries, please refer to Web Path: **7.File Transfer**, at bottom line

(Sample MEM File)

Click Sample MEM File, you can see the table below. Create the data entries according to this table to speed up installation.

[PREFIX]			
# format:	Prefix	phone number	Type(0:phone/1:iPbx/2:Conference)
	111	12345678	0
	112	88888888	1

[OUTBOUND-TRANSIT]				
# format:	MAC	phone number	Route Type (1:Local/2:Toll/3:specified)	Allowed(0:No/1:Yes)
	00-03-62-11-22-33	12345678	1	0
	00-03-62-12-34-56	88888888	2	1

[PHONE-BOOK]			
# format:	phone number	IP Address	port
	12345678	192.168.0.1	2000
	88888888	192.168.0.2	2000

[ABBR-DIAL]		
# format:	index	phone number
	00	12345678
	01	88888888

[SIP-OUTBOUND-AUTH]				
# format:	entity(0 for all)	"realm"	"username"	"password"
	1	"realmA"	"realmA_user"	"0000"
	2	"realmB"	"realmB_user"	"1111"

[SIP-INBOUND-AUTH]			
# format:	entity(0 for all)	"username"	"password"
	1	"1010"	"1010"
	2	"1011"	"1011"

[SIP-PHONE-BOOK]					
# format:	index	"user_part"	"host_part"	port	via_proxy(0:No/1:Yes)
	8888	"user"	"registrar.com"	5060	0

After modification is done, upload it back to this machine, then the data on this machine will be updated.

Note: The machine had already kept all of the data before you download the .MEM file. So the data is not removed if you remove entries (all or partial) of the .MEM file and upload to machine again.

If you need to remove any records, you should.

- Remove record from Web page

Notice of MEM File Update

After the configuration is finished, please make a backup file for CFG file. It is in case that if the data is lost, you may upload the backup file of CFG file to this machine. If you upload the previous backup file of CFG file to machine after the MEM file is uploaded, the MEM file will be ineffective because the backup file will overwrite the MEM file. You have to re-upload the updated MEM file to this machine to get the correct data.

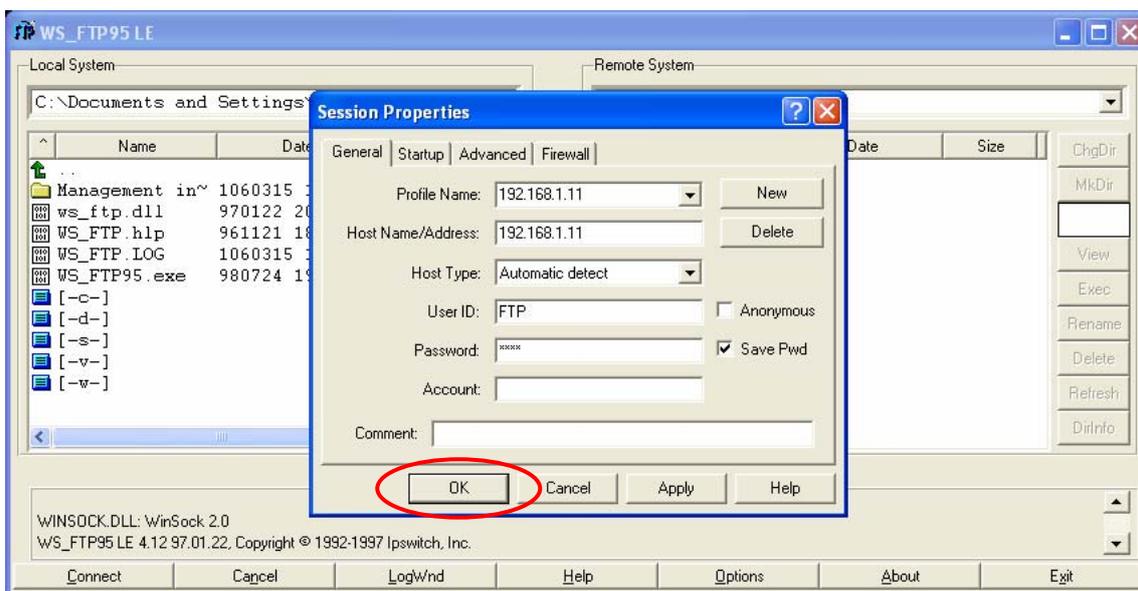
17.3 Update and Backup Firmware via FTP

Firmware update and backup can also be done by a FTP Client

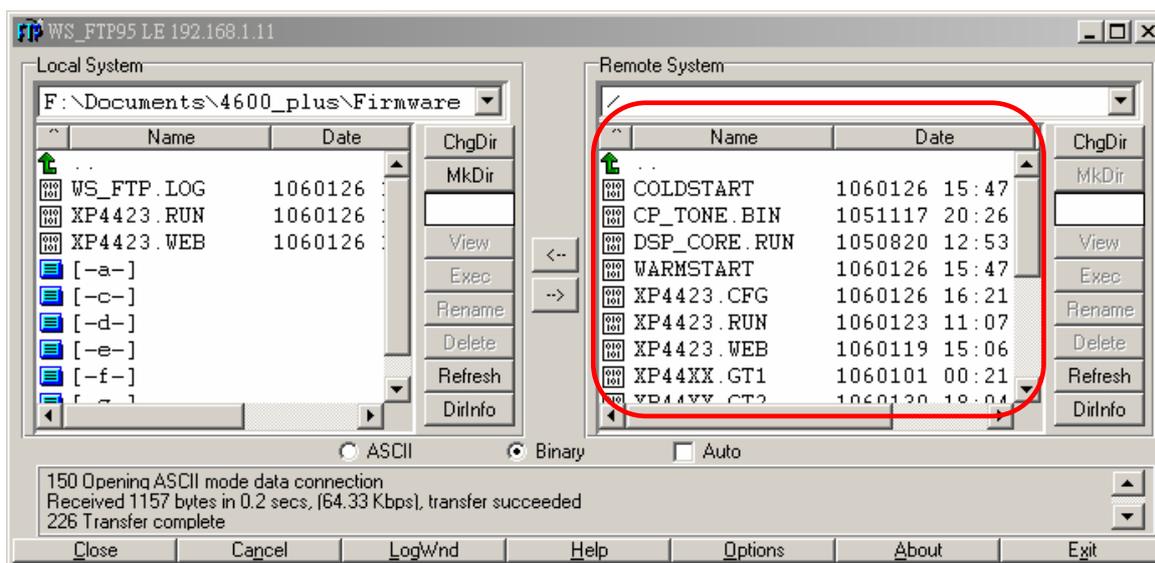
17.3.1 Firmware Update by FTP Software

17.3.1.1 Update RUN and WEB by FTP Software

1. Open a FTP Client Software, such as CuteFTP or WS FTP. Enter IP Address, User Name (default: FTP), Password (FTP and Console password are same by default: blank), and the Fire Wall Port Number to 21

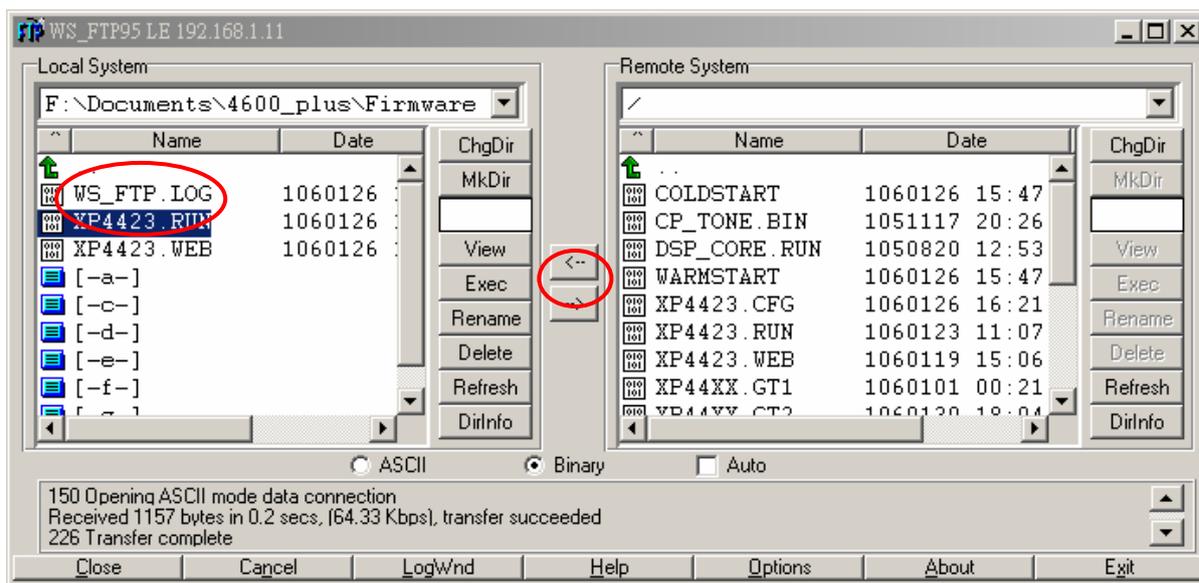


2. Click **Connect** button to get a connection between this machine and the FTP Client. The files of this machine will be displayed if the connection is successful.



3. Be sure that the files to be uploaded are stored on the hard disk. Select the file with extension of .RUN and click the → button. (Note: the file name must be the

same as the file name in the IP-PBX, e.g. XP4423.RUN).



4. After the file is overwritten (check by looking at file's update time), Select the file with extension of .WEB and click → (Note: the file name must be same as the file name in the Gateway, e.g. XP4423.WEB). The step is the same as 3 and 4
5. Do a Cold Restart to finish the Update procedure (this procedure saves and processes all the settings).

17.4 Check Updated Version

Check if the update is successful or not: Enter Web page to verify

Web Path: **1.System\1.1.Basic Information**

Information	
Region ID:	0 (Taiwan)
Software Version:	1.00.2
BootRom Version:	1.01
Hardware Version:	2.00
Module Type:	4 PORT_FXS
SIP Line Capacity:	72
SIP Trunk Capacity:	24
System Up Time:	2 day 14 hr 11 min 55 sec
MAC Address:	00-03-62-80-77-90
Location:	

A red box highlights the 'Software Version' field (1.00.2) with an arrow pointing to a red box containing the text 'Check if the version correct'.

18. Appendix

18.1 Management by System Console and Telnet

18.1.1 List of all commands

A command may have many sub-commands. Input "?" and press Enter for further help.

User Exec
commands :

Enable	Turns on privileged commands
Exit	Exits from the EXEC
Help	Description of the interactive help system
Show	Shows running system information

show :

dns	Shows the IP address of domain name server
ethernet	Fast Ethernet port status and configuration
history	Displays the session command history
ip	Displays IP configuration
log	Shows log filter control
manager	Shows status of management functions
outbound-auto-sync	Shows outbound auto sync
route	Shows route
running-config	Shows current operating configuration
service_port	Shows service port configuration
version	System hardware and software status

Privileged Mode :

Configure	Enters configuration mode
Delete	Resets configuration
Disable	Turns off privileged commands
Exit	Exits from the EXEC
Help	Description of the interactive help system
Ping	Sends echo request to destination
Probe-hook	Probes busy tone cadence
Probe-remove	Stops probe busy tone cadence
Reload	Halts and perform cold start
Restart	Halts and perform warm start
Show	Shows running system information

Global Mode :

Dbflush	DataBase flush
Dns	Sets the IP address of domain name server
End	Exits from configure mode to privileged mode

Exit	Exits from configure mode
Help	Description of the interactive help system
Ip	Global IP configuration subcommands
Log	Control log output
Manager	Enables/Disables the specific management function, including Enable/Disable management of FTP, Telnet and Web.
No	Negates a command or sets its defaults
Outbound-auto-sync	Automatic sync remote outbound entries
Password	Modifies password of enables command
region_id	Sets regional id
service_port	Sets service port number
terminate	Forces channel clear down

18.2 Management by Phone set

There are two levels of management, extension line user and system administrator.

18.2.1 User of Extension Line

Every extension line has a password. Take the phone off hook and at the dial tone, dial *0, **or press the button on the rear panel of machine and then make a call to FXO port of this machine (no *0 or password is required)** and hear the tone of “DuDu”. At this moment you may configure the parameters by using the following instructions.

Attention: button on the rear panel of the machine is for maintenance purposes and do not push it under normal conditions.

Item	Description	Parameter	Remarks
01	Call Forward	0 / 1 / 2 / 3 / 4 / 5 0 : Disable 1 : All Calls 2 : Busy 3 : No Answer 4 : Busy or No Answer 5 : Busy (Cascade, Administrator Only)	Configures “Call Forward” type. Only Forward All Calls take effect if call are from FXO
02	Telephone number for Call Forward	1 ~ 19 Digits	Defines the telephone number of Call Forward
03	Telephone number for Offnet To	1 ~ 22 Digits	Defines the telephone number for Offnet To

Item	Description	Parameter	Remarks
04	Clock Alarm	hh*mm*c: hh: 00~23 Hours mm: 00~59 Minutes c: 1 : one time 2 : periodic Blank: clears the setting for periodic alarm	Phone will ring 3 times when the time is up. You can set alarm start once or periodically
05	Voice Gain	0 : Default 1 : +2 db 2/22/222 : -2/ -4/ -6 db	The Input Gain and Output Gain will be changed
06	Change Password	4 Digits	Default is no password. You may set or change password
07	Activate the Greetings of night mode	0 / 1 0 : Disable 1 : Enable	Only Operator port can change the configuration
08	Pick up is allowed	0 / 1 0 : Disable 1 : Enable	The other line can not pick up calls for you unless this function is enable
09	Enter to system management mode	4 Digits	Refer to section 10.4.2 System Management
10	Play current time	2 Digits	Plays the current time
11	Display Caller ID	0 / 1 0 : Disable 1 : Enable	Displays caller ID or not
12	Do not disturb (DND)	0 / 1 0 : Disable 1 : Enable	Configures DND function. Enable it allows to dial call from that extension, but block all call dial to that extension
13	Clear voice mail waiting tone		Clears waiting tone for voice mail retrieval (DuDuDu dial tone is heard when voice mail has not been retrieved)
21	Read Phone Number of this machine		Press * or # to leave
22	Read Community IP Phone number of this machine		Press * or # to leave
23	Read Prefix Number of this machine		Press * or # to leave

18.2.2 System Administrator

The system Administrator has a special password (the default password is 9999). You can take the phone off hook and dial *0 after the dial tone is heard or **press the button on**

the rear panel of machine and then make a call to FXO port of this machine (no *0 or password is required), you will hear the tone of “Du ..Du..” then dial 09 9999#, then hear the tone of “Du..Du..” again. Now, the system management can dial the following item numbers for management.

Attention: the button on the rear panel of the machine is for maintenance purposes. Don't push it under normal conditions.

Item	Description	Parameter	Remarks
40	Read internal IP Address		If under NAT, access to the current internal IP address
41	Read Subnet Mask		If under NAT, access to the current Subnet Mask
42	Read Default Gateway		If under NAT, access to the current Default Gateway
43	Read Signaling Port		If under NAT, access to the current UDP Port
44	Read the Global IP Address		Access to the current Global IP Address
45	Read Global Signaling Port		Access to the current Global UDP Port
50	Define Area Code	1~3 Digits ; from 1 to 999	Defines the Area Code that the system is allocated
51	Define Phone Number	1~19 Digits of 0~9	Defines the telephone number of the equipment
52	Define PSTN Call DISA Control	0 / 1 0 : Disable 1 : Enable	If DISA answers the PSTN call
53	Define IP Call DISA Control	0 / 1 0 : Disable 1 : Enable	If DISA answers the IP call
54	Set IP Status	0 / 1 / 2 0 : Manual 1 : DHCP	Configures the method to get the IP Address
55	Define IP Address	x : 1 ~ 3 Digits ; 0~255 xxx*xxx*xxx*xxx	Defines IP Address of own equipment
56	Define Subnet Mask	x : 1 ~ 3 Digits ; 0~255 xxx*xxx*xxx*xxx	Defines Subnet Mask of own equipment
57	Define Default Gateway	x : 1 ~ 3 Digits ; 0~255 xxx*xxx*xxx*xxx	Defines default Gateway of own equipment
58	Define Primary DNS Server IP	x : 1 ~ 3 Digits ; 0~255 xxx*xxx*xxx*xxx	Defines Primary DNS Server IP of own equipment

Item	Description	Parameter	Remarks
59	Define Secondary DNS Server IP	x : 1 ~ 3 Digits ; 0~255 xxx*xxx*xxx*xxx	Defines Secondary DNS Server IP of own equipment
60	Define Dial Ending Time	1 Digit ; from 0 to 9	It is dial ending if no digits are dialed before dial ending time out. Default is 0 second and the dial must be ended by #. If the dial ending time is defined and the dial is not ended by #, system will wait until dial ending time out.
61	Change Service Port	1 : FTP , 2 : HTTP , 3 : Telnet 0~65535	Can configure the port number of three kind services.
62	Remote Management Control (TCP)	0/1 0 : Disable 1 : Enable	Enables or Disables Remote Management Control (FTP, Telnet and HTTP (Web))
63	VAD control	0/1 0 : Disable 1 : Enable	Default value of VAD is Enable. When it is enabled, the machine uses G.729AB codec. This Codec use less bandwidth. If this machine connects to other machine that use G.729A codec only, please Disable it. Then this machine use G.729A codec to communicate with others (use more bandwidth)
91	Not restricted by Barring Table		You will hear the dial tone again after dialing "91". Any numbers dial after above process is not restricted by barring table. System manager uses this function for checking and maintenance.
92	Reset the password of individual extension line	2 Digits ; from 11 to 26	When the user forgot the password, user can ask the system manager to reset the password to default value (blank)
93	Define an extension line as Operator	2 Digits ; from 11 to 26	Enters the extension number that will be Operator
94	Configure password of phone set programming	4 Digits ; 0~9	Configures password of phone set programming

Item	Description	Parameter	Remarks
	mode		mode for this machine
95	Configure Region ID	2 Digits ; 00~99	sets a different Region ID
96	Play the greetings recode	1 Digit ; from 1 to 7 # stop	Enters the number of greeting to be played
97	Reset to recover all parameters to default value	1 / 2 1 : recovered by default value 2 : recovered by default value except IP	Returns all parameters to default value
98	Warm Restart	1 / 2 1: Warm restart 2: Cold restart	Executes Restart
99	Record Greeting records	1 Digit, from 1 to 8	Records the voice record of greetings, total 8 voice records