IP PBX SETUP INSTRUCTIONS
Connect the IP PBX to Your LAN

Internet

Router

Ethernet Switch

FXO Ports

PSTN
Access the PBX’s WEB GUI

• The default IP address of WAN is 192.168.1.100
• If the PBX’s IP address is different from your LAN IP address, you need to manually change IP address of your PC to 192.168.1.101, then connect both PBX and PC to the same LAN
  user name: admin
  password: 1234
Built-in Web Server Provisioning Interface

ONLY Firefox web browser is fully supported now
How to Change Configuration of PBX

When any changes made, you need to save the change and apply the changes so it can be effective immediately.
Setup the IP PBX

• Setup Users
• Setup out bound call
• Setup inbound call
• Setup switch board
• Setup other features
  – Audio Conference
  – Call queue
  – Other PBX Features
• Update PBX Firmware
Setup Users

• The default users are from 2000 – 2049
• You can bulk add users or add individual user
• Setup user specific settings
  – SIP password
  – Voicemail password – usually the extension number
  – DTMF mode – out band RFC2833 or in band audio
  – Enable Re-invite for the users in the same LAN with IP PBX, disable Re-invite for the remote users
  – Choose dial plan for users (after the outbound call setup)
  – User defined call forward, find-me, follow-me services
• Assign extensions to the FXS ports
# User Management

Select one or more users to configure

Click EDIT to change configurations or DELETE to delete users
Each configuration item has online tip notification to assist user.
Bulk Add Users

Specify the starting extension and number of continuous extensions to be added.
User Follow me Configuration

Configure follow me features on per extension basis. Can ring number one by one or simultaneously.
Setup Outbound Call

- Configure the FXO hardware in system-hardware to your regional setting, e.g. if you are in US choose country to US
- Configure the FXO trunk
  - each trunk has several FXO ports, the ports in each trunk share the same calling rule
  - Setup busy detection ON and polarity switch ON so PBX can correctly hang up the call
  - Setup gain for each FXO port
- Setup outgoing calling rule for each trunk
- Dial Plan
  - Prefix based number routing
  - Dialing out number policies
  - User group based calling rights control
- SIP Trunking
  - Can setup unlimited SIP trunk to the VoIP service providers.
FXO Hardware Parameter Setup

Digital Hardware

No Digital Hardware detected

Analog Hardware

<table>
<thead>
<tr>
<th>Type</th>
<th>Ports</th>
</tr>
</thead>
<tbody>
<tr>
<td>FXS Ports</td>
<td>--</td>
</tr>
<tr>
<td>FXO Ports</td>
<td>1, 2, 3, 4, 5, 6, 7, 8</td>
</tr>
</tbody>
</table>

Advanced Settings

Module Name: wctdm24xkx

Opermode: PHILIPPINES

a-law override

FXO port information, you can also change the country settings to match your local network.
FXO Hang-up Condition Detection Setup

FXO related information can be setup in trunk menu.

FXO support busy detection or polarity reversal to determine the hang up condition.
Outbound Call Control – Define FXO trunk

Define the trunk features, most importantly you should decide how FXO decides the hang up condition, use polarity reversal or busy tone detection. If busy detection is enabled, the busy tone frequency and cadence has to be set to the country standard where the EVS3080 is deployed.

Give trunk a name and select the FXO ports should be included in the trunk.
For each SIP or FXO trunk, you need to specify a calling rule which defines what calling prefix should be used to access the trunk.

You can also specify a fall over trunk when the trunk is unavailable due to busy or other conditions, the call can be made through fall over trunk.
Outbound Call Control – Define Dialplan

A dialplan is the set of calling rules and special features that user can use. Each user can have its own dialplan setup.
Setup SIP Trunk

Click New SIP trunk to add outgoing sip trunk
Setup SIP Trunk Setup

Service providers’s domain name or IP Address, default port is 5060 or use :xxxx to specify

PBX’ B’s SIP registration name and password

SIP trunk options
## SIP Trunk Status

### System Status

Uptime: 03:01:44 up 52 min, load average: 0.74, 0.31, 0.11

### Trunks

<table>
<thead>
<tr>
<th>Status</th>
<th>Trunk</th>
<th>Type</th>
<th>Username</th>
<th>Port/Hostname/IP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Unregistered</td>
<td>Vonage Business Trunk</td>
<td>sip</td>
<td>test</td>
<td>sip.vonage.com</td>
</tr>
<tr>
<td></td>
<td>pstn trunk</td>
<td>Analog</td>
<td></td>
<td>Ports 1,2,3,4,5,6,7,8</td>
</tr>
</tbody>
</table>

#### Conference Rooms

<table>
<thead>
<tr>
<th>Room</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>505</td>
<td>Not in use</td>
</tr>
<tr>
<td>502</td>
<td>Not in use</td>
</tr>
<tr>
<td>500</td>
<td>Not in use</td>
</tr>
</tbody>
</table>

#### Extensions

<table>
<thead>
<tr>
<th>Extension</th>
<th>Name/Label</th>
<th>Status</th>
<th>Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>2000</td>
<td>User 2000</td>
<td>Messages 1/0</td>
<td>SIP User</td>
</tr>
<tr>
<td>2001</td>
<td>User 2001</td>
<td>Messages 0/0</td>
<td>SIP User</td>
</tr>
</tbody>
</table>
Inbound Call Control

• Multiple Inbound Trunking Interface Options
  – Integrated PSTN analog FXO gateway, configurable 2/4/8 FXO ports
  – Integrated unlimited VoIP trunking interface seamlessly connected with multiple VoIP service provider, with call in/call out.
  – Integrated SIP peering capability enable distributed EVS3080 deployment
  – Enable enterprise with multiple local numbers to more conveniently connected to customers

• Powerful Number Based Call Routing
  – Redirect to auto attendant
  – Redirect to extension number
  – Redirect to voice mail
  – Redirect to call queue
  – Redirect to audio conference room
  – Redirect to voice mail recording
Inbound Call Control

• Each incoming call rules apply to each FXO/SIP trunk
  – Call can be directed to different extensions
  – Call can be directed to auto attendant or other voice menus
  – Call can be directed to conference room or call queue

• Time based rule can be defined for each incoming call rule
Inbound Call Control

Define how call should be directed, can be call queue, conference, voicemail or other special routine.
You can also define callee or caller number based matching.
Switchboard Setup

• Auto attendant
  – Upload voice greetings or record live through phone
  – Can specify the time triggered auto attendant
  – Can be a number or voice prompts or customizable IVR

• Operators
  – Can specify multiple operator numbers
  – Ring together or sequentially ring within operator groups
  – Support blind or non-blind call transfer
  – If the transfer failed, the call can be directed to operator again

• Calling Status Monitoring and Control
  – Can install web based calling status monitoring and control application
Switchboard Setup

Use switch board options to define voice menu, ring group, upload voice prompt files, define user directory and time based routing rules.
# GUI Based IVR Menu

### Menu Details:

<table>
<thead>
<tr>
<th>Name</th>
<th>AA</th>
</tr>
</thead>
<tbody>
<tr>
<td>Extension</td>
<td>013</td>
</tr>
<tr>
<td>Actions</td>
<td></td>
</tr>
<tr>
<td>Answer the call</td>
<td></td>
</tr>
<tr>
<td>Set Music-On-Hold class: 'Default'</td>
<td></td>
</tr>
<tr>
<td>Play demo: Answer &amp; listen for KeyPress events</td>
<td></td>
</tr>
<tr>
<td>Wait 10 sec for the user to enter an extension</td>
<td></td>
</tr>
<tr>
<td>Goto RingGroup Operator</td>
<td></td>
</tr>
</tbody>
</table>

### Add new Step:

- **Allow KeyPress Events**
- **Goto RingGroup Operator**
- **Goto directory, 118, 1**
- **_ _ _**
- **_ _ _**
- **_ _ _**
- **_ _ _**
- **_ _ _**
- **_ _ _**
- **_ _ _**
- **_ _ _**
- **_ _ _**
- **_ _ _**
- **Select an Option**
- **Goto RingGroup Operator**

### Options:

- [ ] **Cancel**
- [ ] **Save**
PBX Feature

• Call Forward
  – Individual setup of the call forward on conditions

• Call transfer
  – A calls B, B push ‘TRANSFER’ button, dial C’s number and push ‘SEND’ or ‘#’, A and C can talk

• Call Park
  – A calls B, B transfer the call to 700 and hang up. B goes to another location and dial the 700 to retrieve the call

• Call Pickup
  – A calls B, B and C are in the same user group, C can get the call by dial “*8”

• Call Hold
  – A calls B, B push ‘HOLD’ button and dial C’s number, B can switch between the call to B and C by push ‘HOLD’
  – A can push ‘CONFERENCE’ to start conference call
Configure the Call Park Features

<table>
<thead>
<tr>
<th>Feature Codes</th>
<th>Call Parking</th>
<th>Application Map</th>
<th>Dial Options</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Call Parking Preferences</strong></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Extension to Dial to Park a call:</td>
<td>700</td>
<td></td>
<td></td>
</tr>
<tr>
<td>What extensions to park calls on:</td>
<td>701-720 (Ex: '701-720')</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Number of seconds a call can be parked for:</td>
<td>120</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

[CANCEL] [SAVE]
Calling Queue—Automatic Call Distribution

- Customizable queue number
- Priority based operator routing, Ring Together or Roundrobin
- Configurable Voice Greetings with prompt files upgrade capability
- Can have customized IVR for each queue
Calling Queue Setup

Define the extension of the call queue, ring strategy, background music.
Other Functions

• Configurable Music-on-hold
  – Allows selection among multiple music files uploaded from PC
  – Audio convert tools for converting mp3 files to music RAW files

• Unified Messaging
  – Dedicated voice mail for each individual user
  – USB flash based disk or hard disk based massive storage
  – Support message waiting indication message
  – Automatically forward voice mail to user specified email account

• IP Broadcasting
  – Dial 1800, all the users in group 0 will hear the broadcast message
Music-on-hold Setup

Use tftp or http to upload music files to the PBX

Here is the IP address of the tftp server, in tftp server please configure the tftp server point to the voice/music file.
Audio Conference Features

- Two built in conference room with 10 participants each
- Customizable access number, VoIP or PSTN
- 5 Ways to dial into conference room
  - Call from PSTN or VoIP directly to the conference room
  - User dial the conference room extension
  - Transferred from AA or operator
  - Dial from other PBX
  - Customizable IVR
- Conference room PIN management
  - Conference moderator has special PIN
  - Normal participants have their own PIN
- Advanced Conference Control
  - Real time conference status monitoring
  - Participants control from web or DTMF, invite participants, convert to the moderator
- Conference Recording
Audio Conferencing Setup
Update PBX Firmware

Put image file in tftp or http server directory, specify the sever name or URL, then upload. After uploading, reboot the PBX.
SIP Server and NAT/Firewall Traversal

• Field-proven SIP Servers
  – SIP Registration
  – SIP Proxy
  – SIP B2B UA

• NAT/Firewall Traversal
  – Built-in Session Border Controller (RTP Relay)
  – SIP regular refreshing registration
  – Media flow relay
Typical PBX Network Setup Scenario – DMZ Mode

Router forward all request initiated by Internet user to the Router’s public IP to PBX
Setup PBX in Router’s DMZ

• Purpose
  – PBX can expose to the Internet so it is easy for other phones or PBX to find it
  – PBX can forward media between the PBX/Phones behind the NAT/Firewall
  – PBX only sees only voice traffic and other traffic still go through router, so PBX can perform better

• Guidelines
  – Go register a dynamic domain name (DDNS) if router uses dynamic IP address, e.g. dyndns.org, www.3322.org
  – Setup router’s DDNS according to specific router instructions
  – Put PBX in the router’s LAN and assign a static IP address to PBX
  – Enables router’s DMZ, pointing to PBX’s IP address
  – Enables router’s port range forwarding to reserve SIP and RTP ports that PBC will use
  – Setup DMZ support in PBX, tell PBX to use router’s public IP in SIP signaling
  – Try to register or access PBX web from other broadband link
Setup DDNS in Router

Choose the DDNS service provider name, DynDns is preferred

Fill in the domain name, user name and password you got when you registered the domain with DDNS service provider.
Assign a static IP for the PBX and enable router’s DMZ to point to PBX IP.
Assign Static IP to WAN in PBX

Please use different subnet for LAN and
Enable DMZ support in PBX

User router's DDNS domain name for external IP address

User router's static IP address for external IP address

Specifies WAN's subnet information
User Setup in DMZ mode

In DMZ mode, for the extension that registers to the PBX from Internet, please disable "Can Re-invite" option in the user property, otherwise one way voice can happen.
IP PBX Deployment Scenario 1 – Trust Host Based Trunking

1. PBX trust each other and does not require authentication – no need to register to other PBXs
2. Each PBX has its network ID – something like area code in PSTN
3. Each PBX defines prefix dialing to interconnect with other PBX
4. Each PBX has its own connection to PSTN, flexing routing rules can be set to land the PSTN calls according to best rate among the PBXs
IP PBX Deployment Scenario 2 – Authentication Based Trunking

1. IP PBX A and IP PBX B has independent extension group, A group is 3xxx, B group is 2xxx
2. IP PBX A log in IP PBX B with 2xxx account, IP PBX B register to IP PBX A with 3xxx account
3. Adding VOIP outgoing dialing rule, in IP PBX A add outgoing SIP account and dialing 2xxx to reach B, same setup in IP PBX B
4. IP PBX A and IP PBX B dial out to PSTN with prefix +9xxx
5. IP PBX A can dial out to IP PBX B’s PSTN with prefix +39xxx, IP PBX B dial out to IP PBX A’s PSTN with prefix +29xxx

1. A and B are independent of each other and have their own PSTN trunking interface
2. A and B has unified extension plan, all the extensions can be accessed from A and B
3. A and B has different local PSTN access number but unified auto attendant
1. All the SIP phones in group B are registered with IP PBX A, IP PBX B only provides local PSTN access and trunking interface
2. No special dialing rules are needed for the calling extensions
3. IP PBX A registers with IP PBX B to provide access to IP PBX B’s PSTN trunking interface
4. Prefix based dialing for PSTN dialing out, +9xxx for IP PBX A local access, +8xxx for IP PBX B local access