InterPBX Communication System
Blaze5000/Blaze1200/Savanna8000

Administrator
Installation and Configuration Guide
# Table of Contents

Chapter 1  InterPBX Communication System Overview ................................................................. 7  
  IP-Based Business Telephone System ....................................................................................... 8  
  Advantages of InterPBX ............................................................................................................ 8  
  Key Components ....................................................................................................................... 9  
    InterServer ............................................................................................................................ 9  
    VMS Server ............................................................................................................................ 9  
    Conference Server ................................................................................................................. 9  
    Recording Servers ............................................................................................................... 9  
    Voice Gateways .................................................................................................................. 9  
    SIP Proxy Server ................................................................................................................... 10  
    NAT Proxy Server ............................................................................................................... 10  
    CTI Solutions ..................................................................................................................... 10  
    Extension Types .................................................................................................................. 10  
    Web-based Management Tools .......................................................................................... 11  
    SH2500 PoE Switching Hub ............................................................................................... 12  

Chapter 2  Installing InterPBX Communication System ............................................................ 13  
  Before You Start ...................................................................................................................... 14  
    Prepare Your Telephone Numbering Plan ......................................................................... 14  
  Installing PBX Server ............................................................................................................ 15  
    Installing and Configuring PBX Server ................................................................................ 16  
    Connecting PBX Server via Console Port .......................................................................... 19  
  Installing VG5000 Voice Gateway ......................................................................................... 20  
    Connecting VG5000 via Telnet ............................................................................................ 24  
    Connecting VG5000 via Console Port ................................................................................. 25  
  Installing IP Phones ............................................................................................................... 26  
    Create IP Phones via Auto Discovery .................................................................................. 27  
    Create IP Phones Manually ................................................................................................. 32  

Chapter 3  System Configuration ................................................................................................ 35  
  General Parameters .................................................................................................................. 36  
    Basic IP Settings ................................................................................................................ 36  
    Email Settings ...................................................................................................................... 37  
    NAT Settings ...................................................................................................................... 37  
    Set QoS ............................................................................................................................... 38  
    Music on Hold ..................................................................................................................... 38  
    Ringing Patterns .................................................................................................................. 39  
  Company Information ............................................................................................................. 39  
    Business Hours .................................................................................................................. 39  
    Holidays .............................................................................................................................. 41  
    System Speed Dialing ........................................................................................................ 41  
    Call Restriction .................................................................................................................. 42  
    Route ................................................................................................................................. 43  
    Function Code ..................................................................................................................... 44  
    Password Management ...................................................................................................... 45  
    Set System Date/Time ......................................................................................................... 46  
    Miscellaneous ...................................................................................................................... 46  
      Timeout Settings .............................................................................................................. 46  
      Authorization Code .......................................................................................................... 47  

Chapter 4  Gateway Configuration ............................................................................................... 49  
  Analog Gateways ..................................................................................................................... 50  
    Creating Analog Gateway List ............................................................................................ 50  
    Editing Analog Gateways .................................................................................................... 51
<table>
<thead>
<tr>
<th>Chapter 5 Extension Management</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>Configuring IP Phone, SoftPhone and SoftConsole</td>
<td>81</td>
</tr>
<tr>
<td>Extension No., Phone Type and MAC Address</td>
<td>81</td>
</tr>
<tr>
<td>User Information</td>
<td>81</td>
</tr>
<tr>
<td>User Password</td>
<td>81</td>
</tr>
<tr>
<td>Off-Hook Access to</td>
<td>82</td>
</tr>
<tr>
<td>Class of Service</td>
<td>82</td>
</tr>
<tr>
<td>Button Mapping Group</td>
<td>82</td>
</tr>
<tr>
<td>CODEC</td>
<td>82</td>
</tr>
<tr>
<td>Jitter Buffer Depth</td>
<td>83</td>
</tr>
<tr>
<td>Silence Suppression</td>
<td>83</td>
</tr>
<tr>
<td>Conference Disabled</td>
<td>83</td>
</tr>
<tr>
<td>Paging Disabled</td>
<td>83</td>
</tr>
<tr>
<td>Enable/Disable Extension</td>
<td>84</td>
</tr>
<tr>
<td>Apply Settings</td>
<td>84</td>
</tr>
<tr>
<td>Button Mapping</td>
<td>85</td>
</tr>
<tr>
<td>Station Speed Dialing</td>
<td>89</td>
</tr>
<tr>
<td>Answer Option</td>
<td>90</td>
</tr>
<tr>
<td>Mailbox</td>
<td>91</td>
</tr>
<tr>
<td>Notification</td>
<td>93</td>
</tr>
<tr>
<td>Distribution List</td>
<td>95</td>
</tr>
<tr>
<td>Virtual Extensions</td>
<td>95</td>
</tr>
<tr>
<td>Analog Extensions</td>
<td>97</td>
</tr>
<tr>
<td>Off-Premises Extensions</td>
<td>98</td>
</tr>
<tr>
<td>SIP Phone List</td>
<td>100</td>
</tr>
<tr>
<td>InterConsole List</td>
<td>101</td>
</tr>
<tr>
<td>Chapter 6 Group Management</td>
<td>103</td>
</tr>
<tr>
<td>CO Line Groups</td>
<td>104</td>
</tr>
<tr>
<td>Creating CO Line Groups</td>
<td>104</td>
</tr>
<tr>
<td>Assigning Members to CO Line Groups</td>
<td>105</td>
</tr>
<tr>
<td>Extension Groups</td>
<td>105</td>
</tr>
<tr>
<td>Creating Extension Groups</td>
<td>106</td>
</tr>
<tr>
<td>Hunting Method</td>
<td>107</td>
</tr>
<tr>
<td>Wrap Up Time</td>
<td>107</td>
</tr>
<tr>
<td>Group Administrator Password</td>
<td>107</td>
</tr>
<tr>
<td>CTI Gateway</td>
<td>72</td>
</tr>
<tr>
<td>NAT Proxy</td>
<td>73</td>
</tr>
<tr>
<td>SIP Proxy</td>
<td>74</td>
</tr>
<tr>
<td>Add SIP Proxy</td>
<td>74</td>
</tr>
<tr>
<td>Add SIP Trunks</td>
<td>75</td>
</tr>
<tr>
<td>Create SIP Trunk ARS</td>
<td>77</td>
</tr>
<tr>
<td>Add SIP Extensions</td>
<td>79</td>
</tr>
<tr>
<td>Off-Premises Gateways</td>
<td>58</td>
</tr>
<tr>
<td>Configuring Off-Premises CO Line Ports</td>
<td>58</td>
</tr>
<tr>
<td>Configuring Off-Premises SLT Ports</td>
<td>59</td>
</tr>
<tr>
<td>Recording System</td>
<td>60</td>
</tr>
<tr>
<td>Setting Store-on-Demand (Available on Blaze5000 Series)</td>
<td>60</td>
</tr>
<tr>
<td>Setting Record-on-Demand (Available on Blaze5000 and Blaze1200 Series)</td>
<td>63</td>
</tr>
<tr>
<td>Setting Professional Recording System</td>
<td>65</td>
</tr>
<tr>
<td>Digital Line Gateways</td>
<td>66</td>
</tr>
<tr>
<td>Adding Digital Gateway</td>
<td>66</td>
</tr>
<tr>
<td>Modifying a Digital Gateway</td>
<td>68</td>
</tr>
<tr>
<td>Setting Trunk Parameters</td>
<td>68</td>
</tr>
<tr>
<td>Setting Trunk Port Parameters</td>
<td>69</td>
</tr>
<tr>
<td>Configuring Off-Premises Gateway List</td>
<td>57</td>
</tr>
<tr>
<td>Configuring Off-Premises Gateway</td>
<td>57</td>
</tr>
<tr>
<td>Configuring Off-Premises CO Line Ports</td>
<td>58</td>
</tr>
<tr>
<td>Configuring Off-Premises SLT Ports</td>
<td>59</td>
</tr>
<tr>
<td>Setting Trunk Port Parameters</td>
<td>69</td>
</tr>
<tr>
<td>Distribution List</td>
<td>95</td>
</tr>
<tr>
<td>Mailbox</td>
<td>91</td>
</tr>
<tr>
<td>Notification</td>
<td>93</td>
</tr>
<tr>
<td>Answer Option</td>
<td>90</td>
</tr>
<tr>
<td>Extension No., Phone Type and MAC Address</td>
<td>81</td>
</tr>
<tr>
<td>User Information</td>
<td>81</td>
</tr>
<tr>
<td>User Password</td>
<td>81</td>
</tr>
<tr>
<td>Off-Hook Access to</td>
<td>82</td>
</tr>
<tr>
<td>Class of Service</td>
<td>82</td>
</tr>
<tr>
<td>Button Mapping Group</td>
<td>82</td>
</tr>
<tr>
<td>CODEC</td>
<td>82</td>
</tr>
<tr>
<td>Jitter Buffer Depth</td>
<td>83</td>
</tr>
<tr>
<td>Silence Suppression</td>
<td>83</td>
</tr>
<tr>
<td>Conference Disabled</td>
<td>83</td>
</tr>
<tr>
<td>Paging Disabled</td>
<td>83</td>
</tr>
<tr>
<td>Enable/Disable Extension</td>
<td>84</td>
</tr>
<tr>
<td>Apply Settings</td>
<td>84</td>
</tr>
<tr>
<td>Button Mapping</td>
<td>85</td>
</tr>
<tr>
<td>Station Speed Dialing</td>
<td>89</td>
</tr>
<tr>
<td>Answer Option</td>
<td>90</td>
</tr>
<tr>
<td>Virtual Extensions</td>
<td>95</td>
</tr>
<tr>
<td>Analog Extensions</td>
<td>97</td>
</tr>
<tr>
<td>Off-Premises Extensions</td>
<td>98</td>
</tr>
<tr>
<td>SIP Phone List</td>
<td>100</td>
</tr>
<tr>
<td>InterConsole List</td>
<td>101</td>
</tr>
<tr>
<td>Chapter 6 Group Management</td>
<td>103</td>
</tr>
<tr>
<td>CO Line Groups</td>
<td>104</td>
</tr>
<tr>
<td>Creating CO Line Groups</td>
<td>104</td>
</tr>
<tr>
<td>Assigning Members to CO Line Groups</td>
<td>105</td>
</tr>
<tr>
<td>Extension Groups</td>
<td>105</td>
</tr>
<tr>
<td>Creating Extension Groups</td>
<td>106</td>
</tr>
<tr>
<td>Hunting Method</td>
<td>107</td>
</tr>
<tr>
<td>Wrap Up Time</td>
<td>107</td>
</tr>
<tr>
<td>Group Administrator Password</td>
<td>107</td>
</tr>
<tr>
<td>Topic</td>
<td>Page</td>
</tr>
<tr>
<td>-------------------------------------------</td>
<td>------</td>
</tr>
<tr>
<td>Redundant Server</td>
<td>156</td>
</tr>
<tr>
<td>Assigning Slave PBX Server</td>
<td>157</td>
</tr>
<tr>
<td>Installing and Setting Slave PBX Server</td>
<td>157</td>
</tr>
<tr>
<td>Joint Server</td>
<td>158</td>
</tr>
<tr>
<td>Creating Joint Server</td>
<td>158</td>
</tr>
<tr>
<td>Setting Joint Server CO Lines Access Control</td>
<td>160</td>
</tr>
<tr>
<td>Setting Links with Joint Server</td>
<td>160</td>
</tr>
<tr>
<td>Setting Joint Server Extension List</td>
<td>161</td>
</tr>
<tr>
<td>Joint Server Status</td>
<td>162</td>
</tr>
<tr>
<td>Appendix A: Function Code List</td>
<td>163</td>
</tr>
<tr>
<td>Appendix B: Retrieving Voice Messages and Recordings</td>
<td>167</td>
</tr>
<tr>
<td>Appendix C: InterPBX Management Website</td>
<td>171</td>
</tr>
<tr>
<td>Appendix D: Terminologies</td>
<td>173</td>
</tr>
<tr>
<td>Appendix E: Default Values</td>
<td>175</td>
</tr>
<tr>
<td>Appendix F: DTMF Programming</td>
<td>177</td>
</tr>
<tr>
<td>Appendix G: System Capacity</td>
<td>183</td>
</tr>
<tr>
<td>Index</td>
<td>185</td>
</tr>
</tbody>
</table>
Chapter 1
InterPBX Communication System Overview

InterPBX Communication System is a new generation business phone system that employs a data network for terminal connectivity and voice transport. Contrary to traditional business phone systems that carry voice over legacy TDM networks, InterPBX delivers voice in digitized packets over LAN or Internet networks.
IP-Based Business Telephone System

Packet switching telephony technology allows enterprises to work upon their existing data network infrastructure for implementing advanced applications and to save costs on long distance and international calls. As a nature of IP-based communication systems, the InterPBX brings significant savings on call tolls, simplifies wiring, improves management and maintenance, and provides advanced voice applications to enterprises’ needs.

Advantages of InterPBX

- Feature-rich – InterPBX supports an abundant number of features, including Auto Attendant, built-in Messaging System, Automated Call Distribution (ACD), Conference, Unified Messaging, Recording System, Auto discovery, Contact Center applications, and other new generation business applications.
- Flexibility – The capacity of InterPBX is not limited to a fixed system chassis. InterPBX provides flexible structure for growing businesses. It can be set up to meet the needs of enterprises from small to large or single to multiple sites. Enterprises can connect each location by VPN or Internet.
- Reliability – InterPBX employs a unique distributed NeuralServer architecture that allows multiple servers to coexist and communicate with one another in a single system. Through synchronization and backup mechanisms, InterPBX provides redundancy in the event of server failure and therefore drastically enhances the reliability of the system.
- Management – Browse-based management tool provides administrators and users a user-friendly interface to set, control, and maintain the system.
Key Components

DSG’s InterPBX Communication System supports flexible components. They could either be integrated as an embedded system or work individually.

- **Blaze 5000 Series**: It contains InterServer, VMS Server, Conference Server, Voice Gateway, and Recording Server.
- **Blaze 1200 Series**: It contains InterServer, VMS Server, Conference Server, Recording Server and Voice Gateway.
- **Savanna 8000 Series**: It contains InterServer, VMS Server and Conference Server.

InterServer

InterServer is the brain (central administration) of InterPBX Communication System. It is a software-based switching solution that handles call signals, call control and voice processing activities and manages all the extensions, voice gateway and other applications within the system.

VMS Server

VMS Server works seamlessly with PBX Server and provides Auto Attendant, Voice Mail, Automated Call Distribution (ACD) and Unified Messaging features.

Conference Server

Conference Server improves communications among employees by providing conference function.

Recording Servers

Recording Server helps enterprises record, monitor and search recording files. It can be integrated seamlessly with users’ IP phones bringing more efficiency for customer service representatives, banking officers or attorneys.

**Note**: Blaze 5000 provides built-in Record-on-Demand and Store-on-Demand functions and Blaze200 provides built-in Record-on-Demand. Savanna 8000 Series requires a stand-alone Recording Server.

Voice Gateways

Voice Gateway connects PSTN and IP networks and allows voice packets to be transferred between Internet and traditional PSTN network. VG5000 Voice
Gateway supports CO Line and/or SLT interfaces, which are able to connect CO lines or analog phones. VG6000 and VG7000 Voice Gateway supports T1/E1 ISDN network.

Note: The Savanna 8000 Series doesn’t provide built-in Voice Gateway.

**SIP Proxy Server**

DSG's SIP Proxy is a full-featured proxy server. It allows your Blaze or Savanna Series IP-PBX to get connected to operators' softswitch and to enjoy the provided SIP Trunk services. DSG SIP Proxy can register the IP-PBX to your service provider. All the extensions under the system, including DSG IP extensions or third parties' SIP phones, can access the SIP Trunks to make long-distance or international calls.

**NAT Proxy Server**

DSG NAT Proxy is a complete solution to solve complicated network problems that may happen when enterprises use the IP PBX Communication System. With DSG NAT Proxy, Blaze or Savanna IP PBX Communication Systems and Off-Premises Phones can traverse more than 95% of NAT or firewall successfully without creating other service ports, and therefore enterprises can communicate via IP-PBX Communication System with pure and stable voice quality.

**CTI Solutions**

DSG BlazeLink is a CTI solution enables business to develop applications to optimize workforce in a call center or contact center. Applications developed on BlazeLink for call control, monitoring, or managing are able to connecting DSG IP-PBX System to the data processing environment. With the supported tools and resources, developers are able to deliver a comprehensive solution for effective communications and better interactive experiences.

**Extension Types**

InterPBX Communication System supports various types of extensions - IP phones, analog phones, software phones and SIP phones. Administrators may set up any types of the phones as users’ needs.

**IP Phone:** IP phones offer various business telephone features. They could be installed on LAN or on remote sites.
SoftPhone: DSG's SoftPhone is an application software installed on PCs offering telecommuters or business travelers full-featured extension functions. Users can adopt their own headsets or use DSG S300X USB Phones when utilizing the SoftPhone.

Off-premises Phone: Off-premises IP phones, SoftPhones or analog phones on off-premises gateways are all able to connect to a company’s InterPBX Communication System via Internet or VPN and perform the same way as the extensions in the head office.

Analog Phone: Analog extensions are connected to the FXS ports of Voice Gateway like VG5000.

Attendant Console: Both software attendant console and hardware DSS console are supported.

Web-based Management Tools

InterPBX provides administrators and extension users a management tool with a GUI interface that allows administrators and extension users to set the preferences of InterPBX or extension via browser. The suggested browser is Internet Explorer 5.0 or later.

Administrator: The Administrator web site provides a web-based GUI interface allowing administrators to configure the InterPBX Communication System through the web browser. It helps administrators to manage and maintain the system easily.
**Extension User**: Extension users log in to user’s section to customize personal settings on phones like Button Mapping, Speed Dial, Message Option, Answer Option, Password and Phone Book.

**SH2500 PoE Switching Hub**

SH2500 is the recommended switching hub for InterPBX Communication System. It supports 802.3af PoE (Power-over-Ethernet), which offers IP extensions the current power over network cables. Users do not need additional power socket when using IP phones.
Chapter 2
Installing InterPBX Communication System

This chapter guides you through the preparation, installation and basic configuration of InterPBX Communication System.
Before You Start

The component of InterPBX Communication System contains servers, gateways and IP phones. Please follow the recommendations below when you install or operate your system in order to avoid any injury and damage.

Safety Recommendation

Always use ESD-preventive tools when you plug the power cord. Do not disassemble or remove chassis cover of any components of your InterPBX Communication System. If there is any problem of your system, please contact our service representatives.

Environmental Prerequisite

InterPBX Communication System needs to be installed in clean, dry, adequately ventilated areas. The server, gateway, and switching hub can be placed in a control room or on a rack. Please remain the control room in a suitable temperature and adequately ventilated environment.

Local Telecommunications Service

You have to apply for the local phone service in your area. The trunk lines of the central office need to be connected to Voice Gateways. DSG VG5000 Voice Gateway supports CO Line interface. Please make sure if your trunk lines support analog interface.

Prepare Your Telephone Numbering Plan

If you are going to replace your traditional telephony system with InterPBX Communication System, you can retain the old dialing plan. InterPBX Communication System supports flexible extension number lengths. The maximum extension number length is “5 digits”. The telephone numbering plan must not overlap or conflict with other numbers that are on your system or on Joint Server’s system.

You need to prepare the followings before you install InterPBX Communication System.

- **CO Line Extension Number**: Prepare an extension number for each CO Line port so that the associated extension number can reach each trunk line that is connected to it.
• **SLT Extension Number**: Prepare an extension number for each SLT port so that the associated extension number can reach each analog phone that is connected to it.

• **IP Phone**: Prepare an extension number for each IP phone.

• **Group Number**: If you plan to group specific trunk lines or extensions as a group, prepare a Group Number for each CO Line Group and Extension Group.

• **AA Menu**: Plan your AA procedure and prepare an access code for each AA Menu.

• **Routes**: Plan your Routes, ARS and Class of Service to be uses when making calls.

---

**Installing PBX Server**

![Blaze 5000 Series Connection Diagram](image1)

![Blaze 1200 Series Connection Diagram](image2)

![Savanna 8000 Series Connection Diagram](image3)
Installing and Configuring PBX Server

1. Connecting to Power Cord
Plug one end of the power cord to the power connector on the rear panel of PBX Server. Plug the other end of the power cord into a power outlet.

2. Connecting to LAN
You need to connect PBX Server to your existing Ethernet network. Connect an Ethernet cable from the “Network” RJ45 port on PBX Server to any 10/100BaseT RJ45 port on a switching hub.

3. Connecting to CO Lines or Extension Lines (For Blaze Series Only)
With Blaze Series, the Server is integrated with Voice Gateway module. Voice Gateway supports CO line or analog extension modules. Please plug a CO line connector to a CO Line port or plug an analog extension to a SLT port. If you would like to install other embedded analog or digital gateways, please refer to the related manual.

**Note:** If you accidentally plug the equipment with CO Line interface to SLT port, it may damage your equipment. Please make sure that your equipment connects to an appropriate interface.

4. Connecting to External Audio Source (For Blaze Series Only)
The MOH (Music on Hold) port can connect to a radio or CD player for playing music or your customized greetings for callers placed on hold.

You can also choose the pre-recorded audio files as the MOH on the PBX Server. If you choose to use the pre-recorded audio files from the system, do not connect the external audio source to the MOH port.

5. Connecting to External Paging Facility (For Blaze Series Only)
The “Paging” port allowing you to connect external paging equipment for broadcasting.

6. Setting Power Failure Transfer (For Blaze Series Only)
When power failure occurs and there is no backup power, the C.O. lines connected to port 1 will be switched to port 24. You can connect single line phones to port 24 to pick up calls. The PFT function only works on port 1 with
FXO interface supported. Port 24 can be FXO, FXS or empty ports. When power is supplied, the PFT function is not available.

7. Preparing a Computer
Prepare a computer with web browser and be sure it is on the same subnet as of your PBX Server. Change the computer’s IP settings if necessary. The suggested web browser is Internet Explorer version 5.0 or later.

The default values of PBX Server are as follows:
- Default IP Address: 192.168.1.200:88 (The service port is 88)
- Default Gateway: 192.168.1.254
- Default Subnet Mask: 255.255.0.0

8. Login PBX Server
- Launch the web browser. On the address bar, type in the PBX Server’s default IP address 192.168.1.200:88 to access PBX Server.

- You will be connected to InterPBX Administration Website. Please click the Administrator icon.

- When you login the InterPBX Administration for the first time, you will see the following picture. Please enter your user name and password and then save the settings. Please remember your user name and password in order to login again in the future. The password shall not exceed 8-character alphanumeric. The user name and password are case sensitive.
When you login the InterPBX Administration next time, you will see the following picture. Please enter your user name and password to login.

- After you login the InterPBX Administration successfully, the screen will display the main menu of InterPBX Administration website as follows:

9. Basic Settings
There are some basic items you need to configure allowing the system to work properly.
- PBX Server IP Settings: Go to Main Menu>System Configuration>General Parameters to set your PBX’s IP. If you have more than one PBX server, go to Main Menu>Multi-Server Management to assign each an ID and set the Joint Server connection.
• License Keys: Go to Main Menu>Operation Management>General Information. Input the license keys you purchased.

• Trunks: Go to Main Menu>Gateway Configuration. Depending on the voice gateway you purchased, add your gateways and assign CO line extension numbers. In addition, you many go to Main Menu>Group Management>CO Line Group to group CO lines for better managing your trunks.

• Extensions: Go to Group Management>Class of Service to set CoS for providing different call permissions to extensions applied. Go to Main Menu>Extension Management to add your extensions or turn on the Auto Discovery allowing the system to detect connected extensions automatically.

• AA Menu: Go to Main Menu>System Configuration>Business Hours to set your company operating hours. Then go to Main Menu>VM Configuration>AA Management to set AA trees. And make sure the Ring Assignment of CO lines is set to the appropriate AA Menu Access Code.

• Accessing CO Lines: Each CoS will have its CO priority. Or you may go to Main Menu>System Configuration>Route to set routes and go to Main Menu>Group Management>Class of Service to further assign ARS for better using your trunk resource.

You may change your computer IP setting to the original values after you finish setting up the Blaze/Savanna Server. You will need to enter the new IP address to connect to the Blaze/Savanna Server next time.

If you encounter any problem or have questions during installation or operation, click on the Help icon from the web setting page for assistance.

**Connecting PBX Server via Console Port**

When you forget the IP address of the PBX Server, you can connect a computer to the console port of the PBX Server in order to check the current IP address, gateway address, and subnet mask.

1. Connect one end of the RS232 transmission line to the computer’s COM port and the other end to the Console port of PBX Server.

2. Open the Hyper Terminal program. Enter the connection name and set the baud rate to 9600 bps.

3. Turn off the power of PBX Server. Wait for a while and then turn on the power again. The computer can start to connect to the PBX Server.
4. You can see the IP address from the “Hyper Terminal” page. You can also change the IP address here. If you are going to change the IP address, please press any key within 3 seconds to setup.

Note: The RS-232 DB-9 connector enclosed with the package is with pin 2/3, pin 4/6, and pin 7/8 shorted.

Installing VG5000 Voice Gateway

Figure: VG5000 Voice Gateway Connection Diagram

This section will guide you through the installation and configuration of VG5000. For more details, please refer to VG5000 Operation Manual. For other model Voice Gateways, please refer to their respective manuals.

1. Connecting to Power Cord

Plug one end of the power cord to the power connector on the rear panel of the VG5000. Plug the other end of the power cord into a power outlet. After you turn on the power switch on the rear panel, you can verify the function by checking if the LED labeled Power is on.
2. Connecting to LAN
You can connect VG5000 to your existing Ethernet network. Please connect one end of the Ethernet cable to the network RJ45 port of VG5000 and connect the other end of the Ethernet cable to any 10/100BaseT RJ45 port on DSG SH2500 Switching Hub or your existing switching hub.

3. Connecting to Trunk Lines or Analog Extensions
VG5000 supports trunk lines and analog extensions. Please plug the trunk line from the Central Office to the CO Line port or plug the analog extension to the SLT port.

Note: Misplacing FXO equipment into FXS interface and vice versa may damage your VG5000 Voice Gateway. Be sure that the FXO/FXS interfaces of VG5000 are connected to suitable telephone lines and equipment.

4. Connecting to External Audio Source (Optional)
InterPBX Communication System supports external and internal MOH (Music on Hold). VG5000 Voice Gateway provides a “MOH” port for connecting to the radio or CD player. When callers are placed on hold, the system will play the music from the radio or CD player. Please insert the plug of your music source to the MOH port to enable this function.

You can also choose the pre-recorded audio files as the MOH on the PBX Server. If you choose to use the pre-recorded audio files from the system, do not connect the external audio source to the MOH port.

5. Connecting to External Paging Facility (Optional)
VG5000 Voice Gateway provides a “Paging” port allowing you to connect external paging equipment for broadcasting to co-workers. Each VG5000 supports a paging port. If you have more than one VG5000, you can separate them into different paging zones.

6. Setting Power Failure Transfer (Optional)
VG5000 Voice Gateway provides PFT (Power Failure Transfer) function. When power failure occurs and there is no backup power, the C.O. lines connected to port 1 will be switched to port 24. You can connect single line phones to port 24 to pick up calls. The PFT function only works on port 1 with FXO interface.
supported. Port 24 can be FXO, FXS or empty ports. When power is supplied, the PFT function is not available.

7. Prepare a Computer
Prepare a computer with web browser and be sure it is on the same subnet as of VG5000. Change the computer’s IP settings if necessary. The suggested web browser is Internet Explorer version 5.0 or later.

The default settings of VG5000 are as follows:
• Default IP Address: 192.168.1.201:89 (The service port is 89)
• Default Gateway: 192.168.1.254
• Default Subnet Mask: 255.255.0.0

8. Login VG5000
• Launch the web browser. On the address bar, enter the VG5000 default IP address http://192.168.1.201:89 to access the VG5000.
• After you access to the web page of the VG5000, click on the Login icon. On the login page, please enter the password. The default password is “1234”.

• After you log into the system successfully, the screen will display the System Information page of DSG VG5000 as below:
• Basic IP Settings: On the System Information page, please enter the IP address, gateway address and subnet mask assigned for VG5000. And then enter the IP address of the PBX Server in “PBX Server.”

• Upgrade VG5000 Software (Optional)
If you need to upgrade the software version of the VG5000 in the future, please click “Upload Pack” item after you access to VG5000.

9. Settings on InterServer
After you install the VG5000 Voice Gateway, you need to set the VG5000 on the PBX Server.

• **Enter License Key**: Please login the PBX Server and go to Main Menu>Operation Management>General Information. Click “Add” and then enter your license key to make sure the system capacity fits your needs.

![General Information](image)

• **Creating Analog Gateway List**: After you install VG5000, you need to set VG5000 on the PBX Server. Each VG5000 has a MAC address. You can see the MAC address from the label on the VG5000 or from the VG5000 web or through Telnet. Please record your MAC address in order to register to the PBX Server. Access the PBX Server and login to the InterPBX Administration Website. Go to Main Menu>Gateway Configuration>Analogue Gateways. Click the Add button to create an
Chapter 2 Installing InterPBX Communication System

Analog Gateway.

![Image showing Add Analog Gateway page]

- **Editing Basic Gateway Data**
  On the “Add Analog Gateway” page, enter the name and the MAC address of your gateway, and the range of CO Line/SLT ports. The port number range starts from 1. For example, the CO Line port range of a 24-port VG5000 with FXO interface is from 1 to 24.

- **External Paging Setup**: If you connect a paging amplifier to the “PAGING” port on VG5000, select the “External Paging Enabled” option to enable this function and assign a specific paging code at “Paging Code” box. For example, if you assign “111” as the paging code of your VG5000, please dial “111” to broadcast.

- **Music On Hold (MOH) Setup**: If the VG5000 Voice Gateway connects to the external music source to play the Music On Hold, please go to Main Menu>System Configuration>General Parameters and then enable the “Music On Hold” function. For more details about MOH, please refer to Chapter 3 System Configuration/System Parameters/Music On Hold.

**Connecting VG5000 via Telnet**
Besides connecting VG5000 via the web browser, you can also connect VG5000 via Telnet.

1. Prepare a computer with Telnet program and be sure it is on the same subnet as of VG5000.
2. Open the Telnet program and connect to the IP address of VG5000, or enter C:\Telnet 192.168.1.201 90 (IP address+space+90) under DOS. In which “192.168.1.201” is the default IP address of VG5000 and the communication port is 90.
3. Enter the default password “1234” to login VG5000.
4. After you have logged into the VG5000 successfully, a window will be shown as below. Please follow the instruction to edit the IP address, Gateway IP, Subnet Mask, PBX IP, Off-premises items and Telnet password of VG5000. After the editing, please save and exit.

Connecting VG5000 via Console Port

When you forgot the IP address for VG5000, you could connect a computer to the console port of VG5000 to look up the current IP address, Gateway IP, and Subnet Mask.

1. Please plug one end of the RS232 cable into the COM port of your computer and the other end into the Console port of VG5000.
2. Open the Hyper Terminal program. Enter the profile name for the connection and set the baud rate to 9600 bps.
3. Turn off the power of VG5000. Wait for a while and turn on the power again. The computer will create a connection with VG5000.
4. You can see the IP address of the VG5000 from the window of Hyper Terminal. You may also change the IP address in this window. If you want
to change the IP address, please press any key within 3 seconds.

**Note:** The RS-232 DB-9 connector enclosed with the package is with pin 2/3, pin 4/6, and pin 7/8 shorted.

**Installing IP Phones**

InterPBX Communication System supports IP phones, such as IP590, IP580 or IP500. IP phones provide various functions like Voice Mailbox, Message indicator light, Auto-Answer, Mute, Replay, Hold, Transfer, DND, and Speaker.

You may allow the system to automatically discover and register all the IP phones connected LAN via Auto Discovery function. Strongly recommend you to install IP phones via Auto Discovery at the first time. You can also create an IP Phone manually.

**Note:** Please do not connect your IP phone to LAN before you start Auto Discovery.
Create IP Phones via Auto Discovery

InterPBX Communication System provides Auto Discovery function that allows PBX Server to automatically search and register IP phones. When you install your IP phones at the first time, we recommend you to start Auto Discovery in order to simplify the installation procedure. Please make sure the relevant License Key has been added.

This section will guide you through the installation of IP580. For other types of phones, please refer to the respective operation manuals.

1. Creating Default Class of Service

Specify a commonly used call limitation as the default Class of Service (CoS). The default CoS will be applied to all IP extensions when registering with PBX Server using Auto Discovery.

- Access the PBX Server and login to InterPBX Administration Website.
- Go to Main Menu>Group Management>Class of Service. And then click the Add button to create a new Class of Service (CoS).
- Assign a name and select the call permissions. Click the Submit button to save and exit. You can modify the CoS or create more entries later. See
2. Turn on Auto Discovery

- Go to Main Menu > Operation Management > Auto Discovery.
- Click the Turn On Auto Discovery button to enable the Auto Discovery function. The Current Mode will display "Auto Discovery On." After the Auto Discovery is turned on, if you don’t turn off this function manually, the system would automatically turn off the Auto Discovery within 2 hours.
- You will find the Class of Service you have created from the list of “Set Default CoS for Auto Discovery.” Select the one you would like to set as the default value and click the Set Default button. You will see an arrow sign pointed to the default CoS. Click the Submit button to save and exit.
3. Enter License Key
Before installing the IP phones, be sure to enter the License Key on the PBX Server set up page. Please login to the PBX Server and then go to Main Menu>Operation Management>General Information. Click “Add” and then enter your license key.

![License Key Image]

4. Settings on IP Phones
You will need to assign the Extension Number, IP Address, Gateway IP Address, Subnet Mask and PBX Server IP to each IP phone. Please DO NOT connect your IP phones to LAN before completing the above setups. Press the middle button to enter the setup menu. Press the downward button next to the middle button to move to the next setting item or the upward button next to the middle button to move to the previous setting item. Use the keypad to input digits.

- Plug in the power cord of the IP phone and please DO NOT connect the IP phone to LAN.
- After the system check, press and hold the middle button for 3 seconds to enter the setup menu.
- You will see “EXT NUMBER” shown on the LCD screen. Press the button to start editing and use the keypad to input the extension number assigned to the IP phone. Press the button again to save.

- Press the downward button and the LCD screen will display “IP ADDRESS.” Press the button to start editing and use the keypad to...
input the IP address assigned to the IP phone. Press the $ button to save.

- Follow the above procedures to assign the “GATEWAY IP” and “SUBNET MASK” to the IP phone

- Keep pressing the downward button until the LCD displays “PBX SERVER IP.” Press the $ button to start editing and use the keypad to input the IP address of your PBX Server. Press the $ button to save.

- Keep pressing the downward button until the LCD displays “EXIT SETUP”. Press the $ button to exit.

The IP Phone will automatically search and download the latest version of software from the PBX Server and then reboot with the latest version.

**In DHCP Environment**

If your LAN is under DHCP environment, you can skip the settings of IP address, Gateway IP and Subnet Mask. Instead, you can move to “DHCP” setup item and enable this function.

- Plug in the power cord of the IP phone and please DO NOT connect the IP phone to LAN.
- After the system check, press and hold the middle $ button for 3 seconds to enter the setup menu.
- You will see “EXT NUMBER” shown on the LCD screen. Press the $ button to start editing and use the keypad to input the extension number assigned to the IP phone. Press the $ button again to save.

- Keep pressing the downward button until the LCD displays “DHCP.” Press the $ button to start editing and then press the downward button.
button to change the selection to “On.” Press the $ button again to save.

- Keep pressing the downward ⬇ button until the LCD displays “PBX SERVER IP.” Press the $ button to start editing and use the keypad to input the IP address of your PBX Server. Press the $ button to save.

- Keep pressing the downward ⬇ button until the LCD displays “EXIT SETUP”. Press the $ button to exit.

**Other Setting Items:**

- **MAC Address:** MAC address shows the hardware address of an IP phone. It is not editable.
- **Image Version:** Display the current software version of IP phones.
- **Set Password:** You may change the password for logging in to the phone. If it is blank, the password will not be required when entering the setting mode. The default password is “1234”.
- **Echo Utility:** You may input the PBX Server IP here and press Test. The phone set will send testing packets to the assigned IP address and reply the round trip time. This can help you test the connection status with the PBX server.

5. **Connect Your IP Phone to LAN**

- Connect your IP phone to the Ethernet Network. Please connect one end of the network cable to the “LAN” port on an IP phone and connect the other end to the RF45 port of 10/100 BaseT switching hub or Hub on LAN. PBX Server will automatically search the IP phones on LAN via Auto Discovery and register them to the phone list.

- During the registration, the LCD screen will display “System Checking”, “Server Searching”, “Authorizing”, “Load Setting”, and then “Extension Number and Current Time.”

If the LCD on the phone displays “Server Searching” or “Server Not Found”, it means there is a connection problem between this phone and PBX Server. Please check the network settings of your phone or your network environment.
again. Please also make sure the IP address and extension number are not used by other extension accidently.

6. Turning Off Auto Discovery
Access PBX Server and login InterPBX Administration Website. Go to Main Menu>Operation Management>Auto Discovery. On the “Auto Discovery” page, click the “Turn Off Auto Discovery” icon to disable the Auto Discovery function. This will prevent your InterPBX Communication System from being accessed by unauthorized users.

After the Auto Discover was turned on, if you didn’t turn it off manually, the system will automatically turn off the Auto Discovery function within 2 hours in order to avoid unauthorized users using the system.

Create IP Phones Manually
Besides creating IP phones via Auto Discovery, you can also create IP phones one by one manually. Please make sure the relevant License Key has been added.

1. The Settings on the Administration Website
   • Before installing the IP phones, be sure to enter the License Key on the PBX Server setup page. Please login to the PBX Server and then go to Main Menu>Operation Management>General Information. Click “Add” and then enter your license key.
   • Go to Main Menu>Extension Management>InterPhone List. Click the Add button to create IP phones.
   • Enter the IP phone’s extension number on “Extension No.”
Chapter 2 Installing InterPBX Communication System

- Select the phone type from the “Phone Type” list.
- Enter the password of the extension in order to edit the personal options from the InterPBX Management Website. The password must only be numbers. If you create IP phones via Auto Discovery, the phone password is blank. You can set the password later.
- Enter the MAC address of the phone in “MAC Address.” MAC is the hardware address of the IP phone. Each IP phone has a unique MAC address. You can see the MAC address of each IP phone by the following procedure. Press and hold  on the phone for 3 seconds to enter the setup menu. And then keep pressing  until the LCD screen displays “MAC ADDRESS.”

You may set the other items later. For other settings, please refer to Chapter 5 Extension Management/Set IP Phones, SoftPhones, and Virtual Phones.

2. Settings on IP Phones
Press and hold  button on the IP phone for 3 seconds. The LCD screen will display “EXT NUMBER.” Please refer to the above IP phones settings to enter the relevant Extension Number, IP Address, Gateway IP, Subnet Mask, and the IP Address of PBX Server.

3. Connect IP Phones to LAN
- After you finish the above procedure, please connect your IP phone to LAN.
- During the registration, the LCD screen of IP phone will display “System Checking”, “Server Searching”, “Authorizing”, “Load Setting”, and finally show the Extension Number and the Current Time.
- Login to InterPBX Administration website. Go to Main Menu>Extension Management>InterPhone List. You will see the list of all the IP phones. You can start to set the phone functions one by one.
Chapter 3
System Configuration

This chapter guides you through the initial and basic configuration of InterPBX Communication System. The basic system settings include PBX Server IP settings, your company information, your business hours, system-wide speed dialing, call restriction, password management, and system date and time settings.
General Parameters

Please login to the InterPBX Administration Website. Go to Main Menu>System Configuration>General Parameters. In this section, you can edit the basic IP settings of your PBX Server.

Basic IP Settings

IP Address, Default Gateway, Subnet Mask

The PBX Server is shipped with default IP settings as below for your initial configuration. You may edit the IP address, Gateway IP and Subnet Mask in this section.

- Default IP Address: 192.168.1.200 (service port 88)
- Default Gateway IP: 192.168.1.254
- Default Subnet Mask: 255.255.0.0

The PBX Server requires a static IP address. Either real IP or pseudo IP can do. The Gateway IP is for the destination host to route the IP packets addressed to a host outside the local subnet. Notice that all the on-site IP extensions need to be set in the same subnet of your PBX Server.

Host Name

Enter the Host Name of your PBX Server. If you have added the host name to your name resolution systems (DNS), you could connect to your InterPBX Communication Server by typing its name on a web browser. If you haven’t added it to your DNS, you can still access PBX Server using its IP address.
DNS 1 and DNS 2
Enter the Domain Name Server (DNS) IP address offered by your ISP. You can enter up to 2 DNS IP addresses.

Email Settings
Please enter the SMTP Server IP address in the field SMTP Server and the e-mail address in the field E-mail Account. When the Unified Messaging function is enabled, the system can send the voice message file via e-mail through the SMTP server to the extension user.

You may also enable the SMTP Authorization function if the SMTP Server needs to verify the e-mail account and password. If you enable SMTP Authorization, please be sure to enter the Account Name and Password.

NAT Settings
If your network environment is behind NAT, you can assign a virtual IP to the PBX Server. If you have off-premises extensions or Joint Servers, you need to enter the NAT information of the PBX Server to allow the off-premises extensions to connect to your PBX Server which is behind NAT. Please check the “Behind NAT” box and enter your real IP address of NAT equipment on the “NAT IP Address” box. If there are no off-premises extensions or Joint Servers in your InterPBX Communication System, you don’t need to edit the settings of NAT.

Open a Communication Port: When there are off-premises extensions or Joint Servers in your PBX Server that is behind NAT, besides the above settings, you also need to open a service port on your NAT Equipment or Firewall in order to allow the off-premises equipments to connect to your PBX Server.

Off-Premises Extensions: When you have off-premises extensions, please open the UDP6046 port on your NAT equipment. If the off-premises extensions are also behind NAT equipment, you also need to open the UDP6046 port for them. Please refer to Chapter 5 Extension Management/ Off-Premises Extensions for more details.

Joint Servers: When you have Joint Servers, please open the UDP6055 port on your NAT equipment. If the Joint Servers are behind NAT, you need to turn on
the NAT settings of the PBX Server and open the UDP6055 port on NAT equipment.

**NAT Proxy:** If you have DSG NAT Proxy, you only need to register the PBX Server to the NAT Proxy without opening the communication port for PBX Server or off-premises extensions.

### Set QoS

The InterPBX Communication System supports 802.1p/Q. You can put a check on “QoS Enabled” box to activate the system’s QoS. You may set VLAN ID and 802.1p Priority. If you activate the QoS function, your settings need to be the same as the settings of Switch.

The QoS of system only covers local IP extensions and Voice Gateway. You can also set the individual QoS on Off-Premises Extensions, Off-Premise Voice Gateway and Joint Server. For more information about QoS settings, please refer to Chapter 4 Gateway Configuration/Off-Premises Gateways, Chapter 5 Extension Management/Off-Premises IP Extensions, and Chapter 10 Multi-Server Management/Joint Server.

**Note:** If the values of VLAN ID and 802.1p Priority are both set as “0”, the QoS will be disabled.

### Music on Hold

When a call is on hold, the system will provide music or prerecorded announcements for callers. Select the check box of “Music on Hold” to enable this function. You can select the Music on Hold (MOH) source from an external audio source connected to a gateway or from a prerecorded file.

- You can play the music by selecting the prerecorded audio file. Please click the check box of “From Files” and select a preferred file from the list.
- If you connect the system with a radio or CD player, you can click the check box “From Gateways” and select the specific gateway from the gateway list. Make sure the gateway is properly connected to the audio source before you select the specific gateway.

You may also amplify the MOH volume if necessary.
**MOH Converter:** DSG provides a MOH Converter program allowing you convert your own music on hold or queue announcement files from WAV or MP3 format to the format compatible to the PBX Server. To know how convert your own MOH files, please consult our dealers or sales representatives.

**Ringing Patterns**
You can assign different ringing patterns to identify internal or external calls. From the “Internal” list, select your preferable ringing pattern for Intercom calls. From the “External” list, select one for calls from CO lines.

**Company Information**
Go to Main Menu>System Configuration>Company Information. In this section, you can edit your company’s contact information.

**Business Hours**
Your office hour settings will affect the behaviors of Automated Attendant, Class of Service (CoS) or Voice Mail System. Automated Attendant may play different greetings when the office is open, close or during lunch breaks. Permissions of making calls after office hour may be limited by the settings of CoS.
1. Go to Main Menu>System Configuration>Business Hours.
2. From the “Operating Mode” list, select your company’s operating mode.
   - Auto: It will switch automatically according to the schedule you set below.
   - Business Hours: The open hours on workdays.
   - Break Hours: The lunch breaks on workdays.
   - After Hours: The hours after open hours on workdays.
   - Closed: The hours of non-workdays.
3. When setting the business hours, select the check box of the workday and enter the opening hours and lunch breaks. If your company doesn’t offer lunch breaks or days off, you may enter “0”.

**Ring Assignment:** The Ring Assignment will follow the operating hours you set to play different greetings for incoming calls. For more information about Ring Assignment, please refer to Chapter 4 Gateway Configurations/Analog Gateways.

**Holidays:** For holidays, you may go to Holiday to set hours and dates. For more information about Holiday, please refer to Chapter 3 System Configuration/Holiday.

**Change Current Operating Mode:** You may manually change your current Operating Mode to be Business Hours, Break Hours, After Hours or Closed. Select the preferable item from the “Operating Mode” list. Notice that your operating mode will be permanently in the mode you selected. To let the operating mode be switched automatically according to the business hours you set, please select “Auto” again.
Night Service: You may assign a specific programmable button as Night Service. And press to switch the Operation Mode to Closed Hours. The AA menu and operators will be changed accordingly. This function is available for Extension Group Button Mapping. Please refer to Chapter 6 Group Management/Button Mapping Groups for more details.

Holidays
In addition to the regular business hour schedule, you can set the holiday list so that the system will play different greetings on holidays. For more information about Business Hours, refer to Chapter 3/System Configuration/Business Hours.

1. Go to Main Menu>System Configuration>Holiday.
2. Click the Add button to create an entry in the Holiday List.
3. Set the date range (month/day) of the holiday. If there is only one day off, enter the same date on the Date range boxes.
4. Input the name of the holiday.
5. Select an AA Menu from the list. If you select the “Enable Holiday” checkbox in CO Lines or the Ring Assignment setting of CO Line groups, the system will play the greeting of AA Menu you assigned here during holidays.

For more information about Ring Assignment, please refer to Chapter 4 Gateway Configuration/Analog Gateways.

System Speed Dialing
You can edit frequently dialed phone numbers in this section to be used throughout the system.
1. Go to Main Menu>System Configuration>System Speed Dialing.
2. On the “System Speed Dialing” list, select one entry.
3. Enter the destination phone number or extension number on the “New Number” box.
4. Enter the name of the destination or other description on the “Comment” box.
5. Click the Assign button to save.

**Note:** You could set up to 1000 sets of system speed dialing. If you would like to set a Speed Dialing Number for an outgoing call, please remember to add the CO line access code (e.g. 0). For long distance calls, please enter your long distance access code (e.g. 1) followed by the area code and phone number. For international calls, please enter your international access code (e.g. 011) and the country code, area code, and phone number.

To make a call with System Speed Dialing:
1. Lift the handset or press the Speakerphone button.
2. Press the System Speed Dialing function code #20.
3. Press the System Speed Dialing number (e.g. 000) to call.

**Call Restriction**
You can set the long distance and international access codes in the “Call Restriction” in order to help the PBX Server to identify the types of outgoing calls (e.g. local, long distance or international calls). Call Restriction allows you to create call restrictions, exceptions, and routing tables in Class of Service.
1. Go to Main Menu>System Configuration> Call Restriction.
2. Set the “Country Code (e.g. 1 for US)” and “Area Code (e.g. 213 for L.A.)” where PBX Server is located.
3. On the list of “Long Distance Call Prefixes”, select one entry and assign the long distance call prefix plus the area code (e.g. 1310 for Beverly Hills) in the “New Number” box. Press the Assign button to save. Repeat this procedure to edit all the area codes of your country.
4. Follow the above-mentioned procedure to set the “International Call Prefixes”. (e.g. 011)

**Route**

If you want to activate the ARS function, please create the available routes for selection. The route types for selection are CO Group, SIP Trunk, or Joint Server.
Chapter 3 System Configuration

1. Go to Main Menu>System Configuration> Route. Click “Add” to create a route.

2. Enter the name of the route.

3. Set the route in different time sections: Business Hour, Break Hour, After Hour, Closed Hour, and Holiday. The route types for selection are as follows:
   - CO Group: Select the CO group from the list. For more information about CO Group settings, please go to Main Menu>CO Line Groups.
   - Account: Please select your SIP Account. For more information about SIP Account settings, please refer to Set SIP Proxy.
   - Joint Server: You may also make the phone call through another InterPBX’s CO line. Please select your Joint Server here.

**Function Code**

The System provides Function Code for administrators to configure so that the extension users can use multiple functions by entering the specified function code directly on their extensions.
Go to Main Menu>System Configuration>Function Code. You will see all the function items available for configuration. Please enter a code for each function and click the save button.

**Password Management**

This section allows the administrator to change Administrator’s Login Name and Password and reset User Password to login the InterPBX Administration website. The administrator will be authorized to edit all the administrative features after login. For security reason, please change the Administrator Password from time to time.

Go to Main Menu>System Configuration>Password Management. Input your current password and new password. Re-enter the new password for confirmation. Your password shall not exceed 8 characters. The password is case sensitive.

Each user will be authorized to edit the user features after login. When users forgot their passwords, you may reset their passwords to the default setting (The default extension password is blank if created by Auto Discovery).
You can connect VMS Server from a telephone remotely and edit the relevant settings. The default remote Tel. Programming Password of VMS Server is “1234”. For more information about DTMF remote control settings, please refer to Appendix F.

**Set System Date/Time**

Go to Main Menu>System Configuration>Set System Date/Time in order to set the current date and time of your time zone for your InterPBX Communication System. The system date and time affect the play contents of business hours and greetings. If you change the system Date and Time, the time display on the phones will be updated after 8 seconds to 2 minutes.

![Set System Date/Time](image)

**Miscellaneous**

Go to Main Menu>System Configuration>Miscellaneous. You could edit system timeout settings and authorization code.

![Miscellaneous Settings](image)

**Timeout Settings**
• **Forward Voice Mail Timeout**: If an incoming call is not answered, the call will be transferred to the voice mail system after the timeout.

• **Call Park Timeout**: When the call is parked, if no one picks up the call, it will be bounced back to the original extension parked the call after the timeout.

• **Call Hold Timeout**: When a call is placed on hold without any further action, the call will ring back to the original extension after the timeout.

• **Transfer Timeout**: When a call is blind-transferred to a target extension but no one answered or the extension is busy, the call will be transferred to the Auto Attendant after the timeout. The Auto Attendant will guide the caller to other selections according to the settings of Transfer Options. If the personal forwarding is set on the target extension, the call will be forwarded according to the personal forwarding setting.

• **First-Digit Timeout**: When a line is accessed, if the first digit is not received within the period, the system will disconnect the call and release the line.

• **Inter-Digit Timeout**: When entering digits, if the next number is not received within the period, the system will see it as a typing ending signal and use the numbers been input to make calls.

• **Auth. Code Timeout**: When entering authorization code, if the next digit is not received before the timeout, the system will deem the input ends and send out the numbers received.

• **Max Trunk-to-Trunk Duration**: The function allows CO line incoming calls to be transferred out through the other CO lines. Please enter the call duration limit time. When the period expires, the system will cut off the call and release the CO line.

**Authorization Code**

Authorization Code function allows callers to make calls, such as long distance or international calls, from an extension with limited call permissions.

**Note**: The Authorization Code function needs to be implemented with ARS.

2. Input the length of Authorization Code.
3. Input the Cancel Symbol allowing users to clear the authorization code been input. This will help the user not to input the phone number again.

Voice Gateways are bridges between IP and traditional (PSTN) telephony networks. Gateways connect the InterPBX Communication System to CO trunk lines or analog extensions. InterPBX supports various gateway types including analog gateways, digital gateways, and off-premises gateways. If you have included Digital Gateway, Recording System, CTI Gateway, SIP Proxy or NAT Proxy in your purchase, please consult their specific documentation for further details on the settings.

Note: Savanna 8000 Series doesn’t include the Voice Gateway. If you would like to add the Voice Gateway into the Savanna 8000 Series IP-PBX system, we offer the VG5000 Voice Gateway or VG6000 Digital Gateway for your options.
Analog Gateways
InterPBX Communication System supports analog gateways such as VG5000. VG5000 Voice Gateway provides up to 24 ports of FXO (for CO Lines), FXS (for SLT), or mixture of both. Every unit of VG5000 also provides interfaces for external music-on-hold and paging equipment.

Creating Analog Gateway List
After the installation of your gateway, you will need to incorporate it into the PBX Server.

1. On the main screen go to Main Menu>Gateway Configuration>Analog Gateways.
2. Click the Add button to add a new gateway to PBX Server’s “Analog Gateway List”.
3. Name this gateway and enter its MAC address. MAC address is the hardware address of the gateway, you will find this information printed in a label at the back of the gateway or alternatively, you may obtain this information from the VG5000 Voice Gateway’s web page or Telnet. Please see Chapter 2 “Installing InterPBX Communication System” for more details.
4. Enter the range of CO Line and/or SLT ports of your gateway. Please note the port number starts from 1, for example, if the VG5000 you purchased is a 24-port FXO gateway, in the “CO Line Ports (From/To)” fields enter “1” and “24” respectively.
5. At “CO Line Extension Base” input the starting number of CO Line extension number. The CO Line extension numbers will be assigned automatically. For example if you set it as 4000 for an 8-port FXO gateway, the CO Line extension number of this gateway will be 4000-4007.
6. At “CO Line Index Base” input the starting number of the CO line display number. For example if you set it as 1, when there is an incoming call from port 1, the IP phones will display “Incoming Call CO 1” instead of the extension number of the port, helping you to identify the CO lines being used.

7. From “Jitter Buffer Depth” drop down menu, depending on your bandwidth and CODEC, start with a Jitter Buffer with the minimum value. Jitter Buffer will dynamically adjust its value according to the party you are talking to and the bandwidth of the call but never below the value you have set. The higher value of Jitter Buffer will reduce the chance of packet loss during a call but might cause delays and the lower value will help to transmit the voice packets faster but might cause packet loss problems.

8. Min. Loop Current Drop Time: Normally, when a telephone call terminates, the telephone company sends a momentary drop in loop current to signal the disconnect event. Please enter the minimum Loop Current Drop time in this field to avoid the IP-PBX system detecting the disconnect signal incorrectly. The value should not be too long or too short. Recommended value is 400 ms.

**Note:** If later on your needs grow and you need to add more CO lines or channels into an existing gateway, you will need to delete the existing gateway from the list and create a new one with new trunk or channel numbers.

### External Paging & MOH Functions

If you connect a paging facility to your gateway, click the “External Paging Enabled” check box to enable the Paging function. Assign a specific paging code to be used for announcement via this gateway.

#### Using External Paging

If you assign “111” as the paging code of your VG5000, you need to dial “111” to page a person via the external paging facility on this VG5000. Refer to Chapter 2 Installing InterPBX Communication System/Configuring VG5000 Voice Gateway for more details.

### Editing Analog Gateways

After the addition of the new gateway, you may edit the settings to control the input and output volume of the CO Line and SLT ports.

1. Go to Main Menu>Gateway Configuration>Analogue Gateways.
2. From the “Analog Gateway List” choose the gateway to edit then click on the Modify button.
3. You will see the gateway type, software version running as well as the ports that correspond to CO Line or SLT.
4. If needed, you may change in this screen the name of the gateway, MAC address, CO Line Index Base, Jitter Buffer as well as to enable the Paging feature. For more detailed information please see above section “Creating Analog Gateway List”.

**Configuring CO Line Ports**

1. Go to Main Menu>Gateway Configuration>Analog Gateways.
2. Select a gateway from the “Analog Gateway List” and click the Port Setting button.
3. Select a CO Line port from the “Port List” and click the Port Parameters button to configure the settings for this port.
Setting CO Line Extension Number
An extension number in InterPBX Communication System might represent a CO line, an analog extension, an IP extension, a CO line group or an extension group.

In “Extension Number” it will display the extension number of this CO Line port automatically based on your settings under “CO Line Extension Base”, you may also change the extension number from here. Please select “Enabled” to activate this port. For maintenance purpose, you may click again on “Enabled” to disable this port. In “Description” input the telephone number or other information for this port.

CODEC
Choose the preferred CODEC from PCM, G.723.1 or G.729 supported by InterPBX Communication System. PCM CODEC compresses and decompresses voice conversation to 64 Kbps providing the best speech quality but consumes larger bandwidths. G.723.1 CODEC uses 6.3 Kbps, the less bandwidth from the three choices but may result in poorer call quality and finally G.729, which uses 8 Kbps.

Fax
Enter the extension number of the SLT port where the fax machine is connected. When detecting fax tones, the fax will be automatically transported to the fax machine. You may set all the CO Lines to the same fax extension or to different fax extensions, as you need.

T.38 Fax: InterPBX system also support T.38 protocol allowing fax to be transmitted across the IP networks. When selected, please also select the T.38 setting at SLT port. If you adopt T.38 fax, at the above Fax Ext Number, you may set a local fax extension number, an off-premises fax machine or a fax of Joint Server.

Ring Assignment
Ring Assignment allows you to assign a specific extension, extension group or AA Menu to answer incoming calls from a specific CO line or CO Line Group. In the Ring Assignment field, choose the option of “Local” or “From CO Group”.

If you select “Local”, input a specific extension number, extension group or AA Menu access code to answer incoming calls for this port in Business Hours, Break Hours, After Hours and Closed. The most usual setting is to assign an AA Menu to answer phone calls, taking advantage of the InterPBX's VMS server. To learn more about AA Menu settings, please refer to Chapter 7 Voice Mail Configuration. If you would like to play specific greetings for holidays, select the “Enable Holiday” checkbox so that the system will play the holiday greetings you assigned on the Holiday List. For more details about Holiday settings, please refer to Chapter 3 System Configuration/Holidays.

If you select “From CO Group”, incoming calls will be answered according to the Ring Assignment of the CO Line Group you have assigned. For more details about CO Group’s Ring Assignment, please refer to Chapter 6 Group Management/CO Group.

**Silence Compression**
The silence compression feature reduces network traffic whenever a period of silence is detected during a conversation. When there is silence in a conversation, silence indicator packets will be sent by the system instead of full voice packets to reduce traffic. Select the check box of “Silence Compression Enabled” to enable this function or uncheck to disable it.

**Caller ID**
InterPBX’s CO Line ports have the ability to detect either DTMF or FSK Caller ID modes. You may select this item to enable this function. Please note that your local telephone company must provide Caller ID service in order to display the incoming party’s number.

**Pickup After Rings**
On the “Pickup After Rings” column, input the number of the ring count. If you set the ring count as “2”, the system will pickup the incoming call after 2 rings.

**Flash Timing**
Under the application of Centrex, you can set the Flash Timing on the IP phones through the system. The Flash function was set on the MENU button of the IP phones. After you complete the Flash Timing setting, you can start to use some of the Centrex functions. Please make sure the IP phone’s Flash Timing must be within system’s Flash Timing.
Voice Gain Level
You may edit the input and output volume of:

- Transmit Gain: this field edits the TX Gain of the CO Line ports, or how loud the external party hears your voice.
- Receive Gain: this field edits the RX Gain of the CO Line ports, or the volume of incoming calls.
- DTMF Volume: this field edits the intensity of the DTMF tones in the CO Line ports.

Apply Settings
After finalizing the settings in this port, if you want them to be identical as this one, you may apply the settings to other CO Line ports present in the system, saving you the hassles of configuring each port individually.

1. Click on “Apply Setting to” and a new window will appear displaying the parameters you have entered.
2. Apply or change the settings, as you desire.
3. Enter in the field “Apply to Port” the first and the last CO Line port you would like to be identical to the present settings. If the range selected includes this port, the new settings, if any, will also be applied to this one.

Configuring SLT Ports

1. Go to Main Menu>Gateway Configuration>Analogue Gateways.
2. Select a gateway from the list then click the Port Setting button.
3. Select a SLT port and click the Port Parameters button.

Flash Time
You may set the Flash Timing of the SLT ports to check whether the Hook action (quick press and release of the phone’s hook) on the analog phones
works fine. Analog extensions require the Hook action to transfer, answer call waiting, or create a conference call. Make sure the phone set’s Flash Timing must be within the system’s Flash Timing.

**Caller ID**
The SLT ports have the ability to detect either DTMF or FSK Caller ID modes. You may this item to enable this function. Note that your local telephone company must provide Caller ID service in order to display the incoming party’s number.

**T.38 Fax**
T.38 is a fax relay protocol. It allows fax to be transported across IP networks between fax machines. When selected, please be sure the T.38 is also enabled at the CO line port setting.

**Note:** You may set the SLT port to get a CO line when off-hooked, without the need to pressing the CO line access code first. Go to Main>Extension Management> Analog Port list for more details.

**Voice Gain Level**
You may edit the input and output volume of:
- Transmit Gain: this field edits the TX Gain of the CO Line ports, or how loud the external party hears your voice.
- Receive Gain: this field edits the RX Gain of the CO Line ports, or the volume of incoming calls.
- DTMF Volume: this field edits the intensity of the DTMF tones in the CO Line ports.
- Dial Tone Volume: this field edits the dialing volume of the SLT ports, or the dial tone volume of analog phones.

**Setting Analog Extension Numbers**
Each of the SLT port in the InterPBX can be connected to an analog phone. After setting up the Analog Gateway, go to Main Menu>Extension Management> Analog Port List. You will see all your available SLT ports in this window. Select a SLT port and click the Modify button to configure your analog extension. For more details, please refer to Chapter 5 Extension Management.

**Off-Premises Gateways**
InterPBX Communication System supports Off-Premises Voice Gateways. By installing Voice Gateways in remote offices or satellite offices, enterprises can enjoy great savings on inter-office communication. Also, more savings can be achieved through the remote gateways by utilizing it to place outgoing local calls to local customers, saving this way the long distance charges since through the remote gateway this call is now same as a local call.

Creating Off-Premises Gateway List

After installing your gateway in the remote office, you will need to add it to the PBX Server.

1. Go to Main Menu>Gateway Configuration>Off-Premises PSTNGW.
2. Click the Add button to add a new gateway to PBX Server’s “Off-Premises Analog Gateway List”.
3. Refer to the section above “Analog Gateways” to configure the “Name”, “MAC Address”, “Gateway Type”, “CO Line Ports”, “SLT Ports”, “CO Line Index Base” and “Jitter Buffer Depth”.
4. For Off-Premises Gateways, you will need to enter its “IP Address”. The IP Address assigned to this Off-Premises Gateways has to be a REAL (PUBLIC) IP address. In the Signal Port, InterPBX will automatically preset the port therefore you do not have to make any changes here.
5. Min. Loop Current Drop Time: Normally, when a telephone call terminates, the telephone company sends a momentary drop in loop current to signal the disconnect event. Please enter the minimum Loop Current Drop time in this field to avoid the IP-PBX system detecting the disconnect signal incorrectly. The value should not be too long or too short. Recommended value is 400 ms.
6. If you have connected any type of paging device, select the “External Paging Enabled” to activate this feature, then create and enter a code in the Paging Code field to use this feature.

7. Lastly, if desired, you may activate the QoS feature from your Off-Premises Gateway by enabling it here.

After finishing creating the gateways, you may edit the ports incoming and outgoing volume as described in “Editing Analog Gateways” section above.

**Configuring Off-Premises CO Line Ports**

1. Go to Main Menu>Gateway Configuration>Off-Premises PSTNGW.
2. Select a gateway from the list then click the Port Setting button.
3. Choose a CO Line port and click the Port Parameters button to edit the settings to this port.
4. Refer to the sections above Editing Analog Gateway/Configuring CO Line Ports to configure the “Extension Number”, “CODEC”, “Fax”, “Ring Assignment”, “Silence Compression”, “Caller ID Detection” and “Voice Gain Level” as well as how to duplicate the settings.
5. The system will automatically preset a port to the Media Port. Please do not change this port number.

![Image of CO Line Port Configuration](image)

**Make a call via off-premises CO Line port**

To make a call via your off-premises CO Line port, pick up the phone, dial the remote CO Line extension number then dial the phone number you need to call. For long distance calls, please dial remote site’s long distance access code followed by the phone number to be dialed.
Configuring Off-Premises SLT Ports

1. Go to Main Menu>Gateway Configuration>Off-Premises PSTNGW.
2. Select a gateway from the list then click the Port Setting button.
3. Choose a SLT port and click the Port Parameters button to edit the settings to this port.
4. Refer to the sections above Editing Analog Gateways/Analog Port to configure this port.
5. Leave the Media Port as it is since it is assigned by the system automatically.
6. Each off-premises SLT port can connect to an off-premises analog phone. Go to Main Menu>Extension Management>Analog Port List and you will see all your available SLT ports in this window. Select a SLT port and click on Modify to configure your off-premises analog extension. For more details, please refer to Chapter 5 Extension Management/Analog Extensions.

Make a call to off-premises SLT port

To make a call to extensions connected on off-premises SLT ports, simply pick up the phone and dial the remote SLT extension number normally as you would for calling extensions in your office. You can make calls to remote extensions as the way you do for calling extensions in the same office.
**Recording System**

The InterPBX System supports recording system to improve enterprise productivity and enhance efficiency of business communication. InterPBX Communication System supports two types of recording systems: the built-in recording system which provides Record-on-Demand and/or Store-on-Demand functions and the stand alone professional recording system.

- **Blaze5000 Series**: It provides built-in Record-on-Demand and Store-on-Demand recording functions. When there is no one on the Store-on-Demand list, Record-on-Demand will still be available. Up to 10 concurrent recording channels will be provided.
- **Blaze1200 Series**: It supports built-in Record-on-Demand with maximum 10 concurrent recording channels.
- **Savanna8000 Series**: It doesn’t support built-in recording function and an additional recording system is required.

**Note**: The recording function does not apply to analog and off-premises extensions.

**Setting Store-on-Demand (Available on Blaze5000 Series)**
The built-in Store on Demand function allows the system to record calls for extensions on the Store on Demand list when conversation starts. The extension user can save the recording at any time during a call by pressing the Record button. Recordings will not be automatically saved for the concern of storage space. User may also retrieve the recording when needed.

1. Go to Main Menu>Gateway Configuration> Recording System.
2. Enter the IP address of the PBX Server in the field “IP Address”.
3. The default value of “Port Num” is “10”, which means the system can record up to 10 extensions at the same time. You need to assign up to 10 extensions on the Store-on-Demand List.
4. Define the “Storage Limitation” of the hard disk and the “Auto Purge Days”. When exceeding the hard disk limit, the system will delete the recordings of days as set on “Auto Purge Days”. For example, if the Storage Limitation is “80%” and the Auto Purge Days is “7”, the system will automatically delete previous 7-days recording data when it reaches the storage limitation.
5. Put a check on the item of “Recording Beep” according to your company’s requirement or the Government’s regulation. When the Recording Beep is activated, the caller and the receiver will hear the beep sound every 15 seconds during a call.
6. Recording Flash: You can enable the Recording Flash function. After the Recording Flash is enabled, the recording lamp on the recorded extension will keep flashing until the call is hung up. If you need to record to an extension confidentially, please do not enable this function.
You can add up to 10 extensions in the Store on Demand list. The system will continuously record to these extensions during a call.

1. Go to Main Menu>Gateway Configuration> Recording System.
2. From the “Extension” list, move the extension number to the “Store on Demand” list. The system can record to up to 10 extensions at once.

**Activate the Recording**

1. During a call, press the specific programmable button or Rec/Play button to active the function.
2. After the call is hung up, the system will stop recording.

**Note:** The recording function does not apply to analog and off-premises extensions.

**Playing the Recording**

The extension users can play the recording from their IP phones directly.

1. Press the specific programmable button or REC/Play button or function code #42 to play recordings
2. Enter the password and then press #. (The password for the recording system is same as your mailbox password. The default password is blank.)
3. Follow the system prompts, press 1 to play the latest recording or press 2 to check the recording by entering date and time. Please follow the procedure below to assist you in listening to the recording.

- In the recording system, you can press * anytime to return to the previous menu or back to the main menu.
- When playing recordings, you can press 0 (zero) to skip the time stamp announcement of the recording.
• If your UMS is enabled and the e-mail is set, after playing a recording, you will be prompted to send the recording you just played to your email.
• For Store on Demand recording, you can only play the latest, previous or next recording for the current day.
• When you input the time for searching recording, the system will play the recent recording after the time.

Setting Record-on-Demand (Available on Blaze5000 and Blaze1200 Series)

The built-in Record on Demand allows the system to record calls after users pressing the Record button. Calls will be recorded and saved after pressing the Record button. User may also retrieve the recording when needed.

1. Go to Main Menu>Gateway Configuration> Recording System.
2. Enter the IP address of the PBX Server in the field “IP Address”.
3. The default value of “Port Num” is “10”, which means the system can record up to 10 extensions at the same time. You may set the permission from extensions’ Class of Service.
4. Define the “Storage Limitation” of the hard disk and the “Auto Purge Days”. When exceeding the hard disk limit, the system will delete the recordings of days as set on “Auto Purge Days”. For example, if the Storage Limitation is “80%” and the Auto Purge Days is “7”, the system will automatically delete previous 7-days recording data when it reaches the storage limitation.
5. Select the “Recording Beep” checkbox according to your company’s requirement or the government’s regulation. When the Recording Beep is activated, the caller and the receiver will hear the beep sound every 15 seconds during a call.
6. Recording Flash: You can enable the Recording Flash function. After the Recording Flash is enabled, the recording lamp on the recorded extension will keep flashing until the call is hung up. If you need to record to an
extension confidentially, please do not enable this function.

**Activate the Recording**
1. During a call, press the specific programmable button or Rec/Play button to activate the function.
2. After the call is hung up, the system will stop recording.

**Note:** The recording function does not apply to analog and off-premises extensions.

**Playing the Recording**
The extension users can play the recording from their IP phones directly.
1. Press the Play button to play the recording. On programmable button supported phones, you can press a specific programmable button that is assigned via Button Mapping to play the recording. Or press the REC/PLAY button, if applicable.
2. Enter the password and then press #. (The password for the recording system is the same as your mailbox password. The default password is blank.)
3. Follow the system prompts, press 1 to play the latest recording or press 2 to check the recording by entering date and time. Please follow the procedure below to assist you in listening to the recording.
In the recording system, you can press * anytime to return to the previous menu or back to the main menu.

When playing recordings, you can press 0 (zero) to skip the time stamp announcement of the recording.

If your UMS is enabled and the e-mail is set, after playing a recording, you will be prompted to send the recording you just played to your email.

For Store on Demand recording, you can only play the latest, previous or next recording for the current day.

When you input the time for searching recording, the system will play the recent recording after the time.

Setting Professional Recording System

InterPBX Communication System can also work with the professional recording system Blaze Logger that provides more powerful recording and searching functions.

1. Go to Main Menu>Gateway Configuration>Recording System.
2. On the “IP Address” field, enter the IP address of the recording system.
   You don’t need to set the other items.

The Port Num will display the license numbers of your Blaze Logger. Please refer to the Blaze Logger Administration Guide for more information.
Chapter 4 Gateway Configuration

Digital Line Gateways

When adopting VG6000 or VG7000 Digital Gateway, your InterPBX IP Communication System, including Blaze, Savanna series IP-PBX, needs to enable the relevant function as well.

Adding Digital Gateway

You need to set the MAC address of Digital Gateway to the PBX Server.

1. Go to Main Menu>Gateway Configuration>Digital Line Gateways, the “Digital Gateway” list will be displayed.
2. Click the Add button to create a Digital Gateway. Before you add a new gateway, make sure the relevant License Keys has been added.
3. Input the required data of this gateway.

- **Name**: Enter a name to describe this Digital Gateway.
- **MAC Address**: Enter the MAC address of the Digital Gateway. You may find the MAC address from the device label.
- **Trunk Number**: Enter the available trunk numbers you may use or purchase.
- **CO Line per Trunk**: Input the available channels of each trunk. For example, input 32 for E1 or 24 for T1.
- **CO Line Extension Base**: Input the starting number of CO Line extension number. The CO Line extension numbers will be assigned automatically. For example if you set it as 4000 for an 8-port FXO gateway, the CO Line extension number of this gateway will be 4000-4007.
- **Jitter Buffer Depth**: From “Jitter Buffer Depth” drop down menu, depending on your bandwidth and CODEC, start with a Jitter Buffer with the minimum value. Jitter Buffer will dynamically adjust its value according to the party you are talking to and the bandwidth of the call but never below the value you have set. The higher value of Jitter Buffer will reduce the chance of packet loss during a call but might cause delays and the lower value will help to transmit the voice packets faster but might cause packet loss problems.
- **Off-Premises and IP address**: For Off-Premises Gateways, you will need to select the check box and enter its “IP Address”. The IP Address assigned to this Off-Premises Gateways needs to be a REAL (PUBLIC) IP address.
Note: If later on your needs grow and you need to add more trunks or channels into an existing gateway, you will need to delete the existing gateway from the list and create a new one with new trunk or channel numbers.

Modifying a Digital Gateway

To modify, select a gateway from the list and click the Modify button. You may change the name, MAC address or Jitter Buffer Depth. The software version of the gateway will also be displayed here.

Setting Trunk Parameters

You can set the parameters of each trunk to adjust the voice quality and other items.

2. Select a gateway you would like to edit from the list and click the Trunk button to edit the trunk parameters.

- **Trunk**: Select a trunk you would like to edit from the list.
• **CO Line Index Bass**: Input the starting number of the CO line display number. For example if you set it as 1, when there is an incoming call from port 1, the IP phones will display “Incoming Call CO 1” instead of the extension number of the port, helping you to identify the CO lines being used.

• **Transmit Gain**: This field edits the TX Gain of the trunk, or how loud the external party hears your voice. The value here should conform to the one on Digital Gateway. The default value is 5.

• **Receive Gain**: This field edits the RX Gain of the trunk, or the volume of incoming calls. The value here should conform to the one on Digital Gateway. The default value is 5.

• **DTMF Volume**: This field edits the intensity of the DTMF tones in the trunk. The value here should conform to the one on Digital Gateway. The default value is 5.

• **Tie Line**: Select Tie Line check box if you would like to establish a tie to connect the Digital Gateway to other PBX using this trunk.

### Setting Trunk Port Parameters

You can edit the ring assignment and other settings for individual ports or channels of a trunk.

2. Select a gateway from the list and click the Port Setting button.
3. Select a port from the port list and click the Port Parameters button to edit.

• **Extension Number**: The extension number will be displayed automatically based on your settings under “CO Line Extension Base”. You may also change the extension number here.
• **Description**: Input the telephone number or other information for this port.

• **Enable**: For maintenance purpose, you may need to temporarily disable some channels. Click the “Enable” check box to enable the channel or uncheck to disable it.

• **CODEC**: Choose the preferred CODEC from PCM, G.723.1 or G.729 supported by InterPBX Communication System. PCM CODEC compresses and decompresses voice conversation to 64 Kbps providing the best speech quality but consumes larger bandwidths. G.723.1 CODEC uses 6.3 Kbps, the less bandwidth from the three choices but may result in poorer call quality and finally G.729, which uses 8 Kbps.

• **DID (Direct Inward Dialing)**: When selected, incoming calls will be connected to extensions directly in stead of been picked up by Auto Attendant. This function is available when you have subscribed the DID service provided by your carrier. When you would like to set a direct line between two PBXs connected by Tie Line, you may also enable this function.

• **Strip Prefix Digit**: When the length of the number sent by DID is longer than your extension numbers, set the digit that needs to be removed. For example, if the DID prefix and adding number is 660x-xxxx, but you have 4-digit extension number, set “1” to delete the first digit sent by your carrier. The default value is 0.

• **Fax Ext. Number**: The InterPBX System can detect the fax signal of CO Line. Please enter the extension number of your fax machine here. When the InterPBX System detects a fax signal, the fax will be transferred to the fax machine.

• **Ring Assignment**: Ring Assignment allows you to assign a specific extension, extension group or AA Menu to answer incoming calls from a specific CO line or CO Line Group. In the Ring Assignment field, choose the option of “Local” or “From CO Group”.
  - **Local**: If you select “Local”, input a specific extension number, extension group or AA Menu access code to answer incoming calls for this port in Business Hours, Break Hours, After Hours and Closed. The most common setting is to assign an AA Menu to answer phone calls. If you would like to play special greetings for holidays, select the “Enable Holiday” checkbox and the system will play the assigned greeting from Holiday List. For more details about AA Menu, Voice Mail and Holiday settings, please refer to InterPBX Administrator
Installation and Configuration Guide.

- **From CO Group**: If you select “From CO Group”, incoming calls will be answered according to the Ring Assignment of the CO Line Group you have assigned. For more details about CO Group’s Ring Assignment, please refer to Administrator Installation and Configuration Guide.

- **Silence Compression Enable**: Silence Compression reduces network traffic whenever silence is detected during a conversation for a specific amount of time. Silence indicator packets will be sent out instead of full voice packets to reduce traffic.

- **Caller ID Enabled**: InterPBX’s CO Line ports can detect both DTMF and FSK Caller ID. You may select or deselect it per your preference. Please note that before using this function, the Caller ID service must be provided by your local telephone company.

- **Pickup After Rings**: Please enter the Pickup After Rings times for the CO Line port. If you set the ring times as 2, the system will pick up a call after it rings twice and then transfer the call as settings.

- **Flash Timing**: You may set the Flash Timing of the SLT ports to check whether the Hook action (quick press and release of the phone’s hook) on the analog phones works fine. Analog extensions require the Hook action to transfer, answer call waiting, or create a conference call. Make sure the phone set’s Flash Timing must be within the system’s Flash Timing.

**Apply Settings**

After finalizing the settings of this port, if you want the other ports to be identical as this one, you may apply the settings to the others, saving you the hassles of configuring each port individually.

1. Click on “Apply Setting to” and a new window will appear displaying the parameters you have entered.
2. Apply or change the settings as you need.
3. Enter in the field “Apply to Port” the first and the last port you would like to be identical to the present settings. If the range selected includes this port,
the new settings, if any, will also be applied to this one.

After you complete the installation of a new digital gateway, you need to set up the associated Route and ARS. For more details, please refer to System Configuration>Route and Group Management>Class of Service>ARS.

**CTI Gateway**

DSG BlazeLink is a CTI solution software enables business to develop applications to optimize workforce in a call center or contact center. Applications developed on BlazeLink for call control, monitoring, or managing are able to connecting DSG IP-PBX System to the data processing environment.

1. Go to Main Menu>Gateway Configuration> CTI Gateway.
2. In the field “IP Address”, please enter the IP address of your CTI gateway.
3. Agent License: It displays the Agent License numbers you purchased. If you purchase 100 Agent Licenses, the system will allow the 100 agents to log in. This function will be released in a later version. The Agent License is
Chapter 4 Gateway Configuration

currently set at BlazeLink.

NAT Proxy

With DSG NAT Proxy, Blaze or Savanna IP PBX Communication Systems and Off-Premises Phones can traverse more than 95% of NAT or firewall successfully without creating other service ports, and therefore enterprises can communicate via IP-PBX Communication System with pure and stable voice quality.

1. Go to Main Menu>Gateway Configuration> NAT Proxy.
2. On the field “IP Address of Master Proxy”, please enter the IP address of the main NAT Proxy Server.
3. If you have the other NAT Proxy server, please also enter its IP address on the field “IP Address of Slave Proxy”.

Note: When the NAT Proxy Server is activated successfully, a symbol @ will be shown beside the “IP Address of Master Proxy” field to represent that the NAT Proxy Server is working.
For more information about the NAT Proxy Server’s settings and operation, please refer to the NAT Proxy Administration Guide.

**SIP Proxy**

SIP (Session Initiation Protocol) is a new generation multimedia communication protocol. It enables voice and other media types such as video to be transmitted over the IP networks. With DSG SIP Proxy, your Blaze or Savanna series IP-PBX will be able to adopt the SIP Trunk services provided by ITSPs or deploy SIP phones as your extensions.

**Add SIP Proxy**

You need to enroll the SIP Proxy IP address to the PBX Server.

1. Launch the web browser. On the address bar, enter your PBX Server’s IP address. (Please use service port 88.)
2. Click on the Administrator icon and use your user name and password to login.
3. Go to Main Menu>Gateway Configuration>SIP Proxy, the “SIP Proxy” list will be displayed.
4. Click the Add button to create a SIP Proxy.

![Add SIP Proxy](image)

- **Name**: Enter a name to describe this SIP Proxy.
- **MAC Address**: Enter the MAC address of the SIP Proxy. You may find the MAC address from SIP Proxy’s administration web site or from the device.
- **Port Number**: Enter the available SIP channels you may use or purchased.
- **Extension Base**: Input the starting number of SIP Trunks. The number shall be a unique one from others on the system such as extension numbers, group numbers or pickup codes. Notice that the SIP Trunk numbers will not be displayed on the IP phone when accessed. Instead, the associated SIP account number will be represented.
- **Off-Premises**: When SIP Proxy is not located in the LAN where PBX Server is located and is deployed on the Internet, select the “Off-Premises” check box and input its real IP address assigned to SIP Proxy.

**Add SIP Trunks**

If you have subscribed the SIP Trunks service provided by your carrier, you can input your account information at this section. Before you start, please prepare the followings.

- **Account and Password**
- **Registrar Server IP address**
- **Domain Name (In some cases, it is the same as the registrar Server)**
- **Specific service port used by your carrier**

1. Go to Main Menu>Gateway Configuration>SIP Proxy.
2. Select a SIP Proxy from the list and click the Account button.
3. Click the Add button to create a SIP Account.

![Add Account to SIP Proxy: SIP Trunk](image)

- **Account Name**: Assign a name for this SIP Trunk. When accessed, the name will be displayed on IP phone.
- **Phone Number**: The account provided by your carrier.
- **Registrar Server IP**: The IP address of your carrier’s Registrar Server.
- **Port**: The service port provided by your carrier. The default setting is 5060.
- **Domain Name**: The domain name provided by your carrier. In some cases, it is the same as the Registrar Server.
- **User ID**: The account provided by your carrier. In some cases, it is the same as Phone Number.
- **Password**: The password provided by your carrier.
- **Registrar Server Type**: Depending on your carrier’s equipment, select a type of Registrar Server from the list.
- **Register Expire Time**: Please input the expiring time of registration. Once registration succeeds, the system will follow the required time of your carrier to reset this item.
- **CODEC**: Select a CODEC supported by the SIP Trunk.
- **Enabled**: Click this item to enable the SIP Proxy.
- **RTP Relay**: When the network is behind NAT, select this item to enable RTP Relay allowing voice packets to traverse NAT and firewall automatically.
- **DID and Strip Prefix Digit**: If you have applied DID service provided by your carrier, select the DID check box and input the digit of prefix that needs to be deleted.
• **Fax Ext Number:** If your carrier supports SIP fax, you may input the extension number connected to your fax machine. When a fax comes from SIP Trunks, it will be transferred to the fax machine.

• **Ring Assignment:** You could assign a specific extension, extension group or AA menu to answer incoming calls. In ring assignment, select “Local” to set individual trunk line or select “From CO Group” to allow this trunk referring to a CO Line Group. For more details about ring assignment, please refer to InterPBX Administrator Installation and Configuration Guide.

**Create SIP Trunk ARS**

After you add SIP Trunks to your system, you need to edit the associated ARS (Automatic Route Selection) policies and procedures allowing extensions to access those SIP Trunks for making calls. Notice that extensions are not able to access SIP Trunk using the assigned extension number.

**Creating Routes:**
1. Go to Main Menu>System Configuration>Route.
2. Click the Add button to increase a route.
3. Input the description of this route.
4. In different business hours, click “Account” and select a SIP Proxy you would like to use from the list.
Setting ARS:
1. Go to Main Menu>Group Management>Class of Service.
2. Select a Class of Service from the list and click the ARS button to apply the route you just created to the selected CoS.
3. In “Dialed String”, define your policy and procedure. You may input the commonly used international access code and country code, such as “01144” for calls to UK, or long distance access code and area code, such as “1212” for calls to New York. You may also set a specific number representing the SIP Trunk code such as “7”. The digit length shall not exceed 28 and the string shall be a unique one.
4. Set the Minimum and Maximum Digits length of dialed numbers.
5. From the “Route” list, select a SIP Trunk route you would like to adopt.
6. When editing SIP Trunk ARS, skip the “Delete” and “Insert” items.
7. Click the Add button to increase this policy and procedure to ARS list. To remove, select one from the list and click the Remove button.
8. You may also click the “Copy from” button and increase one by modifying others.

Making Calls via SIP Trunks
1. Lift the handset. Please make sure your Class of Service is allowed to access SIP Trunk ARS.
2. Depending on the ARS policies and procedures you created, input the SIP Trunk access code followed by the phone number, or simply input the phone number. Press the # key to start dialing.
For example, if you set “7” in “Dialed String” as the SIP Trunk access code, when you dial 7-44-2055551234#, the call to UK will get through your SIP Trunk. If you set “01186” in “Dialed String”, when you dial 011-86-xxxxxxxxxxx#, all calls to China will get through the SIP Trunks.

Add SIP Extensions

SIP Proxy helps you to register third parties’ SIP phones to your InterPBX System and they will be one of your extensions.

1. Go to Main Menu>Extension Management >SIP Phone List.
2. Click the Add button to increase SIP phones.
3. Assign an extension number in “Extension No.”.
4. Assign a password for this extension. Extension users can change the password from InterClient. Password must be numbers.
5. From “SIP Proxy” list, select one for this SIP phone.

For more information about the SIP Proxy Server’s settings and operation, please refer to the SIP Proxy Server Administration Guide.
This chapter describes the installation and configuration of the various types of extensions supported by InterPBX Communication System. These include IP phones, soft-phones, virtual extensions, analog phones, and off-premises extensions. You may at your own evaluation provide the appropriate type of extension according to the user’s needs, whether they are at the head office, branch offices or business travelers.
Configuring IP Phone, SoftPhone and SoftConsole

You may create extensions via Auto Discovery or manually. For more information about settings, please refer to Chapter 2 Installing InterPBX Communication System.

Go to Main Menu>Extension Management>InterPhone List. Select one of the extensions from the list and then click the Modify button to edit.

Extension No., Phone Type and MAC Address

If the extension is registered to the PBX server using Auto Discovery, these info will be recorded automatically. You do not need to change them.

If you need to increase extensions manually, please set these info. For software phone and console, the MAC address is the one of user’s PC.

User Information

Fill in the information about extension users, including First Name, Last Name, Title, Location and Department. The user’s name will be displayed on the IP phone’s LCD and the extension list. The information about extension users will be applied to Caller ID of internal calls.

User Password

With User Password, administrators are allowed to modify the password for logging in InterPBX Management Website. With Auto Discovery, the extension password is blank. In case password is forgotten, the system allows setting password back to default through Password Management. This is located at Main Menu>System Configuration>Password Management.
Off-Hook Access to
InterPBX provides extensions different Off-Hook selections that enable users directly off-hook access to extension, CO line, operator, or hotline. Please select a direct access port for your extension from the “Off-Hook Access to” list.

- **Intercom**: When you lift the headset, you can dial to an extension directly. To dial the CO line number, you have to press the CO line access code first.
- **CO Line**: When you lift the headset, it will retrieve the CO line access code automatically. To dial the extension number, you have to press the Intercom button first.
- **Operator**: When you lift the headset, it will directly connect to the operator. For the settings of operator, please refer to Chapter 6 Group Management.
- **Hotline**: When you lift the headset, it will automatically connect to a specific extension or extension group. Please enter the hotline number in the field of Hotline.

Class of Service
Assign one Class of Service to restrict extensions to make unnecessary calls. With Auto Discovery, the settings in Class of Service will be applied to the extension automatically.

Calling permission on Class of Service will be performed with top priority. Please refer to Chapter 6 Group Management/Class of Service for more details about call permission.

Button Mapping Group
Select a “Button Mapping Group” which supports programmable keys on IP phone as feature access buttons to enable speed dialing, call park and to access CO lines. You can set Button Mapping Group as a template to be adopted by the IP phones.

For more information about Button Mapping, please refer to Chapter 6 Group Management/IP Phone/Button Mapping and Chapter 5 Extension Management/Button Mapping Groups for more details about Button Mapping Extension.

CODEC
From the CODEC list, select PCM format G.711 that compresses a call to 64 Kbps, providing higher call quality but using higher bandwidths. Other alternatives include G.723.1 compressing to 6.3 Kbps consuming smaller bandwidth, but with poorer voice quality and G.729 compressing to 8 Kbps.

**Jitter Buffer Depth**

From “Jitter Buffer Depth”, select “Low” based on the bandwidth and CODEC. Jitter Buffer will be adjusted to accommodate the callers and bandwidth, but the Depth will never be lower than the “Low” standard. The Depth of Jitter Buffer can be specified as None, Low, Middle and High. The None buffer corresponds to 0 packet, Low to 2 packets, Middle to 4 packets and High to 6 packets. Higher Jitter Buffer Depth has more storage capacity when storing temporarily the packets avoiding packets to be discarded, but additional delay may be bothersome at times. On the contrary, smaller Depth offers faster transmission but excessive number of packets might be lost during transmission. For example, G.711 format transmits packets every 20 ms which probably leads to 40 ms additional delay if you set the Jitter Buffer as Low.

**Silence Suppression**

Silence Suppression reduces network traffic whenever silence is detected during a conversation for a specific amount of time. Furthermore, silence indicator packets will be sent out instead of full voice packets to reduce traffic. Select “Silence Suppression” to enable this function or uncheck to disable it.

**Conference Disabled**

Conference allows the extension making an outgoing call to initiate a conference between three or more parties. Select the “Conference Disabled” check box if the user does not need this function.

**To initiate a conference call**

1. During a call, press the Hold button to hold a party.
2. Call the second party where this can be an internal extension or an external line. More parties can join this conference by pressing the Hold button to hold the conversation and dialing to the next party. You can hold up to 4 parties.
3. After the call is connected, press the Transfer button and “#40” to start a Conference.
4. To invite more parties, hold this conference and dial to the new party. When connected, press the Transfer button and “#40” to start.

Each conference allows for up to 18 parties in different sessions. But each session requires one party to be a direct extension under the InterPBX.

Note: Off-premises extensions are not allowed to initiate a conference but can be invited.

Paging Disabled
You may select “Paging Disabled” when this is not needed.

How to activate External Paging
1. Pick up the handset.
2. Dial the Paging Code assigned on the gateway where the Paging equipment is connected.
3. Broadcast through the handset or the speakerphone on the phone.
4. After broadcasting, hang up the phone.

How to activate Internal Paging
1. Pick up the handset.
2. Dial #38.
3. Dial the pre-defined extension number or Extension Group Code for internal paging.
4. Broadcast through the handset or the speakerphone on the phone.
5. After broadcasting, hang up the phone.

Enable/Disable Extension
For maintenance purpose, you may need to temporarily disable some extensions. Click the “Enable” check box to enable the extension or uncheck to disable it.

Apply Settings
After configuring one extension, apply the settings to other extensions to save time on initial setups. Press “Apply Setting to” and a window will pop up showing all your pre-defined settings. Apply or change the settings, as you need. Enter the range of extensions in “Apply to Extension”. If the range
selected includes the extension you are using, the new settings will also be
applied to this extension.

**Button Mapping**

Under Button Mapping, set programmable keys on IP phones as feature access
dr buttons to enable Speed Dialing, Call Park, Access CO Lines or Conference.
Depends on phone types, the available number of Programmable Buttons are
different.

1. Go to Main Menu>Extension Management>InterPhone List.
2. Select one of the extensions from the list and press Button Mapping to edit
the programmable keys. Please refer to Action List below, which lists
details about the action, numbers and functions required.
3. Select one programmable button and scroll down to the preferred Action.
4. After the action is selected, enter the correspondent number in the
"Number" field. For example, if you select “System Speed Dial” from the
“Action” drop down menu and enter a speed dial number. When you press
this programmable key, the system will dial the phone number assigned on
this system speed dial number. If the “Number” field is left blank, press the programmable key and followed by any speed dial number you desire.

<table>
<thead>
<tr>
<th>Action</th>
<th>Designated Number</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Access Ext.</td>
<td>An extension number of a station, extension group, CO line, CO line group, or an outside phone number</td>
<td>To make calls to the assigned number. Please note to add CO line access code, e.g. 05551234 for an outbound call. Notice that if Trunk Group is set on a button, when parts of trunks are available, the LED displays available status for other people.</td>
</tr>
<tr>
<td>After Call Work</td>
<td>-</td>
<td>To allow agents to have a period of after call work time between two calls.</td>
</tr>
<tr>
<td>Ask Member Login</td>
<td>An Extension Group Number</td>
<td>Check the login status of the extension group. Hearing a dial tone means members are logged-in successfully and a busy tone means not logged-in.</td>
</tr>
<tr>
<td>Auto Line Access</td>
<td>-</td>
<td>To get a CO Line</td>
</tr>
<tr>
<td>Auto-In</td>
<td>-</td>
<td>Agents’ extensions will pick up the next call automatically.</td>
</tr>
<tr>
<td>Auxiliary Time</td>
<td>-</td>
<td>Agents can enable this function to notify the system stop assigning incoming calls to their extensions so that the agents can leave their seats temporarily without logging out the system.</td>
</tr>
<tr>
<td>Action</td>
<td>Designated Number</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------------</td>
<td>-------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Call Appearance</td>
<td>-</td>
<td>Press to make or receive calls. It functions similar to intercom. When more than one button is assigned, buttons with lowest number will take the first call.</td>
</tr>
<tr>
<td>Call Hold Retrieve CO</td>
<td>An extension number of a CO line</td>
<td>To retrieve an incoming call from a specific CO line placed on hold. (You should be able to see the CO line extension number. For using this function.) If this CO line is set on the programmable button, users can directly press the flashing button to retrieve the incoming calls.</td>
</tr>
<tr>
<td>Call Hold Retrieve Ext</td>
<td>An extension number of a station</td>
<td>To retrieve an internal call placed on hold. If this extension number is set on the programmable button, users can directly press the flashing button to retrieve the call.</td>
</tr>
<tr>
<td>Call Park</td>
<td>A slot number (0-9)</td>
<td>To park a call to a specific slot.</td>
</tr>
<tr>
<td>Call Pickup CO Line</td>
<td>-</td>
<td>To answer the least recent incoming call ringing on the system.</td>
</tr>
<tr>
<td>Call Pickup Directed</td>
<td>An extension number</td>
<td>To answer a call ringing at another extension.</td>
</tr>
<tr>
<td>Conference Call</td>
<td>-</td>
<td>Start Conference with the callers placed on Hold.</td>
</tr>
<tr>
<td>Action</td>
<td>Designated Number</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------</td>
<td>-------------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Forward All Calls</td>
<td>An extension number + #</td>
<td>To forward all the incoming calls to a specific extension or an external phone number automatically. Press again to disable. When setting external number, please add the CO line access code such as 0.</td>
</tr>
<tr>
<td>Headset</td>
<td>-</td>
<td>Press to allow voice been transmitted from the attached headset, instead of the handset.</td>
</tr>
<tr>
<td>Internal Paging</td>
<td>An extension number or extension group number</td>
<td>Broadcast for members through the extension or the extension group.</td>
</tr>
<tr>
<td>Manual-In</td>
<td>-</td>
<td>To allow agents to pick up the next incoming call manually by pressing the specific button of Manual-In function.</td>
</tr>
<tr>
<td>Member Login</td>
<td>An Extension Group Number</td>
<td>Login to be one of the members in the Extension Group.</td>
</tr>
<tr>
<td>Member Logoff</td>
<td>An Extension Group Number</td>
<td>Logout from the Extension Group.</td>
</tr>
<tr>
<td>Personal Speed Dial</td>
<td>A personal speed dial number(e.g. 00)</td>
<td>To dial a number defined on the Personal Speed Dial Number.</td>
</tr>
<tr>
<td>Record on Demand</td>
<td>-</td>
<td>Press to save the recording or start recording. When recording, the LED will be flashing.</td>
</tr>
<tr>
<td>Retrieve Msg</td>
<td>-</td>
<td>To access mailbox.</td>
</tr>
<tr>
<td>Retrieve Record</td>
<td>-</td>
<td>To play the saved recordings.</td>
</tr>
<tr>
<td>System Speed Dial</td>
<td>A system speed dial number(e.g. 000)</td>
<td>To dial a number defined on the speed dial number.</td>
</tr>
<tr>
<td>Action</td>
<td>Designated Number</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------</td>
<td>----------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Transfer to Ext VM</td>
<td>An extension number</td>
<td>To transfer a call to the extension’s voice mailbox.</td>
</tr>
<tr>
<td>Transfer to AA Tree</td>
<td>An AA menu access code</td>
<td>To transfer a call to the AA menu.</td>
</tr>
<tr>
<td>Virtual Extension</td>
<td>A virtual extension no.</td>
<td>Press to act as the assigned virtual extension to make calls.</td>
</tr>
</tbody>
</table>

**Note:** A symbol “Lock” will show on the screen when the programmable keys are assigned as Group’s Button Mapping and only the unlocked ones can be used. Please refer to Chapter 6 Extension Management/Button Mapping for more details about Button Mapping and Appendix A for Function Codes List.

**Station Speed Dialing**
InterPBX Communication System provides System Speed Dialing and Station Speed Dialing functions. You can assign frequently dialed phone numbers at “Station Speed Dialing” for individual use. For more details about System Speed Dialing, please refer to Chapter 3 System Configuration/System Speed Dialing.

1. Go to Main Menu>Extension Management>InterPhone List.
2. Select one of the extensions from the list and press Speed Dialing button.
3. From the “Station Speed Dialing” list, select one entry.
4. Enter the targeted phone number or extension number in the "New Number" box.
5. Give details about New Number in the “Comment” box.
6. Click the Assign button to save.
You can create up to 100 entries. If you would like to set an outside phone number, please remember to add the CO line access code such as “0” in front of the phone number (e.g. 05551234). For long distance calls, please add the long distance code (e.g. 02125551234). For international calls, please add the international code and country code (e.g. 001188628861558).

**Making Calls using Personal Speed Dialing**

1. Lift the handset or press the Speakerphone button.
3. Press the specific personal speed dialing number, such as “00”.

**Answer Option**

You can better manage the incoming calls by setting the forward targets for all calls or when not available.

1. Go to Main Menu>Extension Management>InterPhone List, select an extension from the list and click the Answer Option button to select Call Forward.
2. Select All Calls, Busy or Ring-No-Answer as per your preference.
3. Select one of the targets to answer incoming calls.
   - **Extension**: When selected, calls will be forwarded to the other extension or an external phone number. Enter the target extension number or phone number in the “Number” box.
     **Note**: When setting an external phone number, remember to input your CO line access code.
   - **Voice Mail**: When selected, incoming calls will be sent to your voice mailbox. Note: Notice that your administrator needs to set a DTMF button in the Transfer Option menu allowing callers to leave messages.
• **Auto Attendant:** When selected, incoming calls will be forwarded to the voice mail system, but callers cannot leave messages.

4. **Ring-No-Answer Timeout:** When ring-no-answer, the incoming calls will be redirected to the Ring-No-Answer forward target after the defined timeout. The unit used here is in seconds.

5. **Call Waiting:** You may enable the “Call waiting” function to pick up the second incoming call during a call. After Call Waiting is enabled, if your line is busy, the second incoming caller will hear the normal ringing tone, and you will hear the “beep” tone indicating that you have a call waiting. The “Call Waiting” function can allow only one more incoming call to stay on queue.

**Note:** If the forward destination is busy or no answer, the forwarded calls will be redirected to the original destination’s busy or no answer setting.

If Personal Answering Option is not configured, the calls will be directed to Auto Attendant when the requested extension is busy or not available. The administrator needs to configure the Transfer Option of VMS for taking calls. Please refer to Chapter 7 Voice Mail Configuration/Transfer Options for more details.

**Enable Call Forward from Phone:** Users may also press #44 + forward number +# from IP phone directly to enable Forward All Call function.

**External Call Forward:** To allow the External Call forward functioning well, both internal callers and receivers’ COS need to support external calls. To allow external calls been forwarded to external numbers, the administrator needs to proper set the Trunk to Trunk Class of Service. For more details, please go to Chapter 6 Group Management/Class of Service.

**Mailbox**

In Mailbox, users can select language options, Message Play Priority and Message Forward.
1. Go to Main Menu>Extension Management>InterPhone List, select an extension from the list and click the Mailbox button to edit the mailbox preferences.

2. Select a language from the “Language” list and the system prompts will be announced in the language you have selected. InterPBX can support up to 4 languages, please consult your distributor regarding multiple language options.

3. Set Message Play Priority which allows users to select “Last-In-First-Out” for playback of the latest message first or “First-In-First-Out” to play the oldest one first.

4. Enter the extension number in “Message Forward to” to allow the system to forward messages to pre-defined mailboxes.

5. Play Time Stamp: If “Play Time Stamp” is enabled, the system will play the message recorded time when you play the message.

6. Call Screen: This function allows you to filter callers before answering calls. When enabled, the voice mail system will require callers to say the name, hold the call, and then make calls to the receiver and announce the name of the caller. Receivers will be offered options to answer or not to answer the call. If the receiver chooses to answer, the call will be connected. If not, the voice mail system will reply the caller that the receiver is not available.

7. External Forward: When you are not available, incoming calls will be bounced back to the VMS. Depending on the Transfer Options settings, callers can connect to the operator, voice mail or a desired external phone number if this option is enabled by the Administrator. Set your external forward number here allowing the VMS to forward your incoming calls. You may also set the external forward number from your IP phone.

How to retrieve messages
1. From a phone set, press the “Message” button or “##” if messages are retrieved via an analog phone.
2. Enter the password of the voice mailbox followed by the “#” sign (on its first initial use, no password is required as the default password is null).
3. To retrieve messages or modify personal options simply follow the voice guidance prompts.

**Notification**

InterPBX provides options for message notification. Go to Main Menu>Extension Management>InterPhone List, select an extension from the list then click the Notification button.

**Internal Notification**

The Internal Notification is Ring Notification. When the extension receives new messages, the system will notify the receiver by ringing the extension. This function can be used on different kinds of extensions such as assistant extensions, analog phones, and virtual extensions.

1. **Internal Notification Method:** Select “Extension” from the drop-down menu to enable the internal notification.
2. **Setting the Notification Target:** If you enable the internal notification, input the desired extension number. The system will ring the extension number you assigned when receiving new messages. If you want to enable your station’s notification function, please input your extension number or simply leave it blank.
External Notification

1. Click the “Enabled External Notification” check box to enable the External Notification function, or click the “Urgent Messages Only” check box to notify you only when receiving urgent messages. The notification of urgent messages will be delivered even when you disable the external notification.

2. Set the “Notification Schedule” to deliver notification during business hours or specified hours to avoid interruptions.

3. Set the notification sequence. Set the destination type and its associated phone number or pager ID. You don’t need to add the CO line access code such as “0” here.

4. Set the interval and trial times. The system will repeat the notification, if failed, after the interval period.

The system will send notification to the destination’s phone/pager numbers from 1 to 5. If the number in sequence 1 is busy or unavailable, the system will retry after the interval time you set. If failed, the system will start from sequence 2 and repeat until it reaches the “Try Times.”

Note: If the Answer Options of the destination’s phone or extension is set, the notification will also follow the settings.

Note: Sending a notification via phone or pager is limited by your call permission in the Class of Service. Please consult your administrator for your Class of Service.

Unified Messaging

You may enable the unified messaging function allowing the system to forward the voice messages to your e-mail. The new voice messages will be as saved as a WAV file and sent to your e-mail address.

1. Select “Enable UMS”

2. Select “Keep as new” or “Save as old” after messages are forwarded to e-mail.

3. Input the e-mail address in “E-mail”. You can input up to 3 different e-mail addresses.

When the “Unified Messaging” is enabled, if you select “Save as old”, the internal and external notifications won’t be delivered due to all new messages will be switched to old messages.
Note: If you are on the recording list, when Unified Messaging is enabled, you will be prompted for asking if you want to send recordings to e-mail when finish listening recordings.

Note: Set the SMTP Server when Unified Messaging is enabled. For more details about SMTP, please refer to Chapter 3 System Configuration/General Parameters/SMTP Server.

**Distribution List**

Users can record or forward voice messages to a personal voice mailbox or members in Distribution List. Users can create the Distribution Lists based on his preferences:

1. Go to Main Menu>Extension Management>InterPhone List.
2. Select an extension from the list and then click the Distribution List button.
3. Select a List ID from the list.
4. Select extensions from the Non-Member List and move to Members List.

InterPBX allows editing up to 9 groups for each extension in Distribution List and each can hold up to 15 members.

![Distribution List for Extension 199](image)

**Virtual Extensions**

You can associate Virtual Extensions with regular extensions offering one extension with multiple numbers to be used for different purpose. The regular extensions to be applied need to have programmable buttons.

1. Creating Virtual Extension Number. Go to Main Menu>Extension Management>InterPhone List. Click the Add button and assign the
extension number and password. From the Phone Type drop down menu, select “Virtual”. You do not need to set the MAC address. Refer to the above Chapter 3 Extension Management/InterPhone to set other items.

2. Associating with Extensions. Go to Main Menu>Extension Management>InterPhone List. Select one extension from the list and click the Virtual Ext button. You will see available virtual extension number list. Select one or more to be associated with the regular extension to the Allowed Virtual Ext list. The amount of Virtual Extensions you could assign should not exceed the number of programmable buttons on the phone.

3. Button Mapping. Go to Main Menu>Extension Management>InterPhone List. Select the extension you just edit and click the Button Mapping button. Pick one programmable button and set its action as “Virtual Ext” and input the correspondent virtual extension number. To access the Virtual Extension, simply press the specific programmable button.
**Making Calls:** To place calls, simply press the specific button and dial the number.

**Answering Calls:** When the Virtual Extension receives calls, your phone will ring and the specific programmable button will be flashing. You may simply pick up the handset to answer calls.

**Retrieving Voice Messages**
You could listen to voice messages for Virtual Extensions in many ways.

- Set Virtual Ext message forward to a regular extension. When receiving new message, the regular extension’s message lamp will be lit and a running message will appear on the LCD display.
- Set Virtual Ext message UMS to an e-mail. You will be able to listen to the new messages from e-mail.
- Accessing Voice Mail System. You can also access voice mail system to listen to messages.
  - Press the Message button or press ##, then press * to go back to the main menu of VMS. Follow the prompt to enter the extension number and password to listen to voice messages.
  - Or you could access a specific AA Menu using the AA menu access code or make calls to the company phone number. Select a pre-defined DTMF key to access voice mailbox. Follow the prompt to enter the extension number to retrieve messages. (The administrator need to assign a key functioning as “Access Mailboxes”. Please refer to Chapter 7 Voice Mail Configuration/Auto Attendant for more details.)

**Analog Extensions**
The analog extensions are connected to the analog ports of the gateway like VG5000 Voice Gateway. When you install an Analog Gateway, if it provides FXS interface, you have already created the Analog Extensions. For more details about installing Analog Gateway, see Chapter 4 Analog Gateway/Creating Analog Gateway List.

1. Go to Main Menu>Extension Management>Analog Port List.
2. Select a SLT port from the list and click Modify button.
3. You will find “Gateway Name”, “Gateway Port Number” and “MAC Address” have already been entered when administrators configured the gateway.
4. Input the extension number of the SLT port in “Extension No.”.
5. Fill in information including Name, Title, Location, Department, Change Password, Class of Service, CODEC, Silence Compression, Conference, Paging, Off-Hook Access to CO Directly, Apply Setting to, Station Speed Dialing, Answer Option, Mailbox, Notification and Distribution List. For more details, please refer to Chapter 5 Extension Management/InterPhone.

Off-Premises Extensions

InterPBX is also applied to branches or the staff who work at home. Branches can reach Headquarters via Internet, VPN or Ethernet instead of leased lines as Off-Premises Extensions have the same functions as any other extension in the Headquarters.

Make sure the remote site has a permanent Internet connection using xDSL, Cable Modem or Leased Line. Ideally it is recommended that each of the Off-Premises phones can be assigned with a Real IP address but this is often not the case. If you are not able to assign a Real IP address to each of the Off-Premises phone, you can assign it a virtual IP address and open specific ports on the Off-Premises NAT device. Before you create extensions, make sure the relevant License Keys has been added.
1. Go to Main Menu>Extension Management>Off-Premises List. Click the Add button to create a new Off-Premises extension.

2. Assign an “Extension No.” to this phone.

3. Create a password for this extension.

4. Input the “MAC Address” of this extension. Every IP phone or PC running the SoftPhone have a unique MAC address to be entered into this field.

5. From the “Phone Type” drop down menu select a phone type.

6. Do not change the default Signal Port 6045 and Media Port 5000 provided by the system. If more than one Off-Premises IP phone is used in a single remote site, set the second phone’s Signal Port to 6046 and Media Port to 5001 and so on.

7. Select the “Static IP” will reinforce the system’s security and the system can only be connected by the specified IP address. If a real IP is used on off-premises phone, enter it in “IP Address”. If a virtual IP is used on off-premises phone, enter the NAT equipment’s real IP of the remote site in “IP Address”. If a fixed IP cannot be obtained (e.g. PPPoE) do not select the “Static IP” option.

8. Make sure “Enabled” check box is selected

9. Activate QoS if applicable. All other settings you may leave to be set at a later time. For other setting items, please refer to Chapter 3 Extension Management/InterPhone.

Note: If the Off-Premises extension uses virtual IP, on remote NAT equipment, you will have to open the Signal and Media Ports as well as setup their Port Mapping.
Note: If the PBX Server is also under NAT, please activate PBX Server’s NAT setting then open the NAT equipment’s UDP port 6046. Please refer to Chapter 3 System Configuration/General Parameters/Behind NAT, NAT IP Address for more detailed information. On the Off-Premises phone, input the NAT device’s Real IP address in the “PBX Server IP” item.

Note: When the off-premises extension is registered to the NAT Proxy, input the NAT Proxy’s IP address on phone’s “PBX Server IP” setting item.

Configuring Off-Premises Analog Phones
If the Off-Premises gateway you have installed supports FXS ports, you will see under “Analog Port List” the Off-Premises SLT ports. Please refer to above sections for information on how to install analog extensions. For Off-Premises gateway information, please see Chapter 4 Gateway Configuration/Off-Premises Gateways.

SIP Phone List
SIP Proxy helps you register third parties’ SIP phones to your InterPBX System and they will be one of your extensions.

1. Go to Main Menu>Extension Management >SIP Phone List.
2. Click the Add button to increase SIP phones.
3. Assign an extension number in “Extension No.”.
4. Assign a password for this extension. Extension users can change the password later. Password must be numbers.
5. From “SIP Proxy” list, select one applied for this SIP phone. For other settings, please refer to Extension Management>InterPhone.
For more details, please refer to SIP Proxy Operation Manual.

**InterConsole List**

The InterConsole is a full-featured DSS (Direct Station Selection) Console helping operators or receptionists to monitor the system-wide call status for better handling calls. The InterConsole can monitor up to 1,000 end points and calls right directly from the panel.

InterConsole is a hardware attendant console. It needs to be attached to the main IP phone. Currently InterConsole can be used with IP580 or IP590.

1. Go to Main Menu>Extension Management>InterConsole List.

2. Click Add to increase an attendant console. Assign a name for the InterConsole and select the associated extension number from the drop down menu.

3. From the InterConsole list, select one attendant console and click Button Mapping. Each page supports 100 programmable buttons. You can scroll pages and buttons to configure up to 1,000 buttons.
Choose a programmable button and select an action from the action drop-down menu. Specify the number in the “Number” box if necessary.

For example, to set Ext. 100 at button 1 as a Direct Station Selection, select “Access Ext.” and input “100”.

For more detail about the programmable buttons, please refer to Chapter 3 Extension Management/InterPhone and the InterConsole User’s Guide.
By grouping CO lines and extensions, you can better manage end points. You could also set different templates for groups to make calls.
CO Line Groups

CO Line Groups allows you to define specific CO lines as a CO Line Group. It helps allocate CO lines and simplifies Ring Assignment setting.

Creating CO Line Groups

1. Go to Main Menu> Group Management> CO Line Groups. Press the Add button to create a CO Line Group.
3. Input a name for this group.
4. Under “Hunting Method”, select one of the hunting methods for the CO Line Group. There are two options for hunting method:
   • **In Order**: Incoming calls are always directed to the first member of CO Line Group.
   • **Rotate**: If a call is directed to the 1st member in the group, the next incoming call will be distributed to the 2nd member, and so on.
5. Enter the extension number of the SLT port where the fax machine is connected. When detecting fax tones, the fax will be automatically transported to the fax machine.
6. Under “Ring Assignment”, assign a specific extension number, extension group or AA Menu access code to answer incoming calls for this group during Business Hours, Break Hours, After Hours and Closed Hours. The general setting is to assign an AA Menu to answer phone calls. Please refer to Chapter 7 Voice Mail Configuration for further information about AA Menu setting.
7. If you prefer holiday greetings to be played during holidays, select “Enable Holiday” and the system will play the holiday greetings you assigned on the Holiday List. For more detail about Holiday Settings, please refer to
Chapter 3 System Configuration/Holidays.

If you would like to access the CO Lines, simply press CO Line Group Access Code. You can create up to 50 CO Line Groups.

Assigning Members to CO Line Groups

1. Go to Main Menu>Group Management>CO Line Groups. Select a group from the list and select the Members button.
2. All available CO Lines including corresponding Gateway ports will be displayed on the Non-Members list. Select the CO Line Numbers you want to group together and assign them to “Members”. Each group can contain up to 100 members.

Each CO Trunk Port will be assigned an extension number when you configure the Gateway. For more details, please refer to Chapter 4 Gateway Configuration/Configuring CO Line Ports.

Extension Groups
Extension Groups allows you to set specific extension numbers as a group. For example, when you create an Extension Group for Sales Department, you can then edit Hunting Method and Answering Options for this group. A voice mailbox can be created and Call Pickup code can be assigned accordingly.

Creating Extension Groups

1. Go to Main Menu> Group Management>Extension Groups. Press the Add button to create an Extension Group.
2. Assign a name for this group.
3. Set a Group Number for this group.
4. Assign a Call Pick-Up Code for the group. The members can use the code to answer the calls for other members in the same group.

You can create up to 100 Extension Groups. When an Extension Group is created, voicemail box of the group will be created automatically. Users can select greetings or retrieve messages for group voice mailbox.
**Hunting Method**

Select one of the options from the drop-down box for “Hunting Method” for distributing incoming calls for this group.

- **In Order:** Incoming calls will always be distributed to the 1st extension of the group. It is possible to arrange the sequences for the extensions in Member List to answer the calls.
- **Rotate:** If a call is answered by the 1st extension, the next incoming call will be distributed to the 2nd extension in the group, and so on.
- **All:** Incoming calls will be routed to all extensions in the group at the same time.
- **Longest Idle Time:** Incoming calls will be distributed to the extension with the longest idle time in the group.

**Hunting all members before forwarding**

The default hunting action will ring the target extension for one attempt. If the target extension is busy or there is no answer, the call will be directed to group’s call forward selection. By “Hunting all members before forwarding”, the system will try to search all the extensions in the group. The incoming calls will be directed to group’s Call Forward Selection in case all the extensions are not available to answer the calls.

**Wrap Up Time**

Wrap up time allows the extension to get prepared for the next call after hanging up. With Wrap Up time, the extension will logout temporarily for processing call record and get ready for the next incoming call.

**Group Administrator Password**

Assign a password in the Administrator Password box for accessing the group mailbox or managing the group settings. Administrator Password must not exceed 8 digits in number or character and it is not case sensitive.

**Set Sec. of Rings**

If the requested extension is not answering, the incoming calls will be directed to the option under Group’s Answering Option after the pre-defined time. The number of rings is measured in seconds. Make sure No-Answer Forward under Extension Group is enabled. Please refer Chapter 6 Group Management/Extension Groups/Answering Option for further information.
**Queuing**

When you set Group busy call forward destination to Queue, you need to edit the relevant settings.

**Prompt Interval in Queue**
Set Prompt Interval allows the system to repeat the queue announcement after the period.

**Max Wait-Time in Queue and Transfer Target**
Set “Max Wait-Time in Queue” to avoid callers waiting permanently and set the Transfer destination to take calls when timeout. You may set the destination as an AA menu code, voice mail box, the other groups or others. When callers wait for their calls to be answered over the maximum Wait-Time in Queue, the call will be transferred to the destination you assigned.

**Queuing Announcement**
Select to allow the system to play queuing announcement like “You are the 2nd person in queue. Please wait.” for callers stay in queue.

**Indicate from which ext group**
If selected, when ringing, available members of the group will see the Group No. and the Group Name from IP phones. This will help group members better identify incoming calls.

**Login to Group**
Requiring members to login or logout can help the system to better distribute incoming calls. Calls will be distributed to available members only. Extensions can belong to and log in to different Extension Groups.

**Allow Members Login**
You may select the “Allow members login” check box and each extension will be required to login in order to become an effective member.

**Logout If No Answer**
In order to avoid the incoming calls being routed to the extensions that are logged into the group but is not available, you have options to allow the system
to have the extensions logged out automatically once ring-no-answer occurs. Click the “Logout if no answer” box to enable this function.

**Log in to groups**
1. Lift the handset or press the Speakerphone button.
2. Press #35.
3. Press the Extension Group Number to log in.

**Log out of groups**
1. Lift the handset or press the Speakerphone button.
2. Press #36.
3. Press the Extension Group Number to log off.

**To check the login/logoff status**
1. Lift the handset.
2. Press #37.
3. Press the Extension Group Number.
4. If you hear a dial tone, that means you are currently logged into the group.
   If you hear a busy tone, that means you are currently logged off.

**Assigning Members to Extension Groups**
1. Go to Main Men> Group Management> Extension Groups. Select a group from the list and press the Members button.
2. All the extension numbers are displayed on the Non-Members list. Select the extension numbers you want to group together and move them to the Members by pressing button and select extension members by pressing when you want to delete the extension members. Each group can contain up to 100 members.
3. It is possible to arrange extension members in order. Please select specific members and arrange them in order by pressing or button.

IP phone, analog phone and software phone will be assigned an extension number while they are being installed. Please refer to Chapter 2 Installing InterPBX communication system.
Group Answering Option

When members are not available, you may set alternatives to answer calls.

1. Go to Main Menu>Group Management>Extension Groups. Select an extension group from the list and click the Answering button to edit Group Answer Options.

2. You can select from: “Forward All To”, “Busy Forward To” or “Ring-No-Answer Forward To” to suit your needs during different business hours.

3. Selecting Answering Options:
   - **Extension:** When selected, calls will be forwarded to the other extension or an external phone number. Assign the target extension number or phone number.
Note: When setting an external phone number, remember to input your CO line access code, such as “0”.

- **Voice Mail**: When selected, incoming calls will be routed to the VM server’s Transfer Options. “VM” will be displayed in the box automatically. Note that under Transfer Options, you have to assign a DTMF key for callers to leave a message.

- **Auto Attendant**: When selected, calls will be forwarded to Transfer Options not supporting voice-mail function. The box will show “AA” automatically. Note that even though you assign a DTMF key for leaving a message under Transfer Options, callers will not be able to leave messages.

- **Queuing**: It is only available for Busy Forward. When selected, callers can stay in queue till any one of the Group is available. The system can hold up to 50 callers in the queue. You can go to Main Menu>Group Management>Extension Groups to enable queue announcement, and set interval of playing announcement, queue timeout and the action when timeout.

- **Chain Queue**: It is only available for Busy Forward. Assign the Alternative Extension Group Number. When members of the Major Extension Group are all busy, calls will be redirected to the Alternative Extension Group. If members of the Alternative Extension Group are all busy as well, calls will be bounced back to the Major Extension Group’s Queue till timeout. You can go to Main Menu>Group Management>Extension Groups to enable queue announcement, and set interval of playing announcement, queue timeout and the action when timeout.

Note: If the Answering Option is not enabled, calls will be redirected to the VM Server’s Transfer Options not supporting voice-mail. For more details about Transfer Options, please see Chapter 7 Voice Mail Configuration/Transfer Options.

**Group Mailbox**

Under Mailbox, you can select the language of the voice mail box, message play priority, and the target for message forwarding.
1. Go to Main Menu>Group Management>Extension Groups. Select an extension group from the list and click Mailbox button for editing the voice-mailboxes for this Extension Group.

2. Select “Language” from the list, for your voice mail system language. The system provides a maximum of four different languages, and will play the greetings in pre-defined language. Please consult your distributor for multi-language support.

3. Message Play Priority provides two options to retrieve messages, “Last-In-First-Out” or “First-In-First-Out.”

4. Input the target voice-mail box number which new messages will be forwarded to.

5. It is optional for users to enable “Play Time Stamp” for announcing the time messages are recorded.

To retrieve messages from voice-mailboxes of Extension Group

You could listen to voice messages for Extension Groups in many ways.

- Set Group message forward to an extension. When receiving new message, the extension’s message lamp will be lit and a running message will appear on the LCD display.
- Set Group message UMS to an e-mail. You will be able to listen to the new messages from e-mail.
- Accessing Voice Mail System. You can also access voice mail system to listen to messages.
- Press the Message button or press ##, then press * to go back to the main menu of VMS. Follow the prompt to enter the Group number and Administrator password to listen to voice messages.
- Or you could access a specific AA Menu using the AA menu access code or make calls to the company phone number. Select a pre-defined DTMF key to
access voice mailbox. Follow the prompt to enter the Group number and the Administrator password to retrieve messages. (The administrator need to assign a key functioning as “Access Mailboxes”. Please refer to Chapter 7 Voice Mail Configuration/Auto Attendant for more details.)

Please refer to Appendix B Personal Voice Mailbox for more details.

**Message Notification for Extension Groups**

The system provides several options for administrators to set-up message notification. Go to Main Menu>Group Management>Extension Groups. Select one of the extension groups from the list and click Notification button to edit message notification.

![Notification of Extension 7986](image)

**Internal Notification**

Internal Notification is Ring Notification, which allows the system to dial to an extension to notify users of new messages. The steps for setting up Notification are:

1. Disable or enable internal notification according to your preference.
2. Input the extension number for message notification when the Internal Notification is enabled. It allows the system to ring the pre-defined extension and notify the Extension Group for new messages.

**External Notification**

1. Select “Enable External Notification” or “Urgent Messages Only” to notify the users of new messages marked as urgent. When “Urgent Message
Chapter 6 Group Management

Only" is enabled, users will be notified of urgent messages even if “External Notification” is disabled.

2. Notification Schedule provides flexibility to notify users of new messages during business or personal hours.

3. Set External Notification from the first sequence. Scroll down the “Type” list and select “Phone” or “Pager” for notification or select “Disabled” if not needed.

4. When enabled, input “Phone/ Pager Number” for notification.

5. Set interval period. The system will notify users of new messages after the pre-defined Notification Interval.

6. The system will notify users of new messages according to “Try” cycles. After the system reaches the number of the “Try” cycles or the owners of voice mailbox has saved the message as an old message or deleted them after retrieving the messages, the system will stop notifying users.

The system will deliver notifications of new messages from sequence 1 to sequence 5, if the target set in sequence 1 is busy or not available, the system will keep notifying after the interval till reaching the “Try” cycles. If the system can’t reach the target of sequence 1, it will notify users from sequence 2. Note that if the target phone or extension is set Busy or No-Answer Forward, the notification will be forwarded to the target’s call forward accordingly.

Unified Messaging

UMS function allows the system to forward messages to the e-mail account of the administrators or other extensions. New messages will be forwarded to e-mail as a WAV attachment. The steps for activating UMS are:

1. Select “Enable UMS”

2. Select “Keep as new” or “Save as old” message after e-mail notification.

3. Input the e-mail account of the extension number. You can edit up to 3 e-mail addresses.

When UMS is enabled and “Save as old” is selected, the users won’t be notified of new messages via Internal and External Notification since the messages will be kept as an old after e-mail notification.

Note: When the UMS function is enabled, a SMTP server needs to be set. Please refer to Chapter 3 System Configuration/ System Parameters/ SMTP Server.
**Button Mapping Groups**

Button Mapping allows you to set programmable buttons on IP phones to represent specific features like speed dialing, accessing CO lines, call-park and other frequently used functions. You can set some or all programmable buttons as Button Mapping Groups allowing extensions or extension groups to adopt it. When applying, notice the number of available programmable buttons vary on different model of IP phones.

1. Go to Main Menu> Group Management> Button Mapping Groups. Press “Add” to create a group. You can create up to 50 Button Mapping Groups.
2. Assign a name for this group.
3. You can edit up to 15 programmable buttons and lock them in the group level. Extension users can edit unlocked programmable buttons only in Personal Button Mapping. Depending on your preference, refer to the table below to assign an action and input its corresponding number.
4. Select one of the programmable keys and choose one of the actions for it from “Action”.
5. Either input the targeted number according to the setting in “Number” which is assigned in step 3 above or not key in any information in “Number.” Input the requested number manually while pressing the programmable keys. For example, you set “Personal Speed Dial” under “Action”. When you input someone’s Personal Speed Dial in “Number”, you are capable of dialing the numbers which are pre-defined in “Number.” If you prefer not to enter any digits in “Number”, input the requested number manually after pressing programmable keys. Please refer to the Action list below for more information:

<table>
<thead>
<tr>
<th><strong>Action</strong></th>
<th><strong>Designated Number</strong></th>
<th><strong>Description</strong></th>
</tr>
</thead>
</table>

![Button Mapping Groups](image)
<table>
<thead>
<tr>
<th>Action</th>
<th>Designated Number</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Access Ext.</td>
<td>An extension number of a station, extension group, CO line, CO line group, or an outside phone number</td>
<td>To make calls to the assigned number. Please note to add CO line access code, e.g. 05551234 for an outbound call. Notice that if Trunk Group is set on a button, when parts of trunks are available, the LED displays available status for other people.</td>
</tr>
<tr>
<td>After Call Work</td>
<td>-</td>
<td>To allow agents to have a period of after call work time between two calls.</td>
</tr>
<tr>
<td>Ask Member Login</td>
<td>An Extension Group Number</td>
<td>Check the login status of the extension group. Hearing a dial tone means members are logged-in successfully and a busy tone means not logged-in.</td>
</tr>
<tr>
<td>Auto Line Access</td>
<td>-</td>
<td>To get a CO Line</td>
</tr>
<tr>
<td>Auto-In</td>
<td>-</td>
<td>Agents’ extensions will pick up the next call automatically.</td>
</tr>
<tr>
<td>Auxiliary Time</td>
<td>-</td>
<td>Agents can enable this function to notify the system stop assigning incoming calls to their extensions so that the agents can leave their seats temporarily without logging out the system.</td>
</tr>
</tbody>
</table>


<table>
<thead>
<tr>
<th>Action</th>
<th>Designated Number</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Appearance</td>
<td></td>
<td>Press to make or receive calls. It functions similar to intercom. When more than one button is assigned, buttons with lowest number will take the first call.</td>
</tr>
<tr>
<td>Call Hold Retrieve CO</td>
<td>An extension number of a CO line</td>
<td>To retrieve an incoming call from a specific CO line placed on hold. (You should be able to see the CO line extension number. For using this function.) If this CO line is set on the programmable button, users can directly press the flashing button to retrieve the incoming calls.</td>
</tr>
<tr>
<td>Call Hold Retrieve Ext</td>
<td>An extension number of a station</td>
<td>To retrieve an internal call placed on hold. If this extension number is set on the programmable button, users can directly press the flashing button to retrieve the call.</td>
</tr>
<tr>
<td>Call Park</td>
<td>A slot number (0-9)</td>
<td>To park a call to a specific slot.</td>
</tr>
<tr>
<td>Call Pickup CO Line</td>
<td></td>
<td>To answer the least recent incoming call ringing on the system.</td>
</tr>
<tr>
<td>Call Pickup Directed</td>
<td>An extension number</td>
<td>To answer a call ringing at another extension.</td>
</tr>
<tr>
<td>Conference Call</td>
<td></td>
<td>Start Conference with the callers placed on Hold.</td>
</tr>
<tr>
<td>Action</td>
<td>Designated Number</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------</td>
<td>------------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Forward All Calls</td>
<td>An extension number + #</td>
<td>To forward all the incoming calls to a specific extension or an external phone number automatically. Press again to disable. When setting external number, please add the CO line access code such as 0.</td>
</tr>
<tr>
<td>Headset</td>
<td>-</td>
<td>Press to allow voice been transmitted from the attached headset, instead of the handset.</td>
</tr>
<tr>
<td>Internal Paging</td>
<td>An extension number or extension group number</td>
<td>Broadcast for members through the extension or the extension group.</td>
</tr>
<tr>
<td>Manual-In</td>
<td>-</td>
<td>To allow agents to pick up the next incoming call manually by pressing the specific button of Manual-In function.</td>
</tr>
<tr>
<td>Member Login</td>
<td>An Extension Group Number</td>
<td>Login to be one of the members in the Extension Group.</td>
</tr>
<tr>
<td>Member Logoff</td>
<td>An Extension Group Number</td>
<td>Logout from the Extension Group.</td>
</tr>
<tr>
<td>Personal Speed Dial</td>
<td>A personal speed dial number (e.g. 00)</td>
<td>To dial a number defined on the Personal Speed Dial Number.</td>
</tr>
<tr>
<td>Record on Demand</td>
<td>-</td>
<td>Press to save the recording or start recording. When recording, the LED will be flashing.</td>
</tr>
<tr>
<td>Retrieve Msg</td>
<td>-</td>
<td>To access mailbox.</td>
</tr>
<tr>
<td>Retrieve Record</td>
<td>-</td>
<td>To play the saved recordings.</td>
</tr>
<tr>
<td>System Speed Dial</td>
<td>A system speed dial number (e.g. 000)</td>
<td>To dial a number defined on the speed dial number.</td>
</tr>
<tr>
<td>Action</td>
<td>Designated Number</td>
<td>Description</td>
</tr>
<tr>
<td>------------------------</td>
<td>-------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Transfer to Ext VM</td>
<td>An extension number</td>
<td>To transfer a call to the extension’s voice mailbox.</td>
</tr>
<tr>
<td>Transfer to AA Tree</td>
<td>An AA menu access code</td>
<td>To transfer a call to the AA menu.</td>
</tr>
<tr>
<td>Virtual Extension</td>
<td>A virtual extension no.</td>
<td>Press to act as the assigned virtual extension to make calls.</td>
</tr>
<tr>
<td>Night Service</td>
<td>-</td>
<td>Press to switch Operation Mode to Closed Hours. The AA and operator will be changed accordingly. Press again to release.</td>
</tr>
</tbody>
</table>

**Class of Service**

Class of Service (CoS) allows extension users call permissions of intercom calls, accessing CO lines or activating External Notification. Each extension requires a Class of Service. The default CoS will be granted to IP extensions in the process of Auto Discovery. You can create several types CoS providing different calling permissions for extensions or extension groups. Notice that the calling permission settings on CoS are superior to personal extensions.

**Creating a Class of Service**

1. Go to Main Menu> Group Management> Class of Service. Press Add button to create a type of CoS. You can create up to 50 CoS.
2. Assign a name for the CoS.
3. Set calling permissions for member of this CoS in Business, Break, After, Closed hours and Holidays. The system will refer to your Call Restriction to identify whether calls that are being made are local, long distance or international calls. For more details, please refer to Chapter 3 System Configuration/Call Restriction.
   - **Intercom**: Allow extensions to make phone calls to other extensions.
   - **Local**: Allow extensions to make local calls.
   - **Long Distance**: Allow extensions to make long distance calls.
   - **International**: Allow extensions to make International calls.
   - **Joint Server**: Allow extensions to make phone calls to the extensions on other InterPBX systems.

4. Continue to set calling permissions as below.
   - **Off-Site Notification** (or External Notification): Allows the system to make outbound calls to extension users phones or pagers for notification of new voice messages. The External Notification permission is not limited by users’ long distance or internal call permissions.
   - **Call Waiting**: Allow one incoming call to stay online when the requested extension is busy. The caller will hear the regular ring tone.
   - **Do not disturb**: Allows extensions to activate DND from phones by pressing the DND button. The DND status is similar to ring-no-answer.
   - **Record**: Allows extensions to activate the recording function. Notice that the integrated Record-on-Demand and/or Store-on-Demand supports up to 10 concurrent channels.
   - **Mailbox**: Allows extensions to activate the Mailbox function. Note: To activate the Mailbox function successfully, you also have to enable the VM (Voice Mailbox) function in the setup menu “Answer Option.”
   - **Auto Answer**: Allows extensions to activate the Auto Answer function from phones by pressing the Auto Answer button.
   - **Trunk Disconnect Timer**: Allow trunks to get disconnected automatically after the period. This will set the maximum call duration for callers. The trunk types apply to analog, digital and SIP trunks. Warning beep tones will be provided to both callers and receivers 15 seconds prior to the disconnect time.

**Trunk to Trunk CoS**: In Class of Service, there is a default Trunk to Trunk CoS. If you would like to provide trunk to trunk transfer option, you need to edit that entry. Depending on your needs, enable the long distance, international or
Joint Server call permission. You may go to Main Menu > System Configuration > Miscellaneous to set the Max. trunk to trunk duration to better protect the utilization of your CO lines.

**Note:** The administrator has to assign at least one type of Class of Service.

### Setting Allow and Disallow Table

You can create call exceptions or restrictions on Allow and Disallow Table for each CoS. By setting particular prefixes for international call, long distance call or other toll calls as exceptions, this way you can have better control of your resources for trunk lines. The exceptions will override the limitation by CoS.

**Setting Allow Table**

Go to Main Menu > Group Management > Class of Service. Select a CoS from the list and press Allow Table button.

Under “New Prefix”, enter the access codes for specific long distance or international calls and then click the Add button. Repeat the steps if more access codes are needed. You can create up to 20 entries on “Allow Prefixes”.

For example, in the USA, you can set the toll free number prefix “1-800” in the “Allow Table” for users who are not allowed to make long distance calls.

![Allow Table](image)

**Setting Disallow Table**

Go to Main Menu > Group Management > Class of Service. Select a CoS from the list and press the Disallow Table button.

Under “New Prefix”, enter access codes for not allowing users to make long distance or international calls and then click Add button. Repeat the steps
above if more access codes are needed. You can create up to 20 entries on “Disallow Table”.

For example, in the USA, you can set the pay-per-call number prefix to “1-900” in the “Disallow Table” for not allowing users to make long distance calls.

**Setting CO Priority**

You can assign the priority of CO Lines or CO Line Groups for extensions of the CoS to get when placing outgoing calls. That means when users dial the CO line access code, such as 0, they will get an outgoing line based on the priority.

1. Go to Main Menu > Group Management > Class of Service. Select a type of Class of Service and press the CO Priority button.
2. You can edit CO Line from 0 to 9 in sequence. Select one number from the list and enter the preferable CO Line or CO Line Group extension number and then press the Add button. Repeat this procedure to assign more numbers to the list.
If you have different CO lines for different departments, it is possible to set different types of Class of Service and assign the pre-defined CO Line Priority to CoS. The department will get different CO Line numbers when pressing the CO line access code, such as “0”.

Note: When making call, the system will check the setting starting from AAR, ARS and CO priority.

**Automatic Route Selection (ARS)**

When you have PBX Server/Voice Gateway in different places or when you have SIP Trunks or other cost-saving systems, you can set all the available routes ARS for different Class of Services. Users will be able to access different routes at different time for better utilizing enterprise resource.

1. Go to Main Menu > Group Management >Class of Service. Select a “Class of Service” from the CoS list and then click “ARS”.
2. Enter the number you could like to modify, such as the international code + country code of a specific route in the field “Dialed String”, e.g. “01144” for calls from US to UK. You can enter up to 28 digits of numbers here but the dialed string should be unique.
3. Enter the length limit of the total dialed numbers in the field “Min Digits” and “Max Digits”.
4. Select a route from the “Route” drop-down menu. For more information, please refer to Chapter 3 System Configuration/Route.
5. Enter the length of digits you want to delete in the field “Delete”. For example, if you want to make a long-distance call from New York to Los
Angeles (e.g. 1 213 XXXXXX) through an off-premises gateway in LA, you can enter “4” here. The system will automatically delete the first 4 digits (area code: 1 213). The gateway in LA will only dial the numbers followed by the area code and turn this call to be a local call.

6. Enter the international code or other numbers you want to add in the field “Insert”. For example, if you want to allow all international calls from US to be taken through AT&T service, you may set “1010288” in the “Insert” field.

7. After all the settings are completed, please click “Add” to increase this ARS in the ARS list. To remove the ARS, please select an ARS from the list and then click “Remove”.

8. You may also click “Copy from” to directly copy the ARS list of another CoS.

If you don’t have additional routes for selection, you don’t need to set ARS. Users will access CO lines according to the setting at CO Priority.

**Note:** Before you set ARS, you have to specify the Route List. For more information, please refer to Chapter 3 System Configuration/Route.

**Note:** When making call, the system will check the setting starting from AAR, ARS and CO priority.

**Automatic Alternate Route (AAR)**

You may also edit the AAR (Automatic Alternate Route) list to add or delete string automatically.
1. Go to Main Menu > Group Management > Class of Service. Select a “Class of Service” from the CoS list and then click “AAR”.
2. Enter the digits you could like to modify. You can enter up to 28 digits of numbers here but the dialed string should be unique.
3. Enter the length limit of the total dialed numbers in the field “Min Digits” and “Max Digits”.
4. Enter the length of digits you want to delete in the field “Delete”.
5. Enter the numbers you want to insert.
6. When completed, click the Add button. To remove, please select an entry from the list and then click the Remove button.
7. You may also click “Copy from” to directly copy the AAR list of another CoS.

For Example: If your PBX is connected to the other PBX System with a Tie Line, you may set a rule to replace the original dialing method “09+ext no.” When the extension numbers of the counter part PBX are all “3xxx”, you could set Dialed String=3, Min Digit=4, Max Digit=4, to set the length as 4, and Insert=09. When dialing “3000”, the system will detect “3” and 4 digits you input. The system will then apply the rule and dial “093000”. This will allow you to dial the extension number only.

Note: When making calls, the system will follow the settings starting from AAR, ARS and CO priority.

Operator
You can assign one of the extensions or extension groups as “Operator” during different business hours. Under AA Menu and Transfer Option, the calls will be routed to the pre-defined extensions or extension groups when the calls are transferred to Operator.
1. Go to Main Menu > Group Management > Operator.
3. Enter Operator extension or Extension Group code in Business Hour, Break Hour, After Hour, Closed Hour and Holiday.

About the settings of Business Hours and Holidays, please refer to Chapter 3 System Configuration. If you assign a specific extension group as the operator, you need to set the Hunting Method in order to answer the incoming calls much more efficiently. For more information about the settings of Hunting Method, please refer to Chapter 6 Group Management/Extension Groups/Hunting Method.

Authorization Code
You can assign authorization codes for different users. Users will be able to use the full functions such as placing long distance or international calls as specified on their own CoS from extensions with limited call permissions and CoS.

2. Select one entry from the list and assign the correspondent Authorization Code, Class of Service to be applied and comment. The authorization code length should conform to your setting at Main Menu>System Configuration>Miscellaneous.

Note: The Authorization Code function needs to be implemented with ARS.

Making Calls using Authorization Code
With provided Authorization Code, users can make calls from an extension with limited call permission.

1. Lift the handset or press Speakerphone button.
2. Wait for the dial tone and dial the CO line access code such as “0”.
3. Dial the phone number. You will hear beep tones.
4. Input your Authorization Code. Your call will be connected.

You could use the cancel symbol as assigned at Main Menu > System Configuration > Miscellaneous to cancel and try again without the need to hang up and dial the number from the beginning.

**Boss and Secretary**

Boss and Secretary Group function allows one secretary to handle calls for up to 5 bosses. Calls for bosses will be assigned to bosses and the secretary at the same time. Notice a secretary needs to set associated bosses extensions to programmable buttons after been included to the group.

1. Go to Main Menu > Group Management > Boss-Secretary.
2. Click the Add button to increase one entry.
3. Assign a name for this group.
4. Set the secretary extension number.
5. Set the boss extension number. You could set up to 5 bosses numbers.
6. Set Boss Ring Assignment.
   - **Ring**: When there is an incoming call for the boss, the extension of boss and secretary will ring at the same time.
   - **No Ring**: When receiving calls, boss’s phone will not ring but the intercom button and the LCD display will be flashing and show the incoming call message.
• **Delay Ring:** You could also set the boss phone to ring a little bit later than the secretary. So that the secretary’s phone could ring first and pick up the call. When selected, set the delay ringing time.

7. Go to Main Menu > Extension Management. Select the secretary extension and edit the Button Mapping. You need to set one programmable button as Boss’s extension number allowing the Boss-Secretary function well. Select one programmable button, set the function as “Access Ext” and input boss extension at the “Number” field.

**For the Secretary:** When one of the bosses has incoming calls, the secretary extension will be ringing and the assigned programmable button will be flashing. Simply pick up the phone to answer calls.

**For the Boss:** When the boss receives an incoming call, depending the Ring Assignment setting at Boss-Secretary Groups, your phone will be ringing as well, no ring but you could still see the flashing intercom button and LCD panel or delay ringing. Simply pick up the phone to answer calls. Or leave it to your secretary to handle it.
Chapter 7
Voice Mail Configuration

InterPBX Communication System offers full-featured auto-attendant, voice-mail and unified messaging features within the Voice Mail Server. The Voice Mail Server is integrated with the PBX Server or can be deployed as a stand-alone server.
Voice Mail Parameters

When adding a new extension, its associated mailbox is created automatically. In mailbox, you can edit the parameters and the limit the storage capacity for each voice mailbox to save system storage space. Go to Main Menu> VM Configuration>Voice Mail Parameters.

Language

Select a language from the “Default Language” list. The system prompts will be announced according to the language you select. The system can support up to 4 languages. Please consult your distributor for system language options.

Name Directory

Name Directory allows callers to search extension numbers by entering the name of the person to be reached. Callers can enter either the last name, first name or either one.

After entering the name, the system will announce the associated extension number. Callers can then input the extension number. If the extension users have recorded their names in personal mailbox settings first, personal greeting and extension number will be played when the call is directed to the voicemail box of the desired extension. Name Directory only supports English first name or last name.

Inter-Digit and AA Timeout
Inter-digit timeout is the interval between two digits entered. If callers don’t enter the next digit within the pre-defined time, the system will assume the input action ends.

AA timeout is the duration that callers have to enter the digits after the AA greeting plays. If there are no digits entered within the pre-defined time, the system will lead the caller to AA Menu Timeout Action which is assigned to an Operator, for example. The timeout unit is in seconds.

**Messages Storage Settings**

It is recommended to set storage limits for each personal mailbox to ensure enough system memory space.

- **Max No. of Msgs:** Set the maximum number of old and new messages in each mailbox. When stored messages exceed the limit, callers won’t be led to voice mailbox to leave messages.
- **Max Msg Length:** When the message recording length exceeds the limit, the caller will be prompted to review the message, re-record or save the message. The unit is in seconds.
- **Min Msg Length:** If callers hang-up without hitting any keys after the call ends, any left messages which reach the pre-defined minimum recording length will be saved. However, it does not apply to callers who end the call and leaving messages by selecting save option. The unit is in seconds.
- **Purge New Msg After:** You may allow the system to periodically delete the new messages. The new messages will be purged automatically after the pre-defined number of days. To disable this function, input “0” for this parameter. Notice that the purged messages will be permanently deleted and cannot be restored. The unit is in days.
- **Purge Old Msg After:** This parameter defines how many days old messages should be kept. To disable this function, set “0” for this parameter.

**Ring Notification**

The ring notification allows the system to call and notify extension users when they have new messages. It also assists users who use analog phones or have virtual mailbox to be notified for new messages when there is no dedicated message lamp on the phone.

- **Ring Notification Interval:** It is the pause between attempts to call the extension for notification. When callers leave messages in voice mailbox, the
system will call the pre-defined extension for new messages after Ring Notification Interval.

- **Ring Notification Try**: The system tries to notify users of new messages according to the times of “Try”. The system will stop ring notification when users choose to save or delete the new messages after retrieving them.

- **Ring-No-Answer Timeout**: Ring-No-Answer Timeout controls how long the phone rings when the system notifies the mailbox users.

- **Notification Channel**: The system is able to assign some of the Voice Mail channels to notify users for new messages. The available channels are from 0 to 23 ports.

**Adjust Gain Level**

You may also amplify the volume of prompts or greetings on Voice Mail System if necessary.

**Transfer Options Settings**

Transfer Options allows callers to leave a message directly in the voice mailbox, transfer to extensions or Operator when the requested extension is busy or not available. Go to Main Menu> VM Configuration>Transfer Options.

**Transfer Announcement**

It is optional to allow the system to play transfer prompts, such as “Please hold, while I transfer you to extension 100”, or to disable transfer announcement when the call is being transferred to the requested extension.
• **Announce Call Transfer:** If selected, the system will announce the call transfer prompt, “Please hold, while I transfer you to …” When it is disabled, the call will be transferred without the announcement.

• **Announce Name:** If enabled, the system will announce the extension number or the name recorded in the personal mailbox. If the extension users don’t record names, the system will announce the extension number while the call is transferred.

**Leave Message Directly**

When “Leave Message Directly” is enabled, callers will be directed to the voice mailbox of the requested extension and will be asked to leave messages directly when the extension is not answering or busy. The “Voice Mail” function needs to be enabled on the extension. If it is not enabled, all the calls will be transferred to AA and callers can not leave messages.

**Note:** When “Leave Message Directly” is enabled, the incoming calls can’t be transferred to any targets and DTMF keys 0~9, * and # signs will be also disabled under Transfer Option.

**Transfer Options**

Besides “Leave Message Directly” under Transfer Options, the system also provides callers with more options, including: Transfer to Extension, Operator or Hold for Busy, etc. The caller can press pre-defined DTMF keys (0~9, *, #) for transfer. Transfer Options will be performed when “Leave Message Directly” is disabled. Select DTMF keys (0~9, *, #) to correspond with one of Actions under Transfer Options.

• **No Action:** The system will play an error announcement and then repeat the AA Menu greeting.

• **AA menu:** For AA Menu, the call will be transferred to the target AA Menu. Set the target AA Menu ID in the “Target” box. It offers administrators the flexibility to organize all the AA Menu flow or DTMF programming.

• **Lead to Extension Number:** It is helpful when the first digit of the desired extension has been used or the extension number has a wide range. Callers need to press one pre-defined DTMF keys for being led to desired Extension Number. For example, if the caller wishes to reach extension “100” and “Lead to Extension Number” is set on “3”, the caller should press “3” and then “100” for reaching extension 100.
• **Transfer to Extension**: Select one of DTMF digits as “Transfer to Extension” according to the first digit of extension numbers. For example, set “1” as “Transfer to Extension” when the first digit of the extension numbers start with “1”. If the caller wishes to reach extension “100”, he only presses “100” and the call will be directed to extension 100.

• **Leave a Message**: The caller will be led to the mailbox of the extension and asked to leave a message when the desired extension is not available. The “Voice Mail” function needs to be enabled on the extension. If it is not enabled, all the calls will be transferred to AA and callers can not leave messages.

• **Operator**: The incoming calls will be transferred to the Operator.

• **Repeat Menu**: The system will announce the prompt corresponding to its AA Menu.

• **Hang Up with Announcement**: The call will be disconnected after an announcement is made.

• **Hold for Busy**: It is applicable only when the requested extension is busy. The system will put the caller on hold and attempt to transfer the caller to the extension again.

• **Forward External**: It is applicable when the extension user sets an external phone number in personal mailbox settings. The caller will be connected to the predefined forward number.

• **Hang Up**: The call will be disconnected without announcement.

**Using Forward External from Transfer Options**

In addition to set External Call Forward from extension, you can enable this function from Transfer Options.

1. The administrator needs to define a DTMF key as External Forward function. Go to Main Menu>VM Configuration >Transfer Options. Set a DTMF button as Forward External.

2. Users need to define an external phone number as forward destination. Go to Main Menu>Extension Management >Interphone> Mailbox. Set the target phone number at “Extension Forward to” field.

3. When the extension’s Answering Option set to AA or VM, calls will be redirected to Transfer Options. Callers will hear the prompt guiding them to press a specific DTMF key to be redirected to the preset phone number.

**Set Up Timeout Action**
Select one of “Timeout Action”. If the system gets no response from the caller after transfer announcement is played, the system will perform according to the Action set in “Timeout”, just like “Access Mailbox”, “Operator” or “Hang up”.

**Before Leaving Message Timeout**

The system waits for the caller to input any digits after it plays the transfer announcement. After the duration, the system will transfer the call according to Action set in “Timeout”. The unit is in seconds.

**AA Management**

With the Auto Attendant function, the system is capable of answering incoming calls, transferring the calls to the requested extensions, extension groups or any target. The system also supports multi-level AA Menus which provides options for transfer and corresponding prompts to lead callers to the target. Administrators not only organize AA Menu as preference, but also assign independent AA Trees to different CO Lines or CO Line Groups.

**Planning Your Auto Attendant Tree**

An organized Auto Attendant Tree will lead the callers to the target with efficiency, improve productivity and raise satisfaction. Set-up Auto Attendant Tree by following the steps below:

1. Arrange AA flow according to needs and select Action under AA Menu.
   Please refer to the chart as follows.
2. Record greetings in AA Menu that is activated for leading callers to the target.
3. Assign an AA Menu Access Code to “Ring Assignment” under CO Line or CO Line Group.

For more detail about Ring Assignment, please refer to Chapter 4 Gateway Configuration/Setting CO Line Ports.
Planning an AA Menu

Multi-level Auto Attendant Tree is constructed with independent scripts called AA-Menus. Each AA-Menu has its own greeting and customized action key. From the menu, the caller can be guided to the extensions, Extension Group, Operators and information bulletins, etc. There are 300 AA Menus available. Each has an ID (from 0 to 299) as you can see from the Auto Attendant Tree list.
1. Go to Main Menu > VM Configuration > AA Management.
2. Select an entry from the list and click on the Modify button.
3. In “Description” shown as blank, describe the purpose for this AA Menu, for example, “Business Hours”.
4. Assign an “Access Code” which is needed while Ring Assignment is being configured for this AA Menu. For more details about Ring Assignment, please refer to Chapter 4 Gateway Configuration.
5. You may record the greeting for this AA Menu later.
6. Assign each DTMF tone (0-9, *, #) with one specific AA Menu Action from the list. The system will perform according to AA Menu Action when callers press one of pre-defined DTMF keys. Available actions are as below:
   - **No Action**: The system will play an error announcement then repeat the AA Menu greeting.
   - **AA Menu**: The call will be transferred to the target AA Menu. Set the target AA Menu ID in the “Target” box.
   - **AA Menu in 1st/2nd/3rd/4th Language**: The call will be transferred to another AA-Menu in the specified language. Input the target AA Menu ID in “Target” box. The following system prompts will also be switched to the specified language.
   - **Lead to Extension Number**: It is helpful when the first digit of the desired extension has been used or the extension number has a wide range. Callers need to press one pre-defined DTMF keys for being led to the desired Extension Number. For example, if the caller wishes to
reach extension "100" and “Lead to Extension Number” is set on "3", the caller should press "3" and then “100" for reaching extension 100.

- **Transfer to Extension:** The assigned DTMF digit is the first digit of an extension to be transferred. For example, if the action is set on "1", callers will reach extension 100 when press “100”.

- **Direct Transfer to Extension:** The call will be transferred to the target extension directly. Set the target extension number accordingly. For example, if the action is set on "6" with the target set as "100", callers can press "1" to reach extension 100 directly.

- **Transfer to Mailbox:** The assigned DTMF digit is the first digit of the mailbox to be transferred. For example, if the action is set on "2", callers can dial "200" to leave a message directly at mailbox 200.

- **Direct Transfer to Mailbox:** The caller can leave a message directly at the target mailbox. Set the target mailbox accordingly. For example, if the action is set on "7" with the target set to "200", callers will be taken directly to mailbox 200 by pressing "7".

- **Access Mailboxes:** Callers will be asked to enter voice mailbox number and password to accessing mailbox. It is helpful for business travelers to retrieve messages.

- **Name Directory:** Depending on your setting at the above Voice Mail Parameter, the caller will be prompted to enter the first name, last name or either one of the person to be reached. The system will then announce the associated extension number. It is only applicable to enter English names when the Name Directory is configured.

- **Operator:** The incoming calls will be transferred to the Operator.

- **System Programming Mode:** It allows administrators to enter DTMF Programming Mode to setup the system or to record greetings.

- **Repeat Menu:** The system will announce the prompt corresponding to its AA Menu.

- **Hang Up:** The call will be disconnected without announcement.

- **Hang Up with Announcement:** The call will be disconnected after an announcement is made.

**Recording AA Menu Greetings**

Administrators can arrange to record greetings in each AA Menu to inform callers about service options and leads them to the targets. The system will play default greeting if there is no any greeting recorded in AA Menu.
1. Go to Main Menu> VM Configuration>AA Management.
2. Select an entry from the list and click on the Modify button.
3. If greetings have been recorded in the AA Menu, “Y” will be displayed by
   the Greeting section. If “N” is displayed, it means there is no greeting for
   this AA Menu. Press the Record button to record greetings.
4. Choose an extension, preferably one that is close to you, start recording
   when “Record of a Prompt” dialogue pops out. Input the selected
   extension number in the “Action Ext.” box and click Record button to start
   recording or click Play button to review recordings.
5. The system will ring the defined extension immediately. Pick-up the phone
   and you will hear the prompt announcing the AA Menu ID and guiding
   you to review, edit, save or record prompts. Start recording followed by
   voice guidance and save it after recording is complete.

Administrators may also dial AA Menu access code from any extension to
review AA Menu greetings.

Record Greetings via remote control
It is optional for administrators to record greetings via DTMF Remote Control.
The steps are:
1. Dial into one of the CO Line port numbers on Voice Gateway with a phone,
   for example, 5551234.
2. Enter a pre-defined key for DTMF remote control, ex: “#” sign. Please
   remember to assign the key for DTMF remote control in AA Menu. Please
   refer to Chapter 7 Voice Mail Configuration/ Auto Attendant for more
   details about AA Menu.
3. The administrators will be asked to enter a greeting recording password
   (Default Password: 5678).
4. The administrators will be asked to enter a three digit Function Code in order to edit. Greeting Recording Password only allows administrators to record greetings in Function Code 330.

5. Enter three digits AA Menu ID (000~299) to edit, save or review recordings followed by voice guidance.

**CO Priority of Notification**

You can allocate certain CO lines or groups and set their priority for delivering external notifications. You can set the least used CO lines or groups so as not to interfere with the regular incoming or outgoing calls.

1. Go to Main Menu> VM Configuration> CO Priority.
2. Select Entry 0 from the list, input a CO line or CO line group extension number, and click the Add button.
3. If you would like to add more CO lines, select Entry 1 and repeat the above procedures. You can set up to 10 entries.
4. The voice mail system will follow the CO line priority starting from Entry 0 to deliver notifications.

**Tel. Programming Mode**

It is efficient to program the system via phones when administrators are not available to configure VMS Server via Internet or go to the site. Administrator can dial into Voice Gateway and program parameters or record greetings via DTMF remote control. Please refer to Appendix F for more details about DTMF remote control.
Chapter 8
Operation Management

Operation Management consists of system reports. Including backup, restore and upgrade options for handling system files and configurations to ensure optimum InterPBX performance. Furthermore, administrators can download call logs for review.
Chapter 8 Operation Management

**General Information**


![General Information Screen](image)

**Check Software Version**

- PBX Server: As above mentioned, Administrators can check PBX Server system version on General Information.
- Extensions: Under Devices image version, click “Detail” to check all types of IP phone and SoftPhone versions.

![Check Software Version](image)

- Voice Gateways: Go to Main Menu> Gateway Configuration> Analog Gateways. Highlight one of the gateways from the list and select “Modify” which will display Voice Gateway version. Here you can check what version your system is.
License Key

If you would like to expand the systems capacity, add voice gateways, IP phones, professional recording systems or CTI Gateways, you will need to add License Keys to upgrade the system.

1. Go to Main Menu> Operation Management> General Information.
2. Select “Add” to add a License Key.
3. Add License Keys in “Add a License Key” and press “Submit” to save it.

The newly added License Key will be displayed under License Keys list.

Select “Remove” to remove License Keys or “Detail” to check the contents of License Key in the License Keys list.

Note: After License Keys of IVR, Conference or Recording Server are added in the License Keys list, the system will need to be rebooted manually. Please consult your distributor for more details on License Keys.

Auto Discovery

Auto Discovery allows the PBX Server to detect all IP phones deployed on the network automatically when it is enabled. It is highly recommended to enable Auto Discovery to set up IP phones for the initial installation of IP extensions, this takes away the hassle of setting them up one by one.
Setting Default Class of Service
Before starting the Auto Discovery, set a default Class of Service that will be applied to all IP phones. Afterwards the PBX Server will detect the IP and MAC addresses of IP phones with Auto Discovery.
1. Go to Main Menu> Operation Management> Auto Discovery.
2. Select a CoS type in “Set Default CoS for Auto Discovery” list and click “Set Default”. A displayed arrow indicates to bring Class of Service selected back to default.

Please refer to Chapter 6 Group Management/ Class of Service for more details about Class of Service.

Enable/Disable Auto Discovery
When Auto Discovery is enabled, it allows the PBX Server to detect the settings of all the IP phones or SoftPhone available in the LAN environment.
1. Go to Main Menu> Operation Management> Auto Discovery.
2. Select “Turn On Auto Discovery” to enable Auto Discovery, so that Current Mode shows “Auto Discovery On”.

If Auto Discovery is not disabled after IP phones are registered to PBX Server, the system will disable Auto Discovery automatically after two hours to prevent unauthorized users from login into the InterPBX system.

Backup System Files
Before modifying settings or upgrading the system, it is recommended to create a new folder for system backup files, so that system settings that are backed up
can be restored and uploaded to the system if necessary. The recorded voicemail prompts and greetings will also be saved.

1. Go to Main Menu> Operation Management> Backup System Files.
2. The system will generate one “backup.dat” file while “Starting Backup” is selected.
3. Follow the instructions on how to back up the files and save the system files to a folder on a dedicated PC hard drive.

**Restore System Files**

Administrators can upload the system files which have been saved in the Backup folder when choosing to restore previous settings to the system.

1. Go to Main Menu> Operation Management> Restore System Files.
2. Click Browse button and select the backup path for “backup.dat” file.
3. Click Upload File button.
4. The software version of the backup file will be displayed by “Uploaded file information”. Make sure it is the right version that you need. Click Proceeding Restore Action button to continue the restore procedure.
After uploading system files to the system, the PBX Server needs to reboot. Click Restart Now button to reboot the system immediately or later. Note that there must be no ports occupied when the system reboots.

**Application Upgrades**

It is optional to upgrade the system with new releases or application software. Before upgrading, save the new release or new application software on PC or LAN environment where you can access the files for system upgrade.

1. Go to Main Menu> Operation Management> Application Upgrade.
2. Click Browse button and select the path and files for upgrade.
3. Click Upgrade button to upgrade the system.

You can also upgrade the software version for the PBX Server and IP phones in this section. IP phones will load the latest version from PBX Server automatically when connected to LAN. After the upgrade procedure is complete, the PBX Server needs to be rebooted. Click Restart Now button to reboot the system immediately or you can reboot later. Note that there must be no ports occupied when the system reboots.

**CDR**

DSG InterPBX Communication System provides Call Detail Records (CDR) which reports the activities of each extension, including raw data of date, time, extension number and numbers of outside calls.
1. Go to Main Menu> Operation Management> Download CDR Data.
2. Select the types of calls you want to download, such as international calls, long distance calls, local call or incoming calls.
3. Input the starting and ending dates to retrieve the collection of CDR.
4. Click Download CDR File button. A new page will pop-up and display the CDR summary. You can print it out or select Query.txt on the left upper corner of the page and right click the mouse to save data for further analysis.

CDR Report

- **Type**: The type of calls, such as long distance or local calls.
- **Call ID**: The unique serial number of each call.
- **Dialing SID**: The PBX Server ID initiating calls.
- **Dialed SID**: The PBX Server ID receiving calls.
- **Dialing Ext**: The extension number initiating calls or CO line extension number transmitting incoming calls.
- **Dialed Ext**: The extension number receiving calls or CO line extension number transmitting outgoing calls.
• **Dialled Number**: The phone numbers dialed.
• **Start Time/Duration**: Starting time and ending time of the calls. The unit is in seconds.
• **Auth. Code**: The authorization code been used to make this call.
• **Calling Name**: If your telephone company provides this service, calling party’s name, will be displayed as well.

**Note**: Blaze 1200 Series doesn’t provide the CDR function.

**Billing System**
InterPBX Communication System not only provides the basic CDR (Call Detail Records) but also supports the third party billing system such as “CAS2000”.
Please go to Main Menu> Operation Management> CDR. Enter the IP address of the third party’s billing system in the field “Billing System IP” and click save. Then the InterPBX can transfer the CDR data to the billing system. Please contact your distributor for more details.

**Note**: Blaze1200 IP-PBX Communication System doesn’t provide CDR data and only supports the third party’s billing system as below.

**Reboot**
Please do not switch off or unplug the power cord connected to the system directly to avoid having system files corrupted. Go to Main Menu> Operation Management> Reboot. Click Restart to reboot the system.
You may need to reboot the system under the following circumstances:

- After you upgrade/install PBX Server program or new application software
- After you restore the backup file to the system
- After you change the IP settings of your PBX Server

You may need to turn off the system under the following circumstances:

- Before you make any hardware changes or remove the PBX Server
- Before you expand gateway ports/modules

You do not need to reboot the system when you add, change IP addresses of IP phones or relocate IP extensions or Voice Gateways.

**Reset to Default**

Go to "Main Menu> Operation Management> Reset to Default. Enter User Name and password, select Reset button, and then reboot the system. All the settings will be brought back to default. Please refer to Appendix E System Parameters Default for more details about Reset to Default.
Chapter 9
Report Management

The InterPBX Communication System provides summaries including system information, gateways, joint servers, extension groups, CO line groups, system logs and all extensions for you to review, manage and search the various aspects of the system.
Chapter 9 Report Management

System Summary Report


- **System Information**: this includes System General Information, Business Hours, Call Restriction, System Speed Dialing, Operator Information, and License Information.

- **Gateways**: contains information about VoIP Gateways and Off-Premises VoIP Gateway ports.

- **Joint Servers Report**: shows system configuration about the Joint Server.

- **Extension Group Report**: displays Extension Group settings including Answering Options and members list of each Extension Group.

- **Trunk Group Report**: displays information about CO Line Group including name, access code, ring assignment and members list of each CO Line Group.

![System Summary Report](image)

System Log

The InterPBX Communication System keeps records of system performance and each action made by the administrator. Go to Main Menu> Report Management> System Log. You can review the logs including:

- Administrative Log: Displays information about system configuration made by administrators.

- System Status Log: Display all the activities of the system.
Extension Report

You can view extension number summaries in this section.

2. Select extension type(s) from All, InterPhone, SLT, Off-Premises or CO Line.
3. Select a report type from “Itemize” and “Comprehensives”.
4. Click the Generate Report button. The Extension Report will pop-up.
Chapter 10
Multi-Server Management

InterPBX Communication System employs a distributed NeuralServer Architecture that enhances the scalability and reliability of the system. The NeuralServer Architecture allows multiple PBX Servers to coexist, communicate and even backup each other. With Joint Server technology, companies have the option to interconnect PBX Server located in different branches via public IP networks or VPN connections.
Server Information
When administrating more than one PBX Server, either for the purpose of communication or backup, you need to assign Server IDs for each of the PBX Server to be connected.

1. Go to Main Menu>Multi-Server Management>Server Information.
2. Assign an ID to this PBX Server at “Server ID”. Each PBX Server should have a unique Server ID.
3. Assign the Urgent Event Notification sequence. Select the preferred method from extension, pager or phone then set the associated phone number.
4. Set the number of interval trials when InterPBX attempts to make a notification.
5. You can also input the administrator’s email address in “Send Urgent Event to E-mail” to receive urgent notifications.

Redundant Server
InterPBX Communication System supports system level redundancy, which means that you can designate a second PBX Server as a Slave. The Master and Slave will keep synchronizing data so that in the event of a Master PBX Server failure, the Slave will take over the call processing activities and manage the associated gateways and extensions.

When the Master returns to normal operation, the Slave will give the control back to the Master and synchronize all the data that is different or has changed starting from when the Slave took over. There is no time lag during the shift and the extension users will not notice the process.
Assigning Slave PBX Server

On both Master and Slave PBX Server, you need to assign each system’s role so it can become the redundant server and back up each in the event of an emergency.

1. From the Master PBX Server, go to Main Menu>Multi-Server Management>Redundant Server.
2. Select “Master” to assign this PBX Server as the primary server.
3. Input the Slave PBX Server’s Server ID.
4. Input the Slave PBX Server’s IP address.
5. Set the timeout allowing the Slave to take over the control after this set period. The timeout unit used is in seconds.

After the settings are complete in this section, Master will synchronize the parameters on this InterPBX to the Slave. You then need to install the Slave PBX Server and assign the Master PBX Server as the redundant server.

Note: The redundancy does not cover each PBX Server’s own problems such as power shortage or other reasons such as Voice Gateway problems. Regarding Power Failure Transfer feature, see Chapter 2 Installing InterPBX/Installing VG5000 Voice Gateway.

Installing and Setting Slave PBX Server

The settings on the Slave PBX Server are very straightforward. All you need to do is to configure Slave PBX Server’s IP settings, Server ID and the redundant settings.
1. Install Slave PBX Server, go to Slave PBX Server’s Main Menu>System Configuration>System Parameters then enter the IP address, Gateway IP and Subnet Mask. For more details, please see Chapter 2 Installing InterPBX Communication System/Installing PBX Server and Setup PBX Server’s IP Connectivity. Make sure Slave PBX Server’s IP address is accessible (ping) by Master PBX Server.

2. Next, go to Slave PBX Server’s Main Menu>Multi-Server Management>Server Information and assign the Server ID. The Slave’s Server ID should be different from Master’s ID.

3. Finally go to Main Menu>Multi-Server Management>Redundant Server and click on “Slave” to set this PBX Server as the Slave.

4. Input the Master PBX Server’s Server ID at “Redundant Server ID”.

5. Input the Master PBX Server’s IP address at “Redundant Server IP”.

6. Input the same timeout you have set on the Master.

**Joint Server**

If your company has multiple locations and each site has an InterPBX Communication System, you have the choice to link them via Joint Server technology to enjoy the best InterPBX has to offer. The communication between your site and the remote locations will be carried over your private or public IP networks saving this way all the inter-office communication costs. Taking advantage of the Joint Servers, your system now will have more cost-efficient paths to remote customers.

**Creating Joint Server**
1. Go to Main Menu>Multi-Server Management>Joint Server. Click the Add button to create a Joint Server.

2. Input remote PBX Server’s Server ID at “Master Server ID”.

3. Input the remote PBX Server’s IP address at “Master Server IP”. If the remote PBX Server is installed behind NAT, select the “Behind NAT” check box and input the remote NAT equipment’s public IP address here.

4. If the remote PBX Server has a redundancy, input its Slave Server ID and IP address accordingly.

5. When you have off-premises extensions, select “Access Local through NAT” check box if your system and the Joint Server communicate via VPN or under the same NAT device, or uncheck if both sides have an independent NAT device.

6. If the remote Joint Server supports QoS, you may click on QoS to activate this feature.

**Note:** When Joint Server is activated, if your PBX Server is behind NAT, you will need to open the UDP port 6055. The same principle applies to the unit linked by Joint Server, if this also is behind NAT, you will also need to open the UDP port 6055 on the NAT equipment.

**Note:** If the remote network is under NAT, in the event of a failure, you will not be able to communicate with the Joint Server’s redundancy server when it takes over the control. This is because the remote Master and Slave server are connected to the Internet via the same NAT public IP address. If remote Master cannot resume communication shortly, Joint Server’s administrator will need to open a specific port on the remote NAT equipment and direct it to the Slave.
Note: You have to set your system information on the Joint Server setting section of the remote PBX Server.

Setting Joint Server CO Lines Access Control

After you create a Joint Server, you will have access to remote IP phones, SLT ports and CO Line ports via the links between your InterPBX and the Joint Servers. You can then even make calls through your Joint Server’s CO lines to help reduce internal and external communication costs.

For Local Users
1. Select the “Allow Local Access Remote CO” check box allowing local extensions to connect to Joint Server’s CO Lines.
2. Create a password for local extensions, as this will be required for the extensions to connect to Remote CO Lines. You may leave it blank if you do not wish to set a password.

For Remote Users

To allow Jointer Server’s users to have access to your CO Lines, select “Allow Remote Access Local CO”.

Select the COS from the drop down menu to grant remote users call permissions after accessing your system.

Please note that you have to set Joint Server’s CO Line extension numbers in “Joint Server Member”. Additionally, the Joint Server Administrator has to enable the “Allow Remote Access Local CO” feature.

To place a call via Joint Server’s CO Lines
1. Lift the handset and dial the remote CO Line Extension Number or CO Group Number.
2. Follow the prompt to enter the password.
3. After you hear the dial tone, dial the phone number you would like to call.
   For long distance or international calls, follow the remote site’s dialing habit to place the call.

Setting Links with Joint Server

To ensure the best voice quality and get the most out of the resource used, a good management of the links is necessary since the voice channels created between you and Joint Server are limited to the bandwidth.
1. At “Max Link” box, input the maximum number of calls that can be placed to a Joint Server at a time. The availability of the voice channels is based on the bandwidth between your PBX Server, Joint Server and the CODEC used. When calls reach the bandwidth limit, the system will generate busy tones to the next caller who tries to access a link.

2. Select a CODEC from the list to be used for calls between you and your Joint Server. InterPBX Communication System supports PCM, G.723.1, and G.729a CODECs. Choose one of the CODECs according to the available bandwidth. Keep in mind that PCM’s compression rate is 64 Kbps, it provides excellent voice quality but consumes more bandwidth. G.723.1 uses 6.3 Kbps, much less bandwidth consumption but might provide poorer voice quality. The compression rate of G.729a is 8 Kbps.

3. Set the “Jitter Buffer Depth” based on the bandwidth and the CODEC selected. As Jitter Buffer is divided into None, Low, Middle and High, always select it from the lowest value as the system will auto adjust it according to your usage. None corresponds to 0 packets and 2, 4, 6 corresponds to the number of packets in Low, Middle and High respectively. Lower selections translate into faster packet transfers while higher selections guarantees that no packets are lost during transfers.

**Setting Joint Server Extension List**

After you create a Joint Server, you need to further specify its extension numbers to be reached from your side. The extension numbers might include internal extension numbers, CO Line extension numbers or CO Line Group numbers. Make sure that the extension numbers of Joint Server need to be different from the ones on your system.
2. Select a Joint Server from the list and click the Members button.
3. Input the remote extension number range that can be reached from your side and click the Add button. The extension numbers might include IP phones, analog phones and CO Line port extension numbers. You can also input remote CO Line Group numbers with the associated CO Line extension numbers. When you edit the CO Line extension blocks, click the “CO Line Group?” check box so that users will be required to enter the password when connecting to these CO Line ports. This will protect Joint Server’s CO Line resources from being used by unauthorized users. If unchecked, the password will not be required when accessing these CO Line extension numbers.

**Joint Server Status**

You could view all your Joint Servers from the Joint Server List. When the Joint Server is alive and connected, you could find a “@” sign at the end of the entry.

![Joint Server List](image)

When making internal calls between Joint Servers, the extension number and name will be carried to the Joint Server. The general functions such as paging, internal notification, conference will also function as under one system.
# Appendix A: Function Code List

During a call, if you want to activate a function, please press the Transfer button first and then enter the function code. For example, if you want to initiate a conference call, please put all the members on hold. Then press the Transfer button and #40. Function codes can be customized by the system administrator. Below are the default settings of function access code.

**Note:** Some functions require proper settings or enabled by the system administrator before you can access them.

<table>
<thead>
<tr>
<th>Function</th>
<th>Feature access code</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Access Ext.</td>
<td>An extension number of a</td>
<td>To make calls to the assigned no.</td>
</tr>
<tr>
<td></td>
<td>station, group, CO line,</td>
<td>For an outbound call, start with</td>
</tr>
<tr>
<td></td>
<td>CO line group or a phone</td>
<td>your CO line access code.</td>
</tr>
<tr>
<td></td>
<td>no.</td>
<td></td>
</tr>
<tr>
<td>After Call Work</td>
<td>#48</td>
<td>To allow agents to have a period of after call work time between two calls.</td>
</tr>
<tr>
<td>Ask Member Login</td>
<td>#37 + Ext. Group No.</td>
<td>Check the login status of the extension group. Hearing a dial</td>
</tr>
<tr>
<td></td>
<td></td>
<td>tone means you are logged-in successfully and a busy tone means not</td>
</tr>
<tr>
<td></td>
<td></td>
<td>logged-in.</td>
</tr>
<tr>
<td>Auto Line Access</td>
<td>CO Line Access Code</td>
<td>To get a CO line. In most cases, it is “0” or “9”.</td>
</tr>
<tr>
<td>Auto-In</td>
<td>#46</td>
<td>Allow agents to pick up the next call automatically.</td>
</tr>
<tr>
<td>Auxiliary Time</td>
<td>#45</td>
<td>Allow agents to notify the system stop assigning incoming calls so that</td>
</tr>
<tr>
<td></td>
<td></td>
<td>the agents can leave their seats temporarily without logging out the</td>
</tr>
<tr>
<td></td>
<td></td>
<td>system.</td>
</tr>
<tr>
<td>Call Appearance</td>
<td>A specified programmable</td>
<td>Press to make or receive calls. It functions similar to intercom. When</td>
</tr>
<tr>
<td></td>
<td>button</td>
<td>more than one button is assigned,</td>
</tr>
<tr>
<td>Function</td>
<td>Feature access code</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------------</td>
<td>--------------------</td>
<td>--------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Call Hold</td>
<td>Hold Button or #12 (on analog phones)</td>
<td>To place a call on hold. Press again to retrieve the call.</td>
</tr>
<tr>
<td>Call Hold Retrieve CO</td>
<td>#13 + CO Line Extension No.</td>
<td>To retrieve an incoming call placed on hold. (You should be able to see the CO line extension number for using this function.) If this CO line is set on the programmable key, you can directly press the flashing button to retrieve the incoming call.</td>
</tr>
<tr>
<td>Call Hold Retrieve Ext.</td>
<td>#14 + Extension Number</td>
<td>To retrieve an internal call placed on hold. If this extension number is set on the programmable key, you can directly press the flashing button to retrieve the call.</td>
</tr>
<tr>
<td>Call Park</td>
<td>Transfer + #15 + Slot Number (0-9)</td>
<td>To park a call to a specific slot. To retrieve, press #15 and the specific slot number.</td>
</tr>
<tr>
<td>Call Pickup CO Line</td>
<td>#10</td>
<td>To answer the least recent incoming call ringing on the system.</td>
</tr>
<tr>
<td>Call Pickup Directed</td>
<td>#11 + Extension Number</td>
<td>To answer a call ringing at another extension.</td>
</tr>
<tr>
<td>Call Pickup Group</td>
<td>Group Call Pickup Code</td>
<td>To answer a call ringing at another extension in your call pickup group. (Please consult your administrator for Extension Group Call Pickup Code.)</td>
</tr>
<tr>
<td>Call Waiting</td>
<td>Hold or Hook Flash (on analog phones)</td>
<td>Users can place the current call on Hold and answer another incoming call. Press Hold button again to retrieve the previous call on Hold. (Please enable Call Waiting function.)</td>
</tr>
<tr>
<td>Conference Call</td>
<td>Transfer + #40</td>
<td>Start Conference with callers placed on hold.</td>
</tr>
<tr>
<td>DND Set</td>
<td>#17 + 1 (Enable DND)</td>
<td>Enable or disable Do-Not-Disturb</td>
</tr>
<tr>
<td>Function</td>
<td>Feature access code</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------</td>
<td>--------------------</td>
<td>--------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>External Paging</td>
<td>The External Paging Code of a specific Voice Gateway</td>
<td>Broadcast through an external amplifier connected to Voice Gateway. Please consult your administrator for the external paging code.</td>
</tr>
<tr>
<td>Forward All Calls</td>
<td>#44 + Extension Number + #</td>
<td>To forward all the incoming calls to a specific extension or an external phone number automatically. Press again to disable. When setting external number, please add the CO line access code such as 0.</td>
</tr>
<tr>
<td>Headset</td>
<td>A specified programmable button</td>
<td>Press to allow voice been transmitted from the attached headset, instead of the handset.</td>
</tr>
<tr>
<td>Internal Paging</td>
<td>#38 + Extension Number or Extension Group Number</td>
<td>Broadcast through the extension or the extension group.</td>
</tr>
<tr>
<td>Manual-In</td>
<td>#47</td>
<td>To allow agents to pick up the next incoming call manually by pressing the specific button.</td>
</tr>
<tr>
<td>Member Login</td>
<td>#35 + Extension Group No.</td>
<td>Login to be one of the members in the extension group.</td>
</tr>
<tr>
<td>Member Logoff</td>
<td>#36 + Extension Group No.</td>
<td>Logout from the extension group.</td>
</tr>
<tr>
<td>Personal Speed Dial</td>
<td>#21 + Personal Speed Dial Number(e.g. 00)</td>
<td>To dial a number defined on the Personal Speed Dial Number.</td>
</tr>
<tr>
<td>Record/Play (Store or Record on Demand)</td>
<td>#41</td>
<td>To save recorded calls or start recording. For Store on Demand, recordings from the beginning of the call will be saved. For Record on Demand, recording will start after enable the function.</td>
</tr>
<tr>
<td>Retrieve Message</td>
<td>Message button or ## (on analog phones)</td>
<td>To access mailbox.</td>
</tr>
<tr>
<td>Retrieve Record</td>
<td>#42</td>
<td>To play the saved recording. This function needs to be enabled before</td>
</tr>
<tr>
<td>Function</td>
<td>Feature access code</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------</td>
<td>---------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>System Speed Dial</td>
<td>#20 + System Speed Dial Number (e.g. 000)</td>
<td>To dial a number defined on the System Speed Dial Number.</td>
</tr>
<tr>
<td>Transfer to Ext VM</td>
<td>#24 + Extension No.</td>
<td>To transfer a call to the extension’s voice mail.</td>
</tr>
<tr>
<td>Transfer to AA Tree</td>
<td>#25 + AA Menu Access Code</td>
<td>To transfer a call to the specific AA menu.</td>
</tr>
<tr>
<td>Night Service</td>
<td>#43</td>
<td>To switch the operation mode to Night. AA menu and operators will be changed accordingly. This function needs to be enabled before accessing.</td>
</tr>
<tr>
<td>Virtual Extension</td>
<td>A specified programmable button</td>
<td>Press to act as the assigned virtual extension to make calls.</td>
</tr>
</tbody>
</table>
Appendix B: Retrieving Voice Messages and Recordings

Listening messages from your extension
1. Press the Message button to access mailbox. (Or press ##)
2. Enter your password and then press #. (The default password is blank.)
3. The system will announce the number of messages you have. Please follow the system prompts and press 1 to play the new message, press 2 to play the old messages, press 3 to send a message, or press 4 to change personal options. In the mailbox, you can press * to return to the previous menu or press # to return to the main menu.
4. When you are listening to the messages, you can refer to the following diagram.

![Diagram of Voice Mailbox Options]

Listening messages from the other extension
1. Press the Message button to access mailbox. (Or press ##)
2. Press * button to go back to the main menu of VMS.
3. Follow the system prompts to enter the extension number and password and then press #. (The default password is blank.)
4. The system will announce the number of messages you received. Please follow the system prompts to listen to messages or setup your mail box.
This function also allows you to play the message for virtual extensions or extension groups.

**Listening messages from AA menu**

If the Administrator has assigned a DTMF key as Access Voice Mail in the AA menu, you can listen to your voice messages from other extensions or from any telephones.

1. Call your company’s phone number. The auto attendant will answer your call.
2. Press the dedicated DTMF key for accessing the voice mail system.
3. Input your extension number and your password.
4. Follow the system prompts to listen to your messages, send a message or edit your voice mailbox.

You may also listen to your voice messages from other extensions.

1. Dial the access code of AA Menu.
2. Enter the specific key to access to your mailbox.
3. Enter your extension number and password.
4. Follow the system prompts to listen to your message, send a message, or change your mailbox preferences.

This function also allows you to play the message for virtual extensions or extension groups.

**Listening to Messages from E-mail**

InterPBX supports Unified Messaging function that allows you to receive your voice message via e-mail. When you have new messages, the system will send you an attachment in wav format by e-mail. You can open the attachment to listen to the message.

Please be sure if the following settings are properly set in order to make this function work properly.

1. The administrator has to enable this function and also set the SMTP server on the System Parameter option.
2. Your e-mail address must be set on the Notification option. Please refer to User’s Guide.

**Using the Recording Function**

Blaze5000 provides built-in Record on Demand and/or Store on Demand function. The system will keep on recording calls for users on the Store on Demand list and users on the list can have the option to save the recording or not. Blaze 1200 provides built-in Record on Demand allowing users to record calls after enable the function. A professional recording
system is required by Savanna Series.

**Activate the Recording**
1. During a call, press the specific programmable button or REC/PLAY button to record.
2. After the call completes, the system will stop recording automatically.

**Play Recordings**
1. Press the specific programmable button or REC/Play button or function code #42 to play recordings.
2. Input the password followed by the # key to access the system. (The default password is blank.)
3. Follow the system prompts to retrieve or search recordings.

[Diagram of Play Options]

- 1. Play the Last
- 2. Search by Date/Time
  - Enter the Date (A.D. YYMMDD)
  - Enter the Time (24 Hours Format HHMM)
  - Play the Previous Recording
  - Play the Next Recording
  - # (Return to the Extension Option)
  - * (Return to the Previous Option)
- 3. Reply
- 4. Sends the voice messages to personal email account (UMS function must be activated)
Appendix C: InterPBX Management Website

InterPBX system provides a web-based extension management tool allowing extensions users to edit personal extensions. You can edit options including Button Mapping, Station Speed Dialing, Answer Option, Mailbox, Message Notification, Distribution List, and Change Password.

Login to the Management Website

1. Open the browser and type in your PBX Server IP address in the address bar.

2. Click the User Login icon.

3. On the pop-up window, please key in the following information.
   - Extension No.: Enter your extension number.
   - Password: Enter your password. The default is blank if the extension is created by Auto Discovery.

Setting Items

You can edit personal Button Mapping, Speed Dialing, Answer Option, Mailbox, Notification, Distribution List or Change Password. Please refer to IP Phone User’s Guide for more details.
### Appendix C InterClient Utility

#### Setting of Extension 7128

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>First Name</td>
<td>John</td>
</tr>
<tr>
<td>Last Name</td>
<td>Jane</td>
</tr>
<tr>
<td>Department</td>
<td>Sales</td>
</tr>
<tr>
<td>Title</td>
<td>Manager</td>
</tr>
<tr>
<td>Location</td>
<td>Office</td>
</tr>
</tbody>
</table>

![Image of InterClient Utility interface](image-url)
Appendix D: Terminologies

802.1p: An IEEE standard for QoS. It uses 3 bits allowing switches to reorder packets based on priority level.

802.1Q: An IEEE standard for QoS. It provides VLAN identification, priority and QoS levels.

10BaseT: A type of IEEE802.3 Ethernet network providing 10 M bps with a maximum segment length of 100 meters.

Auto Attendant: A voice mail system feature answering calls with prerecorded voice prompts to guide callers.

Buffer: A temp storage area that compensates for a difference in transmission speeds timing.

Caller ID: A telephone company service displaying callers’ number and/or name.

CDR: Call Detail Records. A database record contains data about a specific call.

CO: Central Office. A telephone company site that hosts PSTN switching equipment.

CO Line: Also known as trunk line. A telephone line from a telephone company.

CoS: Class of Service. A classification assigned to users giving them different rights or privileges.

CODEC: Coder/Decoder. A hardware or software method converts analog audio or video signals to digital and vice versa.

Dial Plan: The national numbering plan defines area codes and long distance codes.


DNS: Domain Name System. It is used for translating name of nodes into IP addresses.

DTMF: Dual Tone Multi-Frequency. The pushed keypad button generates a pair of tones for identifying the button.

FXO: Foreign Exchange Office. An analog trunk side connection between a CO and PBX system.

FXS: Foreign Exchange Station. An analog line side connection between a PBX system and a POTS telephone.

G.711: Also known as PCM. A compression standard encoding/decoding
of speech at 64 Kbps.

**G.723.1**  
A compression standard encoding/decoding of speech at 6.4/5.3 Kbps.

**G.729**  
A compression standard encoding/decoding of speech at 8 Kbps.

**IP Address**  
An 8-bit number (in IPv4) to network enabled device accessing the Internet.

**IP PBX**  
Internet Protocol PBX. An IP PBX system connects endpoints, such as Voice Gateway or IP phones via Ethernet LAN and uses IP packets for transmitting voice conversations and call signals.

**Joint Server**  
A two or more InterServers able to communicate with each other regardless their locations and form a logical single system.

**Jitter**  
A fluctuation of in transmission of signal or packets.

**MAC address**  
Media Access Control address. Also known as hardware address. A unique 48-bit number that stored in a device to identify it on LAN.

**Off-hook**  
A state a telephone handset is lifted off.

**On-hook**  
A state a telephone handset is placed on the hook.

**Paging**  
A public announcement system via external paging equipment (external paging) or speakers on phone sets (internal paging).

**PFT**  
Power Failure Transfer. When the AC power fails and there is no backup power, some trunks connected to the phone system will be switched to single line phones.

**PoE**  
Power over Ethernet. It can transmit DC power to the target device via Ethernet cable 4, 5, 7, 8 pins. It allows the target device such as IP phones not to use AC power outlet.

**Port**  
An interface on a device capable to connect to another device.

**PSTN**  
Public-Switched Telephone Network. The traditional telephone network.

**Voice Gateway**  
A network device connects TCP/IP networks and PSTN transmitting voice packets.

**QoS**  
Quality of Service. A mechanism that defines the performance or priority of packets.

**Redundancy**  
A backup for components of a system. Should one of them fail, the system continues to operate.

**SMTP**  
Simple Mail Transfer Protocol. The TCP/IP protocol for transmitting emails.

**VLAN**  
Virtual LAN. A logical subgroup within a LAN combining workstations or devices regardless their physical segments.
# Appendix E: Default Values

## InterServer Default

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>IP Address</td>
<td>192.168.1.200</td>
</tr>
<tr>
<td>Default Gateway</td>
<td>192.168.1.254</td>
</tr>
<tr>
<td>Subnet Mask</td>
<td>255.255.0.0</td>
</tr>
<tr>
<td>DNS</td>
<td>168.95.1.1</td>
</tr>
<tr>
<td>Login User Name</td>
<td>Admin</td>
</tr>
<tr>
<td>Password</td>
<td>0000</td>
</tr>
</tbody>
</table>

## VG5000 Default

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>IP Address</td>
<td>192.168.1.201</td>
</tr>
<tr>
<td>Default Gateway</td>
<td>192.168.1.254</td>
</tr>
<tr>
<td>Subnet Mask</td>
<td>255.255.0.0</td>
</tr>
<tr>
<td>PBX Server</td>
<td>192.168.1.200</td>
</tr>
<tr>
<td>Password</td>
<td>1234</td>
</tr>
</tbody>
</table>

## IP Phone Default

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>IP Address</td>
<td>192.168.1.100</td>
</tr>
<tr>
<td>Default Gateway</td>
<td>192.168.1.254</td>
</tr>
<tr>
<td>Subnet Mask</td>
<td>255.255.0.0</td>
</tr>
<tr>
<td>DHCP</td>
<td>On</td>
</tr>
<tr>
<td>PBX Server</td>
<td>192.168.1.200</td>
</tr>
</tbody>
</table>
Appendix F: DTMF Programming

There is an alternative way of programming VMS Server. Through a touch-tone telephone, you can edit system parameters or record greeting from anywhere.

**Entering the DTMF Programming Mode**

1. Call from a telephone to one of the CO lines (e.g. Tel. 5551234) connected to the IP-PBX or call into an AA Menu access code (e.g. 4700).
2. Press the key predefined for entering the DTMF Programming Mode (e.g. # button). The key should be defined on the AA menu. Please refer to Chapter 7 Voice Mail Configuration for more details.
3. You will be prompted to enter the password for system-wide settings (the default System Password is “1234”) or to record greeting (the Greeting Recording Password is “0000”).
4. If you have entered the correct password, you will be prompted to enter the 3-digit function code. Please enter the 3-digit function code.
5. You will be prompted the current setting of the function you wish to edit. Please press 1 to edit, press 2 to save or press 3 to replay. Refer to the DTMF and Action Code Table and follow the prompt to edit settings.

**Note**: Please note that if you entered the Greeting Recording Password, you are only allowed to input function code “330” to record AA menu greetings. The AA menu ID is 3-digit (000-299).

**DTMF and Action Code Table:**

Use the code below when entering DTMF signals.

<table>
<thead>
<tr>
<th>Signal</th>
<th>Telephone Input Code</th>
</tr>
</thead>
<tbody>
<tr>
<td>DTMF 1</td>
<td>01</td>
</tr>
<tr>
<td>DTMF 2</td>
<td>02</td>
</tr>
<tr>
<td>DTMF 3</td>
<td>03</td>
</tr>
<tr>
<td>DTMF 4</td>
<td>04</td>
</tr>
<tr>
<td>DTMF 5</td>
<td>05</td>
</tr>
<tr>
<td>DTMF 6</td>
<td>06</td>
</tr>
<tr>
<td>DTMF 7</td>
<td>07</td>
</tr>
<tr>
<td>DTMF 8</td>
<td>08</td>
</tr>
<tr>
<td>DTMF 9</td>
<td>09</td>
</tr>
</tbody>
</table>
### Signal Telephone Input Code

<table>
<thead>
<tr>
<th>Signal</th>
<th>Telephone Input Code</th>
</tr>
</thead>
<tbody>
<tr>
<td>DTMF *</td>
<td>10</td>
</tr>
<tr>
<td>DTMF 0</td>
<td>11</td>
</tr>
<tr>
<td>DTMF #</td>
<td>12</td>
</tr>
<tr>
<td>First Flash</td>
<td>13</td>
</tr>
<tr>
<td>Second Flash</td>
<td>14</td>
</tr>
<tr>
<td>0.5 sec Pause Time</td>
<td>15</td>
</tr>
<tr>
<td>Extension</td>
<td>16</td>
</tr>
<tr>
<td>1 sec Pause Time</td>
<td>17</td>
</tr>
<tr>
<td>Timeout</td>
<td>18</td>
</tr>
<tr>
<td>DTMF A</td>
<td>19</td>
</tr>
<tr>
<td>DTMF B</td>
<td>20</td>
</tr>
<tr>
<td>DTMF C</td>
<td>21</td>
</tr>
<tr>
<td>DTMF D</td>
<td>22</td>
</tr>
<tr>
<td>Extension Digit</td>
<td>23</td>
</tr>
<tr>
<td>Ignore</td>
<td>24</td>
</tr>
<tr>
<td>New Message Number</td>
<td>25</td>
</tr>
</tbody>
</table>

### VMS 3-Digit Function Code List:

<table>
<thead>
<tr>
<th>Func. Code</th>
<th>Description</th>
<th>Input format</th>
<th>Valid Range</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>100</td>
<td>System Password</td>
<td>X + #</td>
<td>X = 0~8 digits</td>
<td>1234</td>
</tr>
<tr>
<td>101</td>
<td>Greeting Recording Password</td>
<td>X + #</td>
<td>X = 0~8 digits</td>
<td>0000</td>
</tr>
<tr>
<td>102</td>
<td>IP Address</td>
<td>Read only</td>
<td>-</td>
<td>192.168.1.200</td>
</tr>
<tr>
<td>103</td>
<td>Default Gateway Address</td>
<td>Read only</td>
<td>-</td>
<td>192.168.1.254</td>
</tr>
<tr>
<td>104</td>
<td>Subnet Mask</td>
<td>Read only</td>
<td>-</td>
<td>255.255.0.0</td>
</tr>
<tr>
<td>105</td>
<td>DNS</td>
<td>Read only</td>
<td>-</td>
<td>168.95.1.1</td>
</tr>
<tr>
<td>106</td>
<td>Server IP</td>
<td>Read only</td>
<td>-</td>
<td>192.168.1.200</td>
</tr>
<tr>
<td>109</td>
<td>DTMF Inter-Digit Timeout</td>
<td>X + #</td>
<td>X = 1 ~ 10 sec(s)</td>
<td>3 sec(s)</td>
</tr>
<tr>
<td>110</td>
<td>DTMF AA-Menu Timeout</td>
<td>X + #</td>
<td>X = 1 ~ 10 sec(s)</td>
<td>5 sec(s)</td>
</tr>
<tr>
<td>121</td>
<td>VMS Server Version Number</td>
<td>Read only</td>
<td>-</td>
<td>2.00e</td>
</tr>
<tr>
<td>Func Code</td>
<td>Description</td>
<td>Input format</td>
<td>Valid Range</td>
<td>Default</td>
</tr>
<tr>
<td>-----------</td>
<td>-------------------</td>
<td>--------------</td>
<td>----------------------------</td>
<td>---------------</td>
</tr>
<tr>
<td>125</td>
<td>Name Directory Listing</td>
<td>X + #</td>
<td>X = 0 (first name), 1(last name)</td>
<td>0 = First Name</td>
</tr>
<tr>
<td>128</td>
<td>Transfer Option Timeout</td>
<td>X + #</td>
<td>X = 1 ~ 10 sec(s)</td>
<td>1 sec(s)</td>
</tr>
<tr>
<td>129</td>
<td>Default Language</td>
<td>X + #</td>
<td>1-4</td>
<td>1</td>
</tr>
</tbody>
</table>
| 330       | AA-Menu Action    | AA-Menu ID = 000-299 | Action:  
00 = No Action  
01 = AA Menu  
02 = AA Menu in 1st Language  
03 = AA Menu in 2nd Language  
04 = AA Menu in 3rd Language  
05 = AA Menu in 4th Language  
06 = Lead to Extension Number  
07 = Transfer to  
DTMF 1 =  07  
DTMF 2 =  07  
DTMF 3 =  07  
DTMF 4 =  07  
DTMF 5 =  07  
DTMF 6 =  07  
DTMF 7 =  07  
DTMF 8 =  07  
DTMF 9 =  07  
DTMF * =  11  
DTMF 0 =  13  
Timeout =  13 | 001-004         |

Steps:  
1. Select one options from 1 to 4. (1 edit, 2 add, 3 delete, 4 review).  
2. Input AA Menu ID to edit, add or delete.  
3. Input DTMF Code + * + Action (+ * + Target) + #  

AA-Menu ID = 000-299  
Action:  
00 = No Action  
01 = AA Menu  
02 = AA Menu in 1st Language  
03 = AA Menu in 2nd Language  
04 = AA Menu in 3rd Language  
05 = AA Menu in 4th Language  
06 = Lead to Extension Number  
07 = Transfer to  
DTMF 1 =  07  
DTMF 2 =  07  
DTMF 3 =  07  
DTMF 4 =  07  
DTMF 5 =  07  
DTMF 6 =  07  
DTMF 7 =  07  
DTMF 8 =  07  
DTMF 9 =  07  
DTMF * =  11  
DTMF 0 =  13  
Timeout =  13 |
<table>
<thead>
<tr>
<th>Func. Code</th>
<th>Description</th>
<th>Input format</th>
<th>Valid Range</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Extension</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>08 = Direct Transfer to Extension</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>09 = Transfer to Mailbox</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>10 = Direct Transfer to Mailbox</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>11 = Access Mailbox</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>12 = Name Directory</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>13 = Operator</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>14 = System</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>Programming Mode</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>15 = Repeat Menu</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>16 = Hang Up</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>17 = Hang Up with Announcement</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>Target = straight numbers</td>
<td></td>
<td></td>
</tr>
<tr>
<td>350</td>
<td>Transfer Options</td>
<td>DTMF Code + * + Action (* + Target) + #</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>Action:</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>00 = No Action</td>
<td>DTMF 1 =</td>
<td>03</td>
</tr>
<tr>
<td></td>
<td></td>
<td>01 = AA Menu</td>
<td>DTMF 2 =</td>
<td>03</td>
</tr>
<tr>
<td></td>
<td></td>
<td>02 = Lead to Extension</td>
<td>DTMF 3 =</td>
<td>03</td>
</tr>
<tr>
<td></td>
<td></td>
<td>03 = Transfer to Extension</td>
<td>DTMF 4 =</td>
<td>03</td>
</tr>
<tr>
<td></td>
<td></td>
<td>04 = Leave a Message</td>
<td>DTMF 5 =</td>
<td>03</td>
</tr>
<tr>
<td></td>
<td></td>
<td>05 = Operator</td>
<td>DTMF 6 =</td>
<td>03</td>
</tr>
<tr>
<td></td>
<td></td>
<td>06 = Repeat Menu</td>
<td>DTMF 7 =</td>
<td>03</td>
</tr>
<tr>
<td></td>
<td></td>
<td>07 = Hang Up with Announcement</td>
<td>DTMF 8 =</td>
<td>03</td>
</tr>
<tr>
<td></td>
<td></td>
<td>08 = Hold for Busy</td>
<td>DTMF 9 =</td>
<td>03</td>
</tr>
<tr>
<td></td>
<td></td>
<td>09 = Forward External</td>
<td>DTMF * =</td>
<td>04</td>
</tr>
<tr>
<td></td>
<td></td>
<td>10 = Hang Up</td>
<td>DTMF 0 =</td>
<td>05</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Target = straight numbers</td>
<td>DTMF # =</td>
<td>00</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Timeout =</td>
<td>04</td>
</tr>
<tr>
<td>Func. Code</td>
<td>Description</td>
<td>Input format</td>
<td>Valid Range</td>
<td>Default</td>
</tr>
<tr>
<td>------------</td>
<td>------------------------------</td>
<td>--------------</td>
<td>------------------------------</td>
<td>-------------</td>
</tr>
<tr>
<td>351</td>
<td>Announce Call Transfer</td>
<td>X + #</td>
<td>X = 0 (disable), 1 (enable)</td>
<td>1 = enabled</td>
</tr>
<tr>
<td>352</td>
<td>Announce Name</td>
<td>X + #</td>
<td>X = 0 (disable), 1 (enable)</td>
<td>1 = enabled</td>
</tr>
<tr>
<td>353</td>
<td>Leave Message Directly</td>
<td>X + #</td>
<td>X = 0 (disable), 1 (enable)</td>
<td>0 = disabled</td>
</tr>
<tr>
<td>406</td>
<td>Ring Notification Interval</td>
<td>X + #</td>
<td>X = 1 ~ 300 min(s)</td>
<td>1 min(s)</td>
</tr>
<tr>
<td>407</td>
<td>Ring Notification Try</td>
<td>X + #</td>
<td>X = 1 ~ 9 try(s)</td>
<td>3</td>
</tr>
<tr>
<td>408</td>
<td>Ring Count</td>
<td>X + #</td>
<td>X = 10 ~ 50 sec(s)</td>
<td>30 sec(s)</td>
</tr>
<tr>
<td>409</td>
<td>Pager Mode</td>
<td>X + #</td>
<td>X: 0 = USA, 1 = Singapore</td>
<td>0 = USA</td>
</tr>
<tr>
<td>410</td>
<td>Detect Pager Vox</td>
<td>X + #</td>
<td>X = 0 (disable), 1 (enable)</td>
<td>0 = disabled</td>
</tr>
<tr>
<td>411</td>
<td>Delay for Pager</td>
<td>X + #</td>
<td>X = 0 ~ 20 sec(s)</td>
<td>0 sec(s)</td>
</tr>
<tr>
<td>414</td>
<td>Maximum Number of Messages</td>
<td>X + #</td>
<td>X = 1 ~ 255 message(s)</td>
<td>20 message(s)</td>
</tr>
<tr>
<td>415</td>
<td>Maximum Message Length</td>
<td>X + #</td>
<td>X = 10 ~ 600 sec(s)</td>
<td>60 sec(s)</td>
</tr>
<tr>
<td>416</td>
<td>Minimum Message Length</td>
<td>X + #</td>
<td>X = 1 ~ 9 sec(s)</td>
<td>1 sec(s)</td>
</tr>
<tr>
<td>417</td>
<td>Auto Purge New Messages</td>
<td>X + #</td>
<td>X = 0 (disable), 1 ~ 30 day(s)</td>
<td>7 days</td>
</tr>
<tr>
<td>418</td>
<td>Auto Purge Old Messages</td>
<td>X + #</td>
<td>X = 0 (disable), 1 ~ 30 day(s)</td>
<td>7 days</td>
</tr>
<tr>
<td>419</td>
<td>Silence Timeout to Stop Recording</td>
<td>X + #</td>
<td>X = 0 ~ 60 sec(s)</td>
<td>0 sec(s)</td>
</tr>
<tr>
<td>420</td>
<td>Supervisor’s Mailbox</td>
<td>X + #</td>
<td>X = valid mailbox number</td>
<td>--</td>
</tr>
<tr>
<td>421</td>
<td>External Notification Port-From</td>
<td>X + #</td>
<td>X = 0 ~23</td>
<td>1</td>
</tr>
<tr>
<td>422</td>
<td>External Notification</td>
<td>X + #</td>
<td>X = 0~23</td>
<td>1</td>
</tr>
<tr>
<td>Func. Code</td>
<td>Description</td>
<td>Input format</td>
<td>Valid Range</td>
<td>Default</td>
</tr>
<tr>
<td>-----------</td>
<td>-------------</td>
<td>--------------</td>
<td>-------------</td>
<td>---------</td>
</tr>
<tr>
<td></td>
<td>Port-To</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
## Appendix G: System Capacity

The maximum system capacity of the InterPBX System is as below but might be limited to the license you purchased.

<table>
<thead>
<tr>
<th>Description</th>
<th>Blaze5000</th>
<th>Savanna8000</th>
<th>Blaze1200</th>
</tr>
</thead>
<tbody>
<tr>
<td>Extensions (incl. IP, analog, software, SIP, off-premises, virtual phone &amp; console)</td>
<td>300</td>
<td>600</td>
<td>100</td>
</tr>
<tr>
<td>CO Lines (port)</td>
<td>144</td>
<td>288</td>
<td>12</td>
</tr>
<tr>
<td>Voice Gateways (set)</td>
<td>10</td>
<td>20</td>
<td>10 (FXS only, No FXO)</td>
</tr>
<tr>
<td>Digital Gateways (set)</td>
<td>8</td>
<td>8</td>
<td>0</td>
</tr>
<tr>
<td>VMS Channels</td>
<td>24~96 (requires another VM8000 when exceeding 24)</td>
<td>24~200 (requires another VM8000 when exceeding 24; another IPC VM server when exceeding 96)</td>
<td>12</td>
</tr>
<tr>
<td>Joint Servers</td>
<td>20</td>
<td>20</td>
<td>50</td>
</tr>
<tr>
<td>Interlinks between Joint Servers</td>
<td>30</td>
<td>30</td>
<td>30</td>
</tr>
<tr>
<td>Embedded Logger (port)</td>
<td>10 (Store on Demand)</td>
<td>0 *</td>
<td>10 (Record on Demand)</td>
</tr>
<tr>
<td>Blaze Logger (port)</td>
<td>100~150 (requires high-end PC when exceeding 100)</td>
<td>100~150 (requires high-end PC when exceeding 100)</td>
<td>100~150 (requires high-end PC when exceeding 100)</td>
</tr>
<tr>
<td>BlazeLink (CTI Gateway, set)</td>
<td>1</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>Agent (set)</td>
<td>300</td>
<td>600</td>
<td>100</td>
</tr>
<tr>
<td>SoftConsole (set)</td>
<td>3</td>
<td>3</td>
<td>3</td>
</tr>
<tr>
<td>InterConsole (set)</td>
<td>10</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>SIP Proxy (set)</td>
<td>2</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>SIP Phone Register Number(set)</td>
<td>300</td>
<td>512</td>
<td>100</td>
</tr>
<tr>
<td>SIP Phone Concurrent Call</td>
<td>48</td>
<td>48</td>
<td>48</td>
</tr>
<tr>
<td>AA Menu</td>
<td>300 (000-299)</td>
<td>300 (000-299)</td>
<td>300 (000-299)</td>
</tr>
<tr>
<td>Description</td>
<td>Blaze5000</td>
<td>Savanna8000</td>
<td>Blaze1200</td>
</tr>
<tr>
<td>-----------------------------------</td>
<td>-----------</td>
<td>-------------</td>
<td>-----------</td>
</tr>
<tr>
<td>Class of Service</td>
<td>50</td>
<td>50</td>
<td>50</td>
</tr>
<tr>
<td>Extension Groups</td>
<td>100</td>
<td>100</td>
<td>100</td>
</tr>
<tr>
<td>Trunk Groups</td>
<td>50</td>
<td>50</td>
<td>50</td>
</tr>
<tr>
<td>Button Mapping Groups</td>
<td>50</td>
<td>50</td>
<td>50</td>
</tr>
<tr>
<td>Conference Sessions</td>
<td>18</td>
<td>18</td>
<td>18</td>
</tr>
<tr>
<td>Parking Slots</td>
<td>10</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>System Speed Dial Numbers</td>
<td>1000 (000-999)</td>
<td>1000 (000-999)</td>
<td>1000 (000-999)</td>
</tr>
<tr>
<td>Personal Speed Dial Numbers</td>
<td>100 (00-99)</td>
<td>100 (00-99)</td>
<td>100 (00-99)</td>
</tr>
<tr>
<td>Holidays</td>
<td>100</td>
<td>100</td>
<td>100</td>
</tr>
<tr>
<td>International Codes</td>
<td>100</td>
<td>100</td>
<td>100</td>
</tr>
<tr>
<td>Long Distance Codes</td>
<td>100</td>
<td>100</td>
<td>100</td>
</tr>
<tr>
<td>Class of Service Allow Numbers</td>
<td>20</td>
<td>20</td>
<td>20</td>
</tr>
<tr>
<td>Class of Service Disallow Numbers</td>
<td>20</td>
<td>20</td>
<td>20</td>
</tr>
<tr>
<td>Members of Extension Group</td>
<td>100</td>
<td>100</td>
<td>100</td>
</tr>
<tr>
<td>Members of Trunk Group</td>
<td>100</td>
<td>100</td>
<td>100</td>
</tr>
<tr>
<td>Paging Slots</td>
<td>10</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Calls in ACD Queue</td>
<td>99</td>
<td>99</td>
<td>99</td>
</tr>
<tr>
<td>Authorization Code</td>
<td>1000</td>
<td>1000</td>
<td>1000</td>
</tr>
<tr>
<td>ARS</td>
<td>150</td>
<td>150</td>
<td>150</td>
</tr>
<tr>
<td>AAR</td>
<td>150</td>
<td>150</td>
<td>150</td>
</tr>
<tr>
<td>Boss-Secretary Group</td>
<td>50</td>
<td>50</td>
<td>50</td>
</tr>
</tbody>
</table>

* Savanna 8000 does not provide the built-in recording function. You can purchase the Blaze Logger Recording System if you need this function.
Index

802.1p/Q ................................................. 38
AA Menu .............................................. 136
AA Menu Greetings ............................... 139
AAR ..................................................... 124
Add SIP Extensions ................................. 79
Add SIP Proxy ......................................... 74
Add SIP Trunks ....................................... 75
Adding Digital Gateway ......................... 66
Adjust Gain Level ................................... 132
Allow and Disallow Table ....................... 121
Analog Extension ................................... 97
Answer Option ...................................... 90
Apply Settings ...................................... 55, 71, 84
ARS ..................................................... 123
Authorization Code ............................... 47, 126
Authorization Code Timeout .................... 47
Auto Discovery ...................................... 145
Backup System Files .............................. 146
Billing System ....................................... 150
Boss and Secretary ............................... 127
Business Hours ..................................... 39
Button Mapping ..................................... 85
Button Mapping Group ........................... 82
Button Mapping Groups ......................... 115
Call Forward ........................................ 90, 91, 111
Call Hold ............................................. 47
Call Park .............................................. 47
Call Restriction ..................................... 42
Call Screen .......................................... 92
Call Waiting ........................................ 91
Caller ID ............................................ 54, 56
CDR ..................................................... 148
Chain Queue ....................................... 111
Check Gateway Software Version ........... 68
Check Software Version ......................... 144
Class of Service .................................... 82, 119
CO Line Extension Number .................... 53
CO Priority of Class of Service ............... 122
CO Priority of Notification ..................... 140
CODEC .............................................. 53, 82
Company Information ........................... 39
Conference Disabled ............................. 83
Configuring CO Line Ports ..................... 52
Configuring SLT Ports ........................... 55
Connecting PBX Server via Console Port 19
Connecting VG5000 via Console port ....... 25
Connecting VG5000 via Telnet ................ 24
Create SIP Trunk ARS ............................ 77
Creating Analog Gateway List .................. 50
Creating CO Line Groups ....................... 104
CTI Gateway ........................................ 72
Default Class of Service ......................... 146
Digital Line Gateways ......................... 66
Distinctive Ringing ............................... 39
Distribution List .................................... 95
dns ..................................................... 37
DTMF Programming .............................. 179
Editing Analog Gateways ....................... 51
Email Settings ....................................... 37
Enable/Disable Extension ....................... 84
Enter License Key ................................. 23, 29
Extension Group Members ..................... 109
Extension Groups ................................ 105
Extension Paging Zone ......................... 84
Extension Report .................................. 155
External Forward ................................. 134
External Forward ................................. 92
External Paging .................................... 24
External Paging & MOH Functions .......... 51
Fax .................................................... 53
<table>
<thead>
<tr>
<th>Topic</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>Store on Demand List</td>
<td>61</td>
</tr>
<tr>
<td>System Capacity</td>
<td>185</td>
</tr>
<tr>
<td>System Date/Time</td>
<td>46</td>
</tr>
<tr>
<td>System Log</td>
<td>154</td>
</tr>
<tr>
<td>System Speed Dialing</td>
<td>41</td>
</tr>
<tr>
<td>System Summary Report</td>
<td>154</td>
</tr>
<tr>
<td>T.38 Fax</td>
<td>53, 56</td>
</tr>
<tr>
<td>Tel. Programming Mode</td>
<td>140</td>
</tr>
<tr>
<td>Transfer</td>
<td>47</td>
</tr>
<tr>
<td>Transfer Announcement</td>
<td>132</td>
</tr>
<tr>
<td>Transfer Options</td>
<td>133</td>
</tr>
<tr>
<td>Transfer Options Settings</td>
<td>132</td>
</tr>
<tr>
<td>Trunk Disconnect Timer</td>
<td>120</td>
</tr>
<tr>
<td>Trunk to Trunk Transfer</td>
<td>91</td>
</tr>
<tr>
<td>Unified Messaging</td>
<td>94</td>
</tr>
<tr>
<td>Upgrade VG5000</td>
<td>23</td>
</tr>
<tr>
<td>Upgrades</td>
<td>148</td>
</tr>
<tr>
<td>Using the Recording Function</td>
<td>170</td>
</tr>
<tr>
<td>Virtual Extensions</td>
<td>95</td>
</tr>
<tr>
<td>Voice Gain Level</td>
<td>55, 56</td>
</tr>
<tr>
<td>Voice Mail Language</td>
<td>130</td>
</tr>
<tr>
<td>Voice Mail Parameters</td>
<td>130</td>
</tr>
<tr>
<td>Wrap Up Time</td>
<td>107</td>
</tr>
</tbody>
</table>