YV3 User’s Guide
Directory

1 Overview ................................................................................................................... 5
  1.1 Features .............................................................................................................. 5
  1.2 Technical Specification ..................................................................................... 6
    1.2.1 Call Control Capability ............................................................................... 6
    1.2.2 Real-Time Voice Streaming ......................................................................... 6
    1.2.3 NAT and Firewall ......................................................................................... 7
  1.3 Management .................................................................................................... 7
  1.4 Hardware ........................................................................................................... 8
  1.5 Key .................................................................................................................... 8
    1.5.1 Fixed Keys .................................................................................................. 8
    1.5.2 Alphabets and Numbers .............................................................................. 8
    1.5.3 Soft Keys .................................................................................................... 9
      1.5.3.1 Context-Sensitive Soft Keys ................................................................. 9
    1.5.3.2 User Programmable Keys ................................................................. 9
  1.5.4 Key ............................................................................................................... 10

2 Prerequisites .......................................................................................................... 11
  2.1 SIP Address-of-Record (SIP AoR) .................................................................. 11
  2.2 How to Configure Your Terminal ..................................................................... 11
    2.2.1 Network ...................................................................................................... 13
      2.2.1.1 DHCP ........................................................................................................ 13
      2.2.1.2 Static IP (Fixed IP) .................................................................................. 14
      2.2.1.3 PPPoE ........................................................................................................ 14
      2.2.1.4 Verify Network Configuration ................................................................. 14
    2.2.2 SIP Service ................................................................................................. 14
    2.2.3 Configure NAT and Firewall ...................................................................... 16
  2.3 Initialization ...................................................................................................... 16
    2.3.1 Registration ................................................................................................. 16
    2.3.2 Register On Demand ................................................................................... 17
    2.3.3 Idle ............................................................................................................... 19

3 Making Calls ......................................................................................................... 21
  3.1 Calling ............................................................................................................... 21
    3.1.1 Auto-Redial ............................................................................................... 22
  3.2 Dial Scheme ...................................................................................................... 23
    3.2.1 Guarding Time ........................................................................................... 25
    3.2.2 ENUM Sample ............................................................................................ 26
  3.3 Redial ............................................................................................................... 29
  3.4 Address Book .................................................................................................... 29
  3.5 Favorite ............................................................................................................. 33
  3.6 Call History ...................................................................................................... 37
  3.7 Speed Dial ........................................................................................................ 39
  3.8 Call Return ....................................................................................................... 42
  3.9 One-Touch Dial ............................................................................................... 42

4 Taking Calls ......................................................................................................... 44
  4.1 Incoming Call while Idle ................................................................................... 44
  4.2 Incoming Call while Talking ............................................................................ 44
  4.3 Forward and DND ............................................................................................ 46
    4.3.1 Do Not Disturb (DND) ................................................................................ 46
    4.3.2 Call Forward ................................................................................................ 46
11.1 Terminal Presence Status ................................................................. 94
11.2 IMPP Manipulation by Keypad ........................................................ 96
11.3 IMPP Manipulation on Web ............................................................... 97

12 NAT Traversal ......................................................................................... 103
12.1 Public Internet Configuration ............................................................ 103
12.2 LAN Configuration to Traverse NAT and Firewall ............................. 104
  12.2.1 Static NAT Route ................................................................. 104
  12.2.2 NAT Traversal by STUN ....................................................... 107
  12.2.3 NAT Traversal by UPnP ....................................................... 108

13 Web-Specific System Administration ..................................................... 109
13.1 Export Personal Data and Configuration ........................................... 109
13.2 System Administration ........................................................................ 110
  13.2.1 Issue Commands by HTTP Get ............................................... 111
13.3 SIP Status Code .............................................................................. 112
13.4 Firmware Upgrade ............................................................................ 113

Appendix A – Trouble Shooting ................................................................. 115
1 Overview

1.1 Features

- DHCP or PPPoE for host IP, gateway, network mask, DNS (optional, 2 DNS at most), TFTP server, NTP server, TTL and SIP server; all those settings could be static assigned as well.
- If provided by DHCP server, it could use DHCP option code to get NTP, date/time offset, TFTP and SIP server (option code 120 as per RFC3361).
- Ear-phone, speaker-phone for hand-free, handset and loud-speaker support (all are volume adjustable). Changeable ringing tone.
- Support 2 concurrent calls for call transfer (blind/semi-supervised/attended), and 3-way local conferencing.
- Multiple service domains for easy access to different ISPs (3 domains at most).
- Address book (up to 1,000 entries) and call history (20 most recently received calls, 20 most recently missed calls and 20 most recently dialed numbers).
- Call return, speed dials (20 numbers), redial, auto-redial, call-screening (100 numbers) and detail records of the latest three calls.
- Call forwarding: configurable forwarding number, unconditionally forward all calls, forward calls on busy and forward calls on no response (adjustable waiting time).
- Call processing includes: hold (music on hold), mute, Caller ID, Call waiting (alerting tone and screen popup), call transfer (blind transfer, consultative transfer, semi-supervised transfer, and take-back), call forward, call reject, do not disturb (DND).
- Call preferences include: call waiting, auto-answer (server-side invoked, locally activated and selectively auto-answer) or dial-timeout, adjustable hold recall timer, auto-hold on call switch, auto-redial criterion (stop-on-ringing or stop-on-connected), inter-digit time-out, message alerting, and per call Calling Line Identification Restriction.
- Directed Call Pickup and group call pickup.
- Join an existing conversation to have a conference among all joined parties, or ask somebody to join your undergoing conversation to have a 3-way local conference among them as per RFC3911.
- Click-to-Call via web interface.
- User downloadable music on hold (MoH)
- Multiple audio CODEC negotiation, including G.711A, G.711μ, G.729A/B, G.723.1A (5.3/6.4 kbps) and G726 (32 and 40 kbps, which G726-32kbps is identical to G721).
- Support both in-band DTMF mixed with RTP voice stream, out-of-band DTMF over RTP (RFC2833), and DTMF relay over SIP signaling channel by INFO (RFC2976).
- Message Waiting Indication (MWI)
- 50 IMPP (SIMPLE) contacts for Presence (availability from RFC3265, RFC3856, RFC3863, RFC4479, RFC4480), Busy-Lamp field and Shared-line Appearance or BLF (RFC4235).
- Hot line and 100 favorite contacts for easy access.
- 4 programmable hardware keys and 12 programmable software shortcuts for easy access to various features. (such as one-touch dial).
- 180 locally generated ring-back tone and 183 remotely generated call progress tone.
- Configurable tone cadence to fit into various country regulations.
- Out-of-dialog instant messaging, and flashing SMS without user interaction.
- NAT & firewall support by STUN or pre-configured NAT Gateway port mapping; Auto-update or notify the change of NAT IP by STUN if NAT employs DHCP as well (such as xDSL dial-up).
- Symmetric RTP flow for cases where only one endpoint is behind a NAT.
- Voice activity detection to reduce network bandwidth consumption.
- Comfort noise generation and dynamic de-jitter buffer to deliver better voice quality.
- Regular alarm and one-time alarm.
- Menu driven configuration by keypad, Web browser or TELNET.
- Use of Simple Network Time Protocol (SNTP) to synchronize time with network time server and adjust to time-zone (configurable) and daylight saving time (configurable).
- Use of Trivial File Transport Protocol (TFTP) and HTTP for auto-provisioning and image update. Support AES of 192-bit key length encryption for auto-provision.
- All configuration data can be exported as provision file, which can be imported again by a web browser or used as the auto-provision file. Exported data can be encrypted by AES of 192-bit key length.
IEEE 802.1Q VLAN tagging support
Support both 802.1P link layer precedence bits and IP layer type-of-service (ToS) bits for voice streaming, such that YV3 would perfectly replace your desktop analog phone within a switched network.
SNMPv2 for network management and supervision.
Call-related statistics, including total inbound/outbound calls, average conversation duration, connected ratio of the last 50 calls, and the conversation time distribution (less than 3 minutes, 3-20 minutes and longer than 20 minutes) during the last 72 hours, or since system startup.

1.2 Technical Specification

1.2.1 Call Control Capability
- Fully complies with RFC3261 (SIP) with RFC2543 backward compatible.
- Fully complies with RFC2327 and RFC4566 (SDP) and RFC3264 for capability negotiation based on SDP offer and answer model.
- Multiple Outbound Proxy, Registrar and Redirect server support, up to 3 different service domains.
- Support SIP server authentication procedure (HTTP digest authentication scheme)
- On-demand registration and auto-registration again on network configuration changed (auto-detect the changed IP of host, NAT, Dynamic DNS)
- Support DHCP option code 120 for SIP server (RFC 3361).
- Auto-locating SIP server (RFC3263) by DNS NAPTR/SRV record (RFC2782) lookup
- Supports SIP multicast registration to 224.0.1.75
- Support RFC3262 for reliable provisional response transmission (100rel and PRACK).
- Support RFC3323 and RFC3325 for anonymous call (“Privacy: id”, “P-Preferred-ID” and “P-Asserted-ID”).
- Message waiting indication, MWI, (RFC3842).
- Fully Implementation of RFC2916 (E.164 and DNS) for ENUM translation by NAPTR (RFC2915)
- Configurable SIP signaling port (default 5060), support both UDP and TCP.
- Support rport and received in VIA header (RFC3581) (Configurable)
- Redundancy SIP proxy server support by DNS NAPTR/SRV/A records.
- Support DTMF relay by INFO (RFC2976)
- Support Session Timers as per RFC4028.
- Support “alert-info” header for distinctive ring.
- Support directed and group call pickup as per
  - RFC3265-SIP Event Notification
  - RFC4235- INVITE Initiated Dialog Event Package
  - “draft-ietf-sipping-service-examples-10”
  - RFC4662-Event Notify for Resource List.
  - RFC2387-Multipart-Related MIME Type.
- Support RFC3911 Join header.
- Busy Lamp Field, BLF, for early (ringing)/confirmed (conversation)/terminated (idle) states (RFC4235, RFC2387, RFC4662).
- Shared-Line Appearance, SLA or SCA for Shared-Call Appearance as in “BROADWORKS SIP ACCESS SIDE EXTENSIONS INTERFACE SPECIFICATIONS” release 13.0 version 1 from BroadSoft Inc.
- Support REGISTER, INVITE, ACK, CANCEL, BYE, OPTION, REFER, SUBSCRIBE, NOTIFY, INFO, PRACK methods.
- Support both 180 locally generated ring-back tone and 183 remotely generated call progress tone.
- Configurable IP type-of-service, ToS, of signal packets.

1.2.2 Real-Time Voice Streaming
- Fully complies with RFC1889 (RTP / RTCP), RFC1890 (AVT profiles), RFC3551 (RTP Profile for Audio and
Video Conference with Minimal Control) and RFC3555 (MIME Type Registration of RTP Payload Formats).

- Support both in-band DTMF mixed with RTP voice stream and out-of-band DTMF over RTP (RFC2833).
- Dynamic RTP de-jitter buffer and lost packets concealment management.
- Speech CODEC supports: G.711 (A-law and µ-law), G.723.1/G723.1A (both 5.3 and 6.4 kbps), G729A/G.729AB and G.726 (both 32 and 40 kbps, which 32-kbps is identical to G.721). CODEC precedence is configurable to adjust to your network link speed.
- 3-way local conferencing
- Voice activity detection (VAD) and comfort noise generation (CNG).
- Voice and ringer volume control
- Real-time acoustic echo canceller.
- IP Type of Service (ToS) bits set for RTP/RTCP packet prioritization
- 802.1P precedence bits support to prioritize RTP voice frames within switched network.
- Configurable RTP / RTCP ports.

1.2.3 NAT and Firewall

- Support static NAT mapping (both NAT IP and SIP/RTP ports are configurable)
- Support Universal Plug and Play, UPnP.
- Support NAT keep alive by sending empty UDP packets to SIP registrar server to keep NAT port mapping open.
- Support Simple Traversal of UDP through NAT, (RFC3489 STUN).
- Support auto-detect (auto-update) the change of NAT IP by STUN (in case the NAT has no static IP and employ dial-up to public internet).
- Support RFC3605, Real Time Control Protocol (RTCP) attribute in SDP.
- Symmetric RTP flow for the cases where only one endpoint behind NAT.
- Support STUN server redundancy by DNS SRV/A records

1.3 Management

- TFTP and HTTP for Auto-Provision and support AES of 192-bit key length encryption for auto-provision.
- HTTP configuration by web browser
- Key-pad configuration
- TELNET configuration
- SNMPv2 for network management:
  - MIB2: RFC1213
  - Get and Set operation for internal state (Proprietary Enterprise MIB for system configuration access).
- Trap:  
  - System startup
  - System shutdown (by command/SNMP/Image upgrade)
  - SIP Registrar availability
  - Call-Channel Status.
1.4 Hardware

[Diagram of YV3 hardware components]

1.5 Key

1.5.1 Fixed Keys

8 fixed keys:

- 【】: Call Hold
- 【】: Call Transfer
- 【】: Redial the last dialed number
- 【】: Speed Dial
- 【】: Menu
- 【】: Message Waiting Indication, MWI: Access to voice mail system
- 【】: Mute
- 【】: Hands-free (switch between handset and speaker-phone/ear-phone) mode.

1.5.2 Alphabets and Numbers

【0-9 | * | #】: circular input by pressing the same key

<table>
<thead>
<tr>
<th>Key</th>
<th>Alphabet &amp; Number</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>2</td>
<td>a-&gt;b-&gt;c-&gt;A-&gt;B-&gt;C</td>
</tr>
<tr>
<td>3</td>
<td>d-&gt;e-&gt;f-&gt;D-&gt;E-&gt;F</td>
</tr>
<tr>
<td>4</td>
<td>g-&gt;h-&gt;i-&gt;G-&gt;H-&gt;I</td>
</tr>
<tr>
<td>5</td>
<td>j-&gt;k-&gt;l-&gt;J-&gt;K-&gt;L</td>
</tr>
<tr>
<td>6</td>
<td>m-&gt;n-&gt;o-&gt;M-&gt;N-&gt;O</td>
</tr>
<tr>
<td>7</td>
<td>p-&gt;q-&gt;r-&gt;s-&gt;P-&gt;Q-&gt;R-&gt;S</td>
</tr>
<tr>
<td>8</td>
<td>t-&gt;u-&gt;v-&gt;T-&gt;U-&gt;V</td>
</tr>
</tbody>
</table>
1.5.3 Soft Keys

1.5.3.1 Context-Sensitive Soft Keys

Under certain circumstances, when more than 4 soft buttons are needed, user may press navigation keys 【∧】 and 【⋀】 to scroll to next page.

For example:
When trying to show Func-1 ~ Func-7
At first, the soft key panel is shown as:

```
Func-1  Func-2  Func-3  Func-4
```

If 【⋀】 is pressed once, the soft key bar changes to:

```
Func-5  Func-6  Func-7
```

If 【∧】 is pressed, the soft key bar changes back to:

```
Func-1  Func-2  Func-3  Func-4
```

1.5.3.2 User Programmable Keys

16 user-defined programmable keys (ID-5 to ID-16) will be shown during call state whenever [DSS] is pressed excluding those NO-Function keys. The very first 4 programmable keys, from ID-1 to ID-4, go to the 4 soft-keys on the top half of the LCD Panel.

- **1. Favorite**
- **2. IMPP**
- **3. AddrBook**
- **4. SIP Realm**

<table>
<thead>
<tr>
<th>1. Favorite</th>
<th>2. IMPP</th>
<th>3. AddrBook</th>
<th>4. SIP Realm</th>
</tr>
</thead>
<tbody>
<tr>
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</tr>
</tbody>
</table>

a. Show ‘✓’ to indicate activation status of “Auto-Answer”, “DND”, “DTMF Relay by INFO”; and unmap of these features will turn off selected feature as well.
b. *Shoe ‘✓’ to indicate successfully registered to all active SIP service domains and a ‘X’ symbol to indicate none of the active domains are registered. No symbols shown if register to any of the active SIP service domains succeeded.*
c. Show display or user-part of its email-like address whenever possible for One-Touch Dial.
d. Show Show for “One-Touch Dial”.
e. Otherwise prompt to selected menu for further interaction.
f. Please refer to section 3.2 “Programmable Keys” on “YV3 Administration” for detail

The other 12 user-defined keys (not NO-Function) will be shown during call state whenever...
[DSS] is pressed.

<table>
<thead>
<tr>
<th>1. Call History</th>
<th>2. My Presence Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>3. Call Return</td>
<td>4. Do Not Disturb (DND)</td>
</tr>
<tr>
<td>7. Regi ster</td>
<td>8. DTMF List</td>
</tr>
<tr>
<td>9. Auto-Answer</td>
<td>10. DTMF Relay by INFO</td>
</tr>
<tr>
<td>11. Reject</td>
<td>12. Forward</td>
</tr>
</tbody>
</table>

- [Back]: return.
- 【√】: Activate selected feature
- 【 ▼】 and 【 ▲】: navigate through list.
  a. Show ‘√’ to indicate activation status of “Auto-Answer”, “DND”, “DTMF Relay by INFO”; and unmap of these features will turn off selected feature as well.
  b. Show ‘√’ to indicate successfully registered to all active SIP service domains and a ‘X’ symbol to indicate none of the active domains are registered. No symbols shown if register to any of the active SIP service domains succeeded.
  c. Show display or user-part of its email-like address whenever possible for One-Touch Dial. Besides, show Show ☑ for “One-Touch Dial”.
  d. Otherwise prompt to selected menu for further interaction.
  e. Please refer to section 3.2 “Programmable Keys” on “YV3 Administration” for detail.
2 Prerequisites

2.1 SIP Address-of-Record (SIP AoR)

The general form of SIP address-of-record is:

```
“Display” <protocol:email-like-address>;tag=param
```

- The “Display” field is optional. If present, it consists of any ASCII characters except for ‘<’ and ‘>’. If the “Display” is present, the following address must be enclosed in a paired ‘<’ and ‘>’.
- Protocol: Usually in lower case, such as “sip”, “tel” or “sips”. Note, “sip”, “tel” and “sips” protocol names MUST be specified in lower case, which is stipulated on RFC3261.
- email-like-address: in the form of “user-part@domain” where user-part is optional and the domain part could be either a dotted IP or a domain name record, such as:
  - 3200@SIP.isp.com
  - mike@192.168.192.100
  - 192.168.3.100 (Note, the user-part is optional in direct IP dialing mode)
  - +886-3-5639025
- “tag=param”: Multiple parameters could be present (separated by ‘;’)

**Example**

- Michael <sip:Michael@SIP.isp.com>
- Mike Jackson <sip:3200@SIP.isp.com>
- “Voice Mailbox” <sip:8888@vms.SIP.isp.com> Display with enclosing ‘”’.
- sip:300@SIP.isp.com AoR without display
- sip:192.168.3.100 AoR without user part (in dotted IP)
- tel:+886-3-5639025 ENUM AoR
- sip:+88635639025@SIP.isp.com;user=phone ENUM AoR with SIP proxy support

2.2 How to Configure Your Terminal

Initially, your phone can only be configured via keypad since it bears no valid IP yet. Below list the factory default values on terminal shipping:

<table>
<thead>
<tr>
<th>Settings</th>
<th>Factory Default Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Network Mode</td>
<td>DHCP for IP/Netmask/Gateway/DNS, etc.</td>
</tr>
<tr>
<td>IP</td>
<td>192.168.192.168</td>
</tr>
<tr>
<td>Network Mask</td>
<td>255.255.255.0</td>
</tr>
<tr>
<td>Gateway</td>
<td>0.0.0.0</td>
</tr>
<tr>
<td>DNS IP</td>
<td>0.0.0.0</td>
</tr>
<tr>
<td>Privileged Password</td>
<td>admin/0000</td>
</tr>
<tr>
<td>User-level password</td>
<td>user/0000</td>
</tr>
<tr>
<td>Hardware Reset Password</td>
<td>4273927373738# (“HardwareReset#”)</td>
</tr>
<tr>
<td>Show Terminal’s IP on LCD</td>
<td>#*47 (“#*IP”)</td>
</tr>
</tbody>
</table>

**Note**, if you have ever forgotten your terminal password (privileged and user-level), you may enter hardware reset password, “4273927373738#” (HardwareReset#) to restore everything back to shipping values then auto-reboot. Whenever this string “4273927373738#” is entered as dial-string or as password to unlock the terminal, it will trigger a hardware reset. After properly boot up, the default passwords to both user-level and privileged-level are reset to “0000”. This, however, will clear all settings, including personal information, such as address book, call history, instant messages, etc. Moreover, such hardware reset password can only be entered via keypad applicable in neither telnet nor HTTP web page) for security reasons.

After finishing configuring your network, you could use either a web browser (HTTP port 80) or a TELNET client (TCP port 23) if you have a small number phone-sets to configure (you may call to “#*47” (“#*IP”) to show terminal’s IP on LCD).

However, we recommend you to use TFTP or HTTP for auto-provision if you have to administer large amounts of phone-sets. For TFTP/HTTP auto-provision, please refer to “Auto-Provision on YV3 Administration”.

Basically you have the following ways to configure your YV3.
• Press 【X】to activate the configuration menu and configure it via keypad.

|-------------------|-----------------|----------------------|---------------|--------------|---------|----------------|---------|------------|

- 【X】: Enter or exit menu mode.
- 【∧】 and 【∨】: Navigate through menu items by up and down navigation keys.
- 【↲】: Traverse into selected menu or return to previous menu if no specified function.
- [Back]: return.
- 4 soft-buttons on the bottom of LCD: Context-sensitive buttons.

• Use any modern web browser to configure the phone from a PC (you may call to “#*47” (“#*IP”) to show terminal’s IP on LCD). The default login account/password for administrative account is “admin/0000” and “user/0000” for user-level account.

Enter Network Password

This secure Web Site (at 192.168.3.95) requires you to log on.

Please type the User Name and Password that you use for YV3.

- User Name: admin
- Password:

Save this password in your password list

[OK] [Cancel]

After logging in, you will get into the YV3 status page as below:
TELNET into the phone by any TELNET client (you may call to “#*47” (“#*IP”) to show terminal’s IP on LCD). The default TELNET port is TCP port 23, login password is as the same as your phone password set in “3.Terminal Settings” / “Password” on “YV3 Administration” (the default phone password is “0000”) and the max concurrency is 4. YV3 will not apply the changes until user presses [Ctrl] + ’s’ to apply the modifications or the client disconnected by [Ctrl] + ’c’.

Auto-provision on phone startup. Please refer to section-“Auto-provision” on “YV3 Administration”.

Note, before you can configure your phone-set from your PC, such as by a TELNET client or point your browser to the phone-set, you must have configured its IP via keypad properly.

2.2.1 Network

To configure your network:

- Press 【6.Network】
- Go to 【6.Network】 \ 【General】

Please configure the phone based on your network configuration: DHCP, static IP or PPPoE.

2.2.1.1 DHCP

- Pick 【1.Mode】 \ 【1.DHCP】
- Disable 【DNS Server】 / 【Use Static DNS】 by choosing 【DHCP】

Note: if you want to assign a different domain name server instead of using those obtained by DHCP, you should choose 【1.Static DNS】 and set the IP of your specific DNS into 【DNS Server】 , such as “1.Primary DNS” = 192.168.3.254 (modify this as necessary)

The supported DHCP options are:

- Client PC address
2.2.1.2 Static IP (Fixed IP)

- Pick 【1.Mode】 \ 【Static assign】
- Go to 【2.Static Settings】 , and enter your network configurations based on your ISP. For example:
  1. Host IP = 210.201.210.132 (modify this as necessary)
  2. Network mask = 255.255.255.0 (modify this as necessary)
  3. Gateway IP = 210.201.210.128 (modify this as necessary)
- Enable 【DNS Server】 / 【Use static DNS】 by choosing 【Static DNS】
- Assign 【DNS Server】 , such as: 【1.Primary DNS】 = "168.95.1.1" (modify this as necessary)

2.2.1.3 PPPoE

- Pick 【1.Mode】 \ 【3.PPPoE】
- Go to 【3.PPPoE settings】 and enter your PPPoE authentication information, such as:
  1. Login ID = MyPPPoEAccount (modify this as necessary)
  2. Password = PPPoEDialupPassword (modify this as necessary)
  3. Service Name = Optional, some ISP requires it (modify this as necessary).

2.2.1.4 Verify Network Configuration

You may call to “#*47” (“#*IP”) to show terminal’s IP on LCD; alternatively, you can Press 【(*)】 and goes to 【9.Advanced】 \ 【System Status】 \ 【Network】 to check current active network settings:

| IP: 192.168.3.101 |
| MAC:000DC3112233 |
| DNS:192.168.3.253 |

It will display the host IP, Ethernet MAC address and the active DNS IP (secondary DNS IP will be shown if available) in order.

Once finishing network configuration, you should be able to place a point-to-point call. For example, if your phone IP is "192.168.1.10” and you want to dial another SIP phone which IP is “192.168.1.20”, please dial "*20**5060" (or just "*20" if the target phone listens on UDP port 5060; otherwise you must dial the target UDP port as well). This is “LAN dialing” (Refer to section 8.1- "Dialing Scheme" on this document). If the call could be set up correctly, then your network configuration is fine; otherwise, please refer to B-1 on Appendix B- "Trouble Shooting".

Note, if you reside on a LAN without gateway, you should specify the gateway IP as “0.0.0.0” rather than assigning a non-existent or an invalid IP; otherwise the network packets may not be routed correctly (which may result in no voice packets could be sent from this phone)! This constrain applies to DHCP and PPPoE as well: DHCP and PPPoE server should not designate a non-existent or invalid gateway.

2.2.2 SIP Service

Before you start, you should ensure you have SIP-related data from your ISP. For example, if you get the following information from your SIP ISP:

i. Account: Michael
ii. Password: secret
iii. SIP address-of-record: 8888@isp.com
iv. SIP Proxy / Registrar Server: sip.isp.com, which serves on UDP port 5060

Then you could configure your YV3 by a web browser on 【YV3】 / 【SIP Settings】 / 【N-th Domain】
Alternatively, you may go to 【 】=>”7.SIP settings” / “1st Realm” to configure these information by keypad.

i. If you have applied for more than one service domains, please repeat step I and II until all active domains are properly configured. YV3 supports three different service domains at most.

ii. After saving the configuration, the system will try to register to those activated SIP domains. You may press [DSS] soft-button and execute “SIP Domain Status” to check registration results.

Show a symbol to indicate:

a. ‘√’ to indicate successfully registered to all active SIP service domains.
b. ‘x’ to indicate none of the active domains are registered.
c. No symbol to indicate registered to any of the active SIP service domain succeeded.

If pressed (activated), it will show registration status of each active SIP service domain.

Show a symbol preceding each active SIP service domain:

a. ‘√’ to indicate a register success state
b. ‘x’ to indicate a register failed state

c. ‘.’ To indicate a registering state.

YV3 supports up to 3 SIP service domains, which this terminal may register to; and you may circle
active service domains by pressing [Realm] soft-button while making outbound calls. Please refer to section-10.11 “Registration on Demand” and “Multi-domain Registration” on this user’s guide for detail.

iii. If you failed to register your phone to SIP registrar, please refer to Appendix A-“Trouble Shooting” on this document.

2.2.3 Configure NAT and Firewall

If your SIP server locates on public internet, whereas your phone resides on a local area network, please refer to -“NAT Traversal” on this user’s guide if your phone-set is behind network address translator (NAT) and / or firewall.

2.3 Initialization

(a) Startup:

YV3
Version

(b) Check for auto-provision. Please refer to “Auto-Provision” on "YV3 Administration" for detail.

(c) Check Date and Time:

 Specify current date and time on this terminal. Enter the current date, where the time is in 24-hour format and the date format depends on the “3.Date/Time” / “FMT” configuration.

- [Save]: Save changes and return.
- [Back]: Return without changes.
- [<] and [>]: Navigate through fields.
- [0-9]: Enter values.

Note:

- YV3 will go the idle (ready) mode after 5 seconds if user dose not enter any digit. The default system time on start-up is January 1, 2007, 00:00, GMT.

2. The phone will synchronize its time by Simple Network Time Protocol, SNTP, with network time server regularly if SNTP is enabled. If you want to keep the time you manually set previously, you must disable SNTP. Please refer to section-“Date/Time” on “YV3 Administration” for detail.

- You can just ignore the date & time settings on boot, leaving the phone to synchronize its clock with network time server. Please check menu-“Time zone” (by TELNET or keypad) to adjust your time zone otherwise the synchronized time may be several hours late (earlier) than your local time. Please refer to “Date/Time” on this document for detail.

2.3.1 Registration

(a) Registering

Register to
Registrar.ISP.com

The registered on-line timeout is 3600 seconds (1 hour).

(b) Registration Done

Registered
Expired in 3600 seconds

System will refresh registration after 50% of the expiration interval elapsed.

(c) Registration Failed

Register Failed
Unauthorized

This message will freeze the screen for a while, such as 5 seconds, for the user to figure out the reason. Failed registration will shorten the re-registration interval to 90 seconds.

2.3.2 Register On Demand

YV3 supports up to 3 SIP service domains, which this terminal may register to; and you may circle active service domains by pressing [Realm] soft-button while making outbound calls.

You can dictate the terminal to refresh registration immediately to all service domains by invoking programmable feature from context-sensitive button [DSS] => “Register” command.

If you have gone offline (see-below), by executing “Register” command, it will restart regular auto-registration scheduling as necessary too.

You can check the register results by invoking [DSS] => “SIP Domain Status” (acronym: “SIP Realm” if mapped as hardware programmable keys on top-half of LCD):

If pressed (activated), it will show registration status of each active SIP service domain.

1. Favorite
2. IMPP
3. AddrBook
4. SIP Realm

Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:

[A/B] + Indicator( ) + Caller-ID

Status Indicator

User can press [ ] and [ ] navigation key to scroll soft keys on the bottom, where [ ] / [ ] indicates that there are more soft keys, user may use [ ] and [ ] navigation key to scroll to [DSS] context-sensitive soft-button

[DSS]

1. ✓ Auto Answer
2. ✓ DTMF Relay by INFO
3. DND
4. Register
5. SIP Domain Status
6. Call Detail
7. Network Info

Back

1. ‘✓’ to indicate successfully registered to all active SIP service domains.
2. ‘x’ to indicate none of the active domains are registered.
3. No symbol to indicate registered to any of the active SIP service domain succeeded.
Show symbol preceding each active SIP service domain:

a. ‘√’ to indicate a register success state
b. ‘x’ to indicate a register failed state
c. ‘.’ To indicate a registering state.

Alternatively, you may point a web browser to your terminal and check the register status of each domain after signing in (default login account/password is “admin/0000” or “user/0000”, excluding double quotes):

I. To can go off-line explicitly by:

- From keypad (TELNET): going to 【 】/【 9.Advanced】/【System Admin】 and pick 【Logout】.
- From web browser:
  - Go to 『YV3』/ 『Advanced』/ 『System Admin』 => “Un-REGISTER (OffLine)” to go offline.
Alternatively you may issue the following HTTP Get command from your web browser to go online:

http://terminal_ip_address/unregister

Where terminal_ip_address is the IP address of your terminal. Those command web pages are password protected.

Once off-line, the terminal will unregister to all activated SIP service domains and cease regular auto-registration scheduling until the [DSS] = “Register” command is explicitly executed again.

*Note:* reboot the terminal will clear this status and register the SIP address-of-record after boot-up.

II. To can go on-line explicitly by:

- From keypad: execute “Register” command explicitly by pressing [DSS] context-sensitive soft-button.
- From web browser:
  - Point the web browser to your terminal and go to 「YV3」 / 「Advanced」 / 「System Admin」
  - 「YV3」 / 「Advanced」 / 「System Admin」
  - “Re-REGISTER” to go online again.
  - Alternatively you may issue the following HTTP Get command from your web browser to go online:

    http://terminal_ip_address/register

    Where terminal_ip_address is the IP address of your terminal. Those command web pages are password protected.

You could view registration status of each service domain by executing “SIP Domain Status” (which is “SIP Realm” if mapped to hardware programmable keys located on upper-half of the LCD).

### 2.3.3 Idle

1. Favorite

2. IMPP

3. AddrBook

4. SIP Realm

Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:

[A/B] + Indicator( principalTable, isIncoming, inSpeaker, onSpeaker) + Caller-ID
User can press 【∧】 and 【∨】 navigation key to scroll soft keys on the bottom, where 【︽】/【︾】 indicates that there are more soft keys, user may use 【∧】 and 【∨】 navigation key to scroll.

User may configure the 4 programmable keys on top of the panel; however they are not scrollable.

[A] – Channel A

<table>
<thead>
<tr>
<th>Icon</th>
<th>Type</th>
<th>Example</th>
</tr>
</thead>
<tbody>
<tr>
<td>☰</td>
<td>Call Waiting</td>
<td>[接听A]</td>
</tr>
<tr>
<td>☰</td>
<td>Holding Call</td>
<td>[接听A]</td>
</tr>
<tr>
<td>☰</td>
<td>Conference (talking)</td>
<td>[接听A]</td>
</tr>
<tr>
<td>☰</td>
<td>Auto-Redialing</td>
<td>[接听A]</td>
</tr>
</tbody>
</table>

[B] – Channel B

<table>
<thead>
<tr>
<th>Icon</th>
<th>Type</th>
<th>Example</th>
</tr>
</thead>
<tbody>
<tr>
<td>☰</td>
<td>Call Waiting</td>
<td>[接听B]</td>
</tr>
<tr>
<td>☰</td>
<td>Holding Call</td>
<td>[接听B]</td>
</tr>
<tr>
<td>☰</td>
<td>Conference (talking)</td>
<td>[接听B]</td>
</tr>
<tr>
<td>☰</td>
<td>Auto-Redialing</td>
<td>[接听B]</td>
</tr>
</tbody>
</table>

[AB] – Address Book

[CID] – Call History; and it will directly enter [Missed Call] list if there are any unread missed calls.

<table>
<thead>
<tr>
<th>Icon</th>
<th>Type</th>
<th>Example</th>
</tr>
</thead>
<tbody>
<tr>
<td>☰</td>
<td>Call Waiting</td>
<td>8888</td>
</tr>
<tr>
<td>☰</td>
<td>Holding Call</td>
<td>Jason</td>
</tr>
<tr>
<td>☰</td>
<td>Conference (talking)</td>
<td>192.168.3.4</td>
</tr>
<tr>
<td>☰</td>
<td>Auto-Redialing</td>
<td>Richard</td>
</tr>
</tbody>
</table>

[DSS]: User defined programmable keys (ID-1 to ID-4 go to the 4 softkeys on the upper half of LCD; whereas others, from ID-5 to ID-12 will be listed on pressing this key).

Screen Pop-up for Call-waiting, holding calls, conference or auto-redial: 【A/B】 + Indicator + Caller-ID.

<table>
<thead>
<tr>
<th>Icon</th>
<th>Type</th>
<th>Example</th>
</tr>
</thead>
<tbody>
<tr>
<td>☰</td>
<td>Call Waiting</td>
<td>[接听]</td>
</tr>
<tr>
<td>☰</td>
<td>Holding Call</td>
<td>[接听]</td>
</tr>
<tr>
<td>☰</td>
<td>Conference (talking)</td>
<td>[接听]</td>
</tr>
<tr>
<td>☰</td>
<td>Auto-Redialing</td>
<td>[接听]</td>
</tr>
</tbody>
</table>

Phone status indicator (in precedence order):

<table>
<thead>
<tr>
<th>Icon</th>
<th>Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>☰</td>
<td>New Messages</td>
</tr>
<tr>
<td>☰</td>
<td>Do Not Disturb, DND</td>
</tr>
<tr>
<td>☰</td>
<td>New Missed Calls</td>
</tr>
</tbody>
</table>

My Presence Status

<table>
<thead>
<tr>
<th>Icon</th>
<th>Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>☰</td>
<td>Off-Line</td>
</tr>
<tr>
<td>☰</td>
<td>Busy / On-the-Phone</td>
</tr>
<tr>
<td>☰</td>
<td>Away / Be-Right-Back / Out-to-Lunch</td>
</tr>
<tr>
<td>☰</td>
<td>Auto-Answer</td>
</tr>
<tr>
<td>☰</td>
<td>All-Call-Forward;</td>
</tr>
<tr>
<td></td>
<td>and LCD show “FWD:TargetNo”</td>
</tr>
<tr>
<td>☰</td>
<td>Phone Locked;</td>
</tr>
<tr>
<td></td>
<td>and LCD will prompt for password on locked state</td>
</tr>
</tbody>
</table>
3 Making Calls

* IMPP
* DTMF
* Favorite
* Register

B J a s o n < 8 8 8 8 @ I S P . c o m

A p r i v a t e . I S P . n e t

7 7 5 0

Call B Realm Del ⬇

User can press 【▲】and 【▼】navigation key to scroll soft keys on the bottom, where 【︽】/【︾】indicates that there are more soft keys, user may use 【▲】and 【▼】navigation key to scroll.

- [Call]- Dial out collected digits. Alternatively, You may lift the handset (offhook) or turn speaker on by pressing 【】 to dial out collected digits if you have not done so (this is referred as “Preset Dial”). Besides, the phone will dial out the collected number after 4 seconds (the inter-digit timed-out is programmable) if you have hooked off or turns speaker on; otherwise, it will restore to idle mode after 30-second in “Preset Dial” mode.
- [B] – Line switch to another (holding) channel
- [Realm] – Switch service domain.
  The serving domain will appear on the upper-right corner. Press [Realm] to circle the available service domains. With YV3, you could configure up to 3 different service domains and the default service domain of line-N is service-domain-N. For example, the default service domain for channel [B] is the 2nd service domain. If no corresponding service domain is available, it will use the 1st service domain by default.

<table>
<thead>
<tr>
<th>[A]</th>
<th>Private.ISP.net</th>
</tr>
</thead>
<tbody>
<tr>
<td>5201314</td>
<td></td>
</tr>
</tbody>
</table>

- [Del] – Delete one input character
- [123…]/[abc…] – Input method
- [AB] – Address Book
- [CID] – Call History
- [DSS]: User defined programmable keys (ID-1 to ID-4 go to the 4 softkeys on the upper half of LCD; whereas others, from ID-5 to ID-12 will be listed on pressing this key).

3.1 Calling

Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:

[A/B] + Indicator( htonl, htonl, htonl ) + Caller-ID

T r y i n g

7 7 5 0 @ I S P . c o m

B Auto DSS

- [B] – Line switch to another channel.
- [Auto] – Toggle-switch to enable/disable auto-redial (see below “Auto-Redial”).
- [DSS]: User defined programmable keys (ID-1 to ID-4 go to the 4 softkeys on the upper half of LCD; whereas others, from ID-5 to ID-12 will be listed on pressing this key).
3.1.1 Auto-Redial

You can press [AutoRedial] after finishing dialing while making a phone call but before connected or hanging up. After activating auto-redial, the system will launch the auto-redial process and re-dial the target number regularly ‘till “connected” (see below). To manually cancel auto-redial, press [Stop].

<table>
<thead>
<tr>
<th>IMPP</th>
<th>Favorite</th>
</tr>
</thead>
<tbody>
<tr>
<td>DTMF</td>
<td>Register</td>
</tr>
</tbody>
</table>

Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:

[A/B] + Indicator(أخلاق) + Caller-ID

Trying
7750@ISP.com

- [B] – Line switch to another channel.
- [Stop] – Cancel Auto-Redial
- [DSS]: User defined programmable keys (ID-1 to ID-4 go to the 4 softkeys on the upper half of LCD; whereas others, from ID-5 to ID-16 will be listed on pressing this key).

Once Auto-Redial is in progress, you may put handset back to on-hook and turn speaker off (similar to put a call on hold), and the system will keep auto-redialing target number ‘till total duration timer expired at the background in idle mode.

<table>
<thead>
<tr>
<th>IMPP</th>
<th>Favorite</th>
</tr>
</thead>
<tbody>
<tr>
<td>DTMF</td>
<td>Register</td>
</tr>
</tbody>
</table>

Once “connected”, the phone will ring back to alert user then auto-answer it.

This feature will be automatically canceled if either the stop criterion is met or the total activation duration is expired.

The semantics of “connected” could be configured from 『 Preferences 』 / 『 Auto-Redial 』 / 『 Stop Criterion 』

1. Ringing
2. Connected

- Ringing: only when the peer starts ringing back will auto-redial process stops (Default).
- Connected: Only when the peer picks up will auto-redial process stops.

Note: active auto-redial will be canceled whenever the phone is locked out.

To specify the gap (measured in seconds) between two successive re-dials to avoid overflowing the networks with fast retries, please go to 『 Preferences 』 / 『 Auto-Redial 』 / 『 Retry Interval 』. The default redial interval is 15 seconds.

To specify the activation duration of this auto-redial feature once starts (measured in seconds), please go to 『 Preferences 』 / 『 Auto-Redial 』 / 『 Total Duration 』. Once this duration expires, the auto-redial feature will be silently canceled. System default is 1800 seconds (30 minutes).

You may also configure it from web browser by going to 『 Preferences 』 => “Auto-Redial” page.
3.2 Dial Scheme

<table>
<thead>
<tr>
<th>Method</th>
<th>Rule</th>
<th>Example</th>
</tr>
</thead>
</table>
| Pick from address book | 1. Enter address book.  
2. Search for entry.  
3. Press [Call]  
Note: The phone will not use the domain you specified if the domain part of the dialing AoR is different from the one you specified and “Auto-locate” as picked domain, based on the domain part in AoR. | 1. sip:albert@SIP.isp.com  
2. <sip:felix@SIP.isp.com:3000>  
3. Scott@sip:3100@SIP.isp.com>  
4. felix@SIP.isp.com, which is equal to sip:felix@SIP.isp.com  
5. tel:+886-3-5639025 +88635639025, which is equal to “tel:+886-3-5639025” |
| ENUM dialing¹      | 1. Starts with a [#], which will be translated as ‘+’ (if you enter via keypad).  
2. Dial numbers including country code and area code  
3. Press [Call] to confirm.  
4. Valid ENUM dial strings must be longer than 6 (configurable) and containing only digits, optional ‘-‘, spaces, ‘(‘ or ‘)’, such as “#886-3 5639025”, “+86 (3) 5639025” or “#8863”.  
5. Those not recognized as valid | To dial “tel:+886-3-5639025”, please enter “#88635639025#” and it will show on the LCD as:  

```
Call  
+1-1234-5678
```

¹ ENUM capable DNS must be configured and set as the primary DNS of YV3.
<table>
<thead>
<tr>
<th>Dialing Method</th>
<th>Description</th>
<th>Example 1</th>
<th>Example 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>ENUM dialing</td>
<td>The dial string will be dialed “as is” even they start with a ‘#’. For example, “#86” will be dialed as “#<a href="mailto:86@sipdomain.com">86@sipdomain.com</a>”.</td>
<td>If the caller is <a href="mailto:3100@SIP.isp.com">3100@SIP.isp.com</a>, he could call <a href="mailto:3200@SIP.isp.com">3200@SIP.isp.com</a> by dialing “3200”.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Intra-domain Dialing (Both Caller’s &amp; callee’s SIP address-of-record must be the same domain and both have been registered).</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>1. The serving domain will appear on the upper-right corner. Press [Realm] to change the service domain.</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>2. Press [Call] to confirm.</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Contact Dialing $^2$</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>1. Use ‘*’ as ‘@’.</td>
<td>1. Call 3200@192.168.10.200</td>
<td></td>
</tr>
<tr>
<td></td>
<td>2. Use ‘*’ as dot, ‘.’.</td>
<td>3200@192.168.10.200</td>
<td></td>
</tr>
<tr>
<td></td>
<td>3. Use “***” as ‘:’ then follows peer’s SIP UDP port (optional, but must assigned whenever the target agent does not listen on SIP UDP-5060).</td>
<td>2. Call 3200@192.168.10.200:5070</td>
<td></td>
</tr>
<tr>
<td></td>
<td>4. Press [Call] to confirm.</td>
<td>3200@192.168.10.200:5070</td>
<td></td>
</tr>
<tr>
<td></td>
<td>IP Dialing $^3$ (Anonymous Call)</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>1. Use ‘*’ as dot, ‘.’.</td>
<td>1. Call 192.168.10.200</td>
<td></td>
</tr>
<tr>
<td></td>
<td>2. Use “***” as ‘:’ then follows peer’s SIP UDP port (optional, but must assigned whenever the target agent does not listen on SIP UDP-5060).</td>
<td>192.168.10.200</td>
<td></td>
</tr>
<tr>
<td></td>
<td>3. Press [Call] to confirm.</td>
<td>2. Call 192.168.10.200:5070</td>
<td></td>
</tr>
<tr>
<td></td>
<td>LAN Dialing (Caller &amp; Callee must be on the same LAN) $^4$</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>1. Starts with a ‘*’.</td>
<td>If the caller’s contact is 3100@192.168.10.100 and the callee’s IP is 192.168.10.200, dialing “200” will be treated as “192.168.10.200” in IP Dialing.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>2. Dial the last 1/2/3 field(s) of the callee’s IP.</td>
<td>200</td>
<td></td>
</tr>
<tr>
<td></td>
<td>3. Then follows SIP signaling peer’s SIP UDP port (optional, must assigned whenever the target agent does not listen on SIP UDP-5060); use ‘***’ as ‘:’.</td>
<td>ISP.com</td>
<td></td>
</tr>
<tr>
<td></td>
<td>4. “*0” is reserved for server feature access code and will be transmitted as is (no translation), any digits larger than 255 will be dialed “as is” as well (such as “*311”).</td>
<td>192.168.10.200:5070</td>
<td></td>
</tr>
<tr>
<td></td>
<td>5. “*050” is as the same as “*50”.</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>6. You may disable LAN dialing overly if those numbers should send “as is” by disabling this feature from menu</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

$^2$ If the caller does not use default outbound proxy but the callee does, then the conversation may not be hanged up properly if the callee hangs up first (but the other direction could). This is due to the nature of stateful proxy to reply a 481 “Call/Transaction doesn’t exist” response for those calls whose INVITE messages not flowing through the default outbound proxy.

$^3$ If the caller configures to use default outbound proxy, the proxy may reply “404 Not found” due to the missing user part.

$^4$ The same constrains of “IP Dialing” applies to “LAN Dialing” as well. LAN dialing in fact is a short-hand of IP-Dialing.
To facilitate “Contact Dialing”, “IP Dialing” and “LAN dialing” (where most users forget to dial the SIP signaling port of the peer, and end in no responses if the peer doesn’t listen on the standard UDP port 5060 for SIP signaling), YV3 always listens on UDP-5060 for SIP signaling in addition to the user configured SIP service port. However, if UDP-5060 is overlapped with RTP ports for media session, it will not listen on UDP-5060 for SIP signaling to avoid conflict.

Multi-domain note:
A. If the dialed AoR has no user part, it will use auto-locating (ignore the user specified domain) to facilitate IP-dialing.
B. If the dialed AoR has no domain part (such as intra-domain dialing or the number in address book specify no domain), it will use the domain you specified. Besides, if you pick “Auto-locate”, then the default domain of the current channel (A / B call) will be used.
C. If the dialed AoR has both user part and domain part, and you don’t specify “Auto-locate”, then:
   i. If the domain part of dialing AoR could match one of those registered domains, it will use the matched domain instead.
   ii. If the domain part of dialing AoR matches none of those registered domains
      1. If the domain part is in dotted IP format, it will use the user specified domain.
      2. Otherwise, use “Auto-locate”.
D. For all others, it will use the domain you specified while making calls.

3.2.1 Guarding Time
User hooks up already:
◆ The first digit timeout is 15 seconds and it will play network fail tone on expiry. Besides, the default inter-digit timeout is 4 seconds and it will dial out the collected digits on expiry as well. To speed up the dialing process, press [Call] key whenever finishing dialing
User does not hook up yet (Preset Dial):
◆ You may lift the handset (offhook) or turn speaker on by pressing 【①】to dial out collected digits; otherwise, it will restore to idle mode after 30-second in “Preset Dial” mode.
◆ Alternatively, you may press [Call] soft-button to dial out collected digits.
If the call is made by speaker phone, it will return to IDLE state after call finished for 3 seconds.

The inter-digit timeout is configurable:
To change it by TELNET or keypad, go to 【③】=> “5. Preferences” / “Dial Plan” / “Inter-Digit Timeout”.

<table>
<thead>
<tr>
<th>I n t e r - D i g i t T i m e o u t</th>
</tr>
</thead>
<tbody>
<tr>
<td>[ 3 - 9 ] ( s e c o n d )</td>
</tr>
</tbody>
</table>

4

• [Del]: Delete one character
• [Back]: Return without any changes
• 【↓】: Save changes and return.
• 【<】 and 【>】: move cursor one position in edit mode.

To configure it by web browser, go to 【Preferences】 => “Inter-Digit Timeout (s)”
To speed up the dialing process, press [Call] soft-key whenever finishing dialing.

### 3.2.2 ENUM Sample

1. Configure your YV3 to use a NAPTR capable DNS server. If you use auto-locating SIP server instead of default outbound proxy for next hop delivery, your DNS must have SRV and A records like the following (please change those host and IP in red font accordingly):

   ```
   $ORIGIN SIP.isp.com
   ;; Pref Weight Port Target
   _sip_udp.SIP.isp.com. IN SRV 0 0 1000 proxy.sip.SIP.isp.com.
   _sip._tcp.SIP.isp.com IN SRV 10 0 1000 proxy.sip.SIP.isp.com.
   ```

2. Suppose you want to register your mobile number, +886-939342017 as an ENUM number associated with your SIP AoR, sip:Michael@SIP.isp.com. Moreover, we assume that you configure your SIP phone to use the standard E.164 suffix for ENUM, “e164.arpa”, then you should add a NAPTR record under the domain “7.1.0.2.4.3.9.3.9.6.8.8.e164.arpa” (which is the reverse of your mobile phone number appended with “e164.arpa” suffix) like the following:

   ```
   $ORIGIN 7.1.0.2.4.3.9.3.9.6.8.8.e164.arpa
   ;; order pref flags service regexp replacement
   IN NAPTR 100 10 "u" "sip+E2U" "!^.*$! sip:Michael@SIP.isp.com!".
   ```

3. The E.164 suffix for ENUM is configurable from menu-“ENUM & E.164”. Alternatively, you may go to [YV3] / [SIP Settings] page by web browser and configure “ENUM & E.164”.

```
Below illustrates how to configure them by keypad (TELNET): 

1. ENUM suffix for DNS query:
   
<table>
<thead>
<tr>
<th>E</th>
<th>N</th>
<th>U</th>
<th>M</th>
<th>D</th>
<th>N</th>
<th>S</th>
<th>S</th>
<th>u</th>
<th>f</th>
<th>f</th>
<th>i</th>
<th>x</th>
</tr>
</thead>
<tbody>
<tr>
<td>e</td>
<td>l</td>
<td>6</td>
<td>4</td>
<td>.</td>
<td>a</td>
<td>r</td>
<td>p</td>
<td>a</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
   
   Set the ENUM suffix for E.164 query on DNS. Leaving it blank or setting it as “e164.arpa” to comply with RFC2916 standard ENUM suffix, otherwise you may set it to a proprietary suffix, such as “e164.net”. Default is “e164.arpa”.

   - [Del]: Delete one character.
   - [Abc./123.]: Toggle between digits and alphanumeric input, where [Abc.] indicates current input.
method is alphanumeric and [123..] indicates digits input.

- [Clear]: Clear all input.
- [Back]: Return without any changes.
- 【↲】: Save changes and return.
- 【<】 and 【>】: move cursor one position in edit mode.
- 【⋀】 and 【⋁】: move cursor per line in edit mode.

E.g., if the dial string is "+886-3-1234567", the ENUM query string send to DNS server to resolve will be:
  ■ Strip off all non-digits.
  ■ Reverse the string.
  ■ Insert a dot, ".", between each digit.
  ■ Append the ENUM suffix, such as "e164.arpa".
  ■ Send to DNS to resolve.
Thus, the resultant DNS query string would be: "7.6.5.4.3.2.1.3.6.8.8.e164.arpa".

3.2. Minimum Valid ENUM Digit Length:

<table>
<thead>
<tr>
<th>Minimum Valid ENUM Digit Length</th>
</tr>
</thead>
<tbody>
<tr>
<td>M i n</td>
</tr>
<tr>
<td>1 - 16</td>
</tr>
</tbody>
</table>

Configure the minimum digit length of a valid ENUM dial string.
A valid ENUM dial string must starts with a "#", which will be translated to as a '+', and follows at least "Min length" digits. If the rules does not apply, the dial string will be dial "as is" without any attempt to carry out an ENUM resolution. Default is 6 digits.

- [Del]: Delete one character
- [Back]: Return without any changes.
- 【↲】: Save changes and return.
- 【<】 and 【>】: move cursor one position in edit mode.

3.3. If YV3 fails to resolve an ENUM number to SIP address-of-record, AoR, it will render the request line based on the URI format you specified in menu:“URI Format”.

1. SIP URI: YV3 will send the request to the specified SIP service domain proxy in the following form:
   INVITE sip: +PhoneNumber@domain:port;user=phone
   For example, if you dial “+88631234567” and YV3 fails in ENUM resolution, it will send:
   INVITE sip:+88631234567@proxy.SIP.isp.com;user=phone

2. TEL URI: YV3 will send request to the specified SIP service domain proxy in the form:
   INVITE tel: +PhoneNumber
   For example, if you dial “+88631234567” and YV3 fails in ENUM resolution, it will send:
   INVITE tel:+88631234567

- [Back]: Return.
- 【↲】: Save changes and return.
- 【⋀】 and 【⋁】: Scroll menu items.

3.3.1. In either format, you could also configure whether the leading ‘+’ in request line should be replaced or not by configure “Int’l Access Code”.

[28/117]
This setting configures whether the leading ‘+’ should replace with an international access code, which is “00” in most countries, before actually sending it to proxy server. The default is leave the leading ‘+’ intact for proxy (or PSTN gateway) to replace it with the international access code based on their geographic locations.

- [Del]: Delete one character.
- [Abc../123..]: Toggle between digits and alphanumeric input, where [Abc..] indicates current input method is alphanumeric and [123..] indicates digits input.
- [Clear]: Clear all input.
- [Back]: Return without any changes.
- 【↲】: Save changes and return.
- 【<】 and 【>】: move cursor one position in edit mode.
- 【▲】 and 【▼】: move cursor per line in edit mode.

3.3 Redial

Press 【/rfc】 to dial the last dialed number. It will turn on the speaker phone automatically if the user has not done so yet.

3.4 Address Book

You can pick an contact from the address book, press [Call] context-sensitive soft-button to dial out the specified number; or you may lift the handset (offhook) or turn speaker on by pressing 【<rfc>】 to make a call to selected contact as well.

- r affili o : 1.Address book, 1.Search (Or press DSS key 【Address Book】 or context-sensitive soft-button 【AB】 to activate it).

Search for a specific entry on address book. The search criterion is a longest prefix match. If no character is entered, then it will position on the 1st entry on address book

- [Del]: Delete one character.
- [Abc../123..]: Toggle between digits and alphanumeric input, where. [Abc..] indicates current input method is alphanumeric and [123..] indicates digits input.
- [Clear]: Clear all input.
- [Back]: Return.
- 【↲】: Start to search.
- 【<】 and 【>】: move cursor one position in edit mode..

Enter the prefix of the target number (display) to search or press 【↲】 with any input to go to the first entry on address book.
Address Book List Format:
1. List all contacts in alphanumeric order.
2. Position at the best matched item.
3. User may use keypad to jump to the first contact prefixed with entered alphanumeric character.
4. Alternatively, user may use navigation key [▲] and [▼] to scroll contact list.
5. Max size: 1,000

- [Call]: Dial to selected contact. Alternatively, You may lift the handset (offhook) or turn speaker on by pressing 【idental】to make a call to selected contact as well.
- [Add]: Add new contact into address book.
- [Del]: Delete current contact.
- [Back]: Return without any changes
- 【↑】: Edit selected contact.
- 【▲】and 【▼】: Navigate through contacts list.
- On listing mode, press 【2】twice to first entry prefixed with an ‘A’ (or press 【8】consecutively for 3 times will jump to the first entry prefixed with a ‘U’, etc.)
- Press context-sensitive soft-button [Call] to dial the selected number. Alternatively, you may lift the handset (offhook) or turn speaker on by pressing 【目标】to make a call to selected contact as well.

You may configure your address book by pointing web browser to your terminal’s IP and go to page『YV3』/『Address Book』.

All entries on address book are sorted and listed based on the pattern “Display name <Protocol:Address>”. Address Book can accommodate 1,000 entries.
- New: click “New” on the top toolbar to add new contacts into your address book.
You can add max three new contacts into the address book each time.

- **Alias/Nickname/Display**: this field is optional.
  - It consists of any ASCII characters except for ‘<’ and ‘>’.
- **Protocol**: this field is a must.
  - Depends on the URL, you may choose “sip”, “sips” or “tel” as appropriate. Currently YV3 supports only “sip” and “tel”. Default is “sip”. (Protocol field MUST be in lower case, which is stipulated on RFC3261).
- **Username/Phone-No**: this field is a must.
  - You may include port, parameters and tags but not protocol field. For example: you may enter Michael’s SIP address-of-record as “michael@SIP.isp.com”, or his TEL number as “+886-3-5639025”. You may also enter someone’s SIP contact information by enter +886-3-5639025@sipt.SIP.isp.com:5060;user=phone.
  - If you specify only the user part (everything until the ‘@’) as the target number, then the dialled out domain will be the one you specified when you picked this entry from address book. Please refer to “Multi-domain note” in section 8.1-“Dialing Scheme” on “YV3 User’s Guide”.

- Click “Add” to add these new contacts into your address book, and return to address book listing. Those contacts having no “Username/Phone-No” will not be added. Besides, if any contact has been existed on the address book (both Protocol and Username/Phone-No are identical); only the Display part will be overwritten.

- **Edit:**
  - Click on the contact you want to edit.
  - Alternatively, you may check the contact you want to modify then click “Edit” on the top toolbar.
After finishing modification, you may:
- Click “Modify” button to save the changes.
- Click “Delete” button to remove it from your address book.
- Click “Add to Call Screen” button to add this contact into black list (if this contact has not been on the black list), such that all calls and messages originated from this contact will be blocked.
- Click “Remove from Call Screen” button to remove this contact from black list (if this contact has been on the black list), such that this contact may call you or send you messages.
- Click “Call” button to instruct your terminal to make an outbound call to this contact immediately. If the terminal is on hook, then it will enter hands-free mode automatically.
- Click “Cancel” to abandon your changes and return to address book listing.

Delete:
- Check those contacts you want to delete then click “Delete” on the top toolbar to delete it

Delete All:
- Check the “Select All” button to select all records.
- Click “Delete” on the top toolbar to remove all selected records.
3.5 Favorite

You can pick 100 contacts from address-book and configure them into favorite list such that you can have easy access to those frequently used numbers. To make a outbound call to a contact from favorite list, you may press [DSS] soft-button and execute “Favorite” command to activate favorite list; alternatively, you may press programmable keys on the top of LCD pannel to activate “Favorite” list if you have mapped it.

<table>
<thead>
<tr>
<th>1. Favorite</th>
<th>3. AddrBook</th>
</tr>
</thead>
<tbody>
<tr>
<td>2. IMPP</td>
<td></td>
</tr>
<tr>
<td>4\SIP Realm</td>
<td></td>
</tr>
</tbody>
</table>

Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:

[A/B] + Indicator(华盛顿北京上海) + Caller-ID

<table>
<thead>
<tr>
<th>Tue</th>
<th>09-26</th>
<th>Status Indicator(北京上海北京上海)</th>
</tr>
</thead>
<tbody>
<tr>
<td>7754</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

DSS

Besides, you may press [【】] to activate keypad menu and go to “1.Address Book” / “Favorite”:

<table>
<thead>
<tr>
<th>Michael</th>
<th>Mike</th>
<th>Nick</th>
<th>Patrick</th>
<th>Paul</th>
<th>Peter Robbins</th>
<th>Richard</th>
</tr>
</thead>
</table>

| Call | Add | Del | Back |

Favorite contacts are listed in alphanumeric order, and you may use keypad to jump to the first contact prefixed with entered alphanumeric character. Alternatively, you may use navigation key 【A】 and 【V】 to scroll contact list.
• [Call]: Call to selected contact. *Alternatively, You may lift the handset (offhook) or turn speaker on by pressing 【】 to make a call to selected contact as well.*

• [Add]: Add a new entry from address book into favorite list.

<table>
<thead>
<tr>
<th>Michael</th>
<th>Mike</th>
<th>Nick</th>
<th>Patrick</th>
<th>Paul</th>
<th>Richard</th>
<th>Rice</th>
<th>RICHARD</th>
</tr>
</thead>
</table>

- [Back]: return without any changes.
- 【】: Add selected contact into favorite list.
- 【】 and 【】: Navigate through address book.

• [Del]: Remove selected contact from favorite list.

• [Back]: return without any changes.

• 【】 and 【】: Scroll list.

• 【】: Review address-of-record

Show SIP address-of-record (Read only).

- [Call]: Dial to this contact. *Alternatively, You may lift the handset (offhook) or turn speaker on by pressing 【】 to make a call to selected contact as well.*
- [Del]: Remove this contact from favorite list.
- [Back]: Return.
- 【】 and 【】: Scroll Line.

Besides, you may configure favorite list by pointing web browser to the terminal and go to 【YV3】 / 【Address Book】 / 【Favorite】
• Add: Click “Add” button on the top toolbar to pick entries from address book and add them into favorite list. Position cursor on an input box, and click an entry on the “Contacts” panel at the right hand side to select them. To remove a contact, click the “Clear” button. Click “Add” button to finish addition.

• Delete: Remove contacts from favorite list.
Check those contacts you want to delete then click "Delete" on the top toolbar to remove them from favorite list.

Delete All:
- Check the "Select All" button to select all records.
- Click "Delete" on the top toolbar to remove all selected records.

Call: Clic-to-Call
Check the contact you want to call to, then click "Call" on the top toolbar. Then the terminal will make an outbound call to this contact. If the terminal is on hook, then it will enter hands-free mode automatically.
3.6 Call History

You can pick an entry from the call history (missed calls, received calls and dialed numbers); press [Call] soft-button to dial out the specified number. Those calls are listed on a last come first shown manner. You may use keypad to jump to specified entry, such as enter ‘9’ to jump to 9th entry.

- [Call]: Dial to selected record. **Alternatively, You may lift the handset (offhook) or turn speaker on by pressing [keydown] to make a call to selected contact as well.**
- [Del]: Remove selected record.
- [Clear]: Remove all records in this list.
- [Back]: Return.
- [keydown] and [keydown]: Navigate through the list.

To enter call history, you may press [CID] soft-button (or press [DSS] / [CID]):

<table>
<thead>
<tr>
<th>1. Favorite</th>
<th>3. AddrBook</th>
</tr>
</thead>
<tbody>
<tr>
<td>2. IMPP</td>
<td>4. SIP Realm</td>
</tr>
</tbody>
</table>

**Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:**

\[
[A/B] + Indicator(\text{\ding{51}}, \text{\ding{52}}, \text{\ding{53}}) + \text{Caller-ID}
\]
Then goes into call history list:

<table>
<thead>
<tr>
<th>Missed Calls</th>
<th>Received Calls</th>
<th>Dialed Numbers</th>
</tr>
</thead>
</table>

**Note**, you will go directly into “Missed Calls” list whenever there are any new missed calls (on which time there will be a ◀ icon shown on status indicator area and disappeared whenever “Missed Calls” list has been reviewed).

Alternatively, you may press 【】 to enter menu mode and go to “Call History” item

- [Missed]: Show the latest 20 missed calls.
- [Dialed]: Show the latest 20 dialed numbers.
- [Recvd]: Show the latest 20 received calls.
- [Back]: Return

You may configure your address book by pointing web browser to your terminal’s IP and go to page 【YV3】 / 【Call History】 => 【Missed Calls】，【Received Calls】 or 【Dialed Numbers】.

Show the most recent 20 (or fewer if not available) records of missed call, received call and dialed number respectively.

- Time of Call: the date / time format will be rendered based on the settings on 『Terminal Settings』/『Date /Time』 => “Date Format”.

Show the most recent 20 (or fewer if not available) records of missed call, received call and dialed number respectively.

- Time of Call: the date / time format will be rendered based on the settings on 『Terminal Settings』/『Date /Time』 => “Date Format”.
Number: Caller ID.

Edit:
- Click on the record you want to edit.
- Alternatively, you may check the record you want to edit then click “Edit” on the top toolbar.

After finishing modification, you may:
- Click “Modify” button to save the changes of this record.
- Click “Save” button to save it into address book. If this contact has been existed on the address book (both Protocol and Username/Phone-No are identical), only the “Display” part will be overwritten.
- Click “Delete” button to remove it from the call history.
- Click “Call” button to instruct your terminal to make an outbound call to this record immediately. If the terminal is on hook, then it will enter hands-free mode automatically.
- Click “Cancel” to abandon your changes.

Save:
- Check the record you want to save then click “Save” on the top toolbar.
- Modify the record, such as the Display and Click “Save” button to save it into address book, or click “Cancel” to abandon your changes and return to the address book listing.

Delete:
- Check those records you want to delete then click “Delete” on the top toolbar to delete it
- Delete All:
  - Check the “Select All” button to select all records.
  - Click “Delete” on the top toolbar to remove all selected records.

Call: Click-to-Call
Check the record you want to call to then click “Call” on the top toolbar. Then the terminal will make an outbound call to this record. If the terminal is on hook, then it will enter hands-free mode automatically.

3.7 Speed Dial
You may press fixed 【】 key in idle mode to enter speed dial mode. Alternatively, you may enter normal dial mode, then press fixed 【】 function key to enter speed dial mode.

| * IMPP | * Favorite |
Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:

\[ A/B \] + Indicator(\( \begin{array}{c} 8 \\ 9 \\ 0 \end{array} \) ) + Caller-ID

<table>
<thead>
<tr>
<th>A</th>
<th>S p e e d</th>
<th>D i a l</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>2</td>
<td></td>
</tr>
</tbody>
</table>

- [Call]- Dial out collected digits.
- [B] – Line switch to another channel.
- [Del] – Delete one input character
- [DSS]: User defined programmable keys (ID-1 to ID-4 go to the 4 softkeys on the upper half of LCD; whereas others, from ID-5 to ID-16 will be listed on pressing this key).

YV3 supports two different ways to perform speed dial:

i. \( [0-19] + \text{-} + [\text{-}] \); e.g., \( [1] + [\text{-}] \), or “01” + \( [\text{-}] \)

ii. \( [\text{-}] + [00-19] \); e.g., \( [\text{-}] + “01” \)

Note: If you want to dial a single digit only, such as \( [\text{-}] + [1] \), please press an additional [Call] soft-button to finish dialing; otherwise, it will wait for the second digit until inter-digit timeout, which default is 4 seconds. For example, \( [\text{-}] + [1] + [\text{-}] \).

<table>
<thead>
<tr>
<th>Key</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>( [\text{-}] ) +‘0’, ‘0’ + ( [\text{-}] )</td>
<td>Dial the 0th speed dial number.</td>
</tr>
<tr>
<td>( [\text{-}] ) +‘1’, ‘1’ + ( [\text{-}] )</td>
<td>Dial the 1st speed dial number.</td>
</tr>
<tr>
<td>( [\text{-}] ) +‘2’, ‘2’ + ( [\text{-}] )</td>
<td>Dial the 2nd speed dial number.</td>
</tr>
<tr>
<td>( [\text{-}] ) +‘9’, ‘9’ + ( [\text{-}] )</td>
<td>Dial the 9th speed dial number.</td>
</tr>
<tr>
<td>( [\text{-}] ) +‘10’ or ‘10’ + ( [\text{-}] )</td>
<td>Dial the 10th speed dial number.</td>
</tr>
</tbody>
</table>

If he phone have not been hooked off yet, the phone will turn on the speaker phone automatically.

To configure “Speed Dial” map, you may press \( [\text{-}] \) and go to “Address Book” / “Speed Dials”. You can configure up to 20 speed dial mappings (0–19), and you can use keypad to jump to specified entry, such as enter ‘9’ to jump to the 9th entry.

<table>
<thead>
<tr>
<th>0</th>
<th>M i c h a e l</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>M i k e</td>
</tr>
<tr>
<td>2</td>
<td>N i c k</td>
</tr>
<tr>
<td>3</td>
<td>P a t r i c k</td>
</tr>
<tr>
<td>4</td>
<td>P a u l</td>
</tr>
<tr>
<td>5</td>
<td>R i c h a r d</td>
</tr>
<tr>
<td>6</td>
<td></td>
</tr>
</tbody>
</table>

- [Call]: Dial to selected contact Alternatively, You may lift the handset (offhook) or turn speaker on by pressing \( [\text{-}] \) to make a call to selected contact as well.
- [Del]: Remove selected speed dial mapping.
- [Clear]: Remove all speed dial mappings
- [Back]: Return.
- \( [\text{-}] \) and \( [\text{-}] \) : Navigate through the list.
- \( [\text{-}] \) : re-map selected entry

M i c h a e l
M i k e
N i c k
P a t r i c k
Alternatively, you may configure the speed dial numbers by pointing web browser to your terminal’s IP and go to page『YV3』/『Address Book』/『Speed Dials』.

There are at most 20 mappings for speed dialing, i.e., [0-19]. It will list the current mappings of each key to the entry on address book. All entries are listed based on the pattern “Display Name <Protocol:Address>”. If a specific entry has no display name, then only the “Protocol:Address” is shown; otherwise the “Display Name” will be shown.

By default, all entries are unassigned. If the assigned entry has been erased from address book, the mapping will be removed from speed dials automatically.

- To assign a speed dial entry:
  - Position cursor on the text input which you want to assign a speed dial mapping.
  - Click the “Contacts” on the right panel to pick an entry from address book to map to a specific speed dial entry.

- To clear a speed dial mapping:
  - Position cursor on the text input which you want to clear the speed dial mapping.
  - Click “Clear” collocates with each speed dial entry to remove the mapping.

- Call: click-to-call
  - Position cursor on the interested speed dial entry then click “Call” button. Then the terminal will make an outbound call to this contact. If the terminal is on hook, then it will enter hands-free mode automatically.
3.8 Call Return

You can invoke [DSS] soft-button and execute programmable command “Call Return” to make a call back to the last incoming numbers (missed or received call). It differs from server supported call return, such as “*69”, in that it will find the latest incoming calls from “Missed calls” and “Received calls” then dial out the number.

3.9 One-Touch Dial

YV3 supports one-touch dial by mapping some DSS keys such that the user could dial out frequently used numbers by pressing the associated function key(s) directly.

To configure one-touch dial key(s), press 【】to activate menu and go to “3.Programmable Keys” to map a programmable key to desired contact. If mapped, it will list address-book for you to pick one contact to associate with this programmable key. Besides, whenever [DSS] soft-button is invoked, it will show the display (or user-part of its email-like address) whenever possible, and a  icon to indicate this programmable key is associated with “One-Touch Dial”.

Note: the associated contact will be invalidated whenever the corresponding address book entry has been erased.

You may also configure it from web browser by going to 【YV3】/【Terminal Settings】/【DSS Features】.
4 Taking Calls

If you are in menu mode, the phone will start ringing when incoming calls are waiting. To pick up the incoming call, you may
1. Lift the handset (off-hook).
2. Alternatively, you may exit menu first then take it as a normal incoming call (lifting the handset has no effect in menu mode).

4.1 Incoming Call while Idle

<table>
<thead>
<tr>
<th>* IMPP</th>
<th>* Favorite</th>
</tr>
</thead>
<tbody>
<tr>
<td>* DTMF</td>
<td>* Register</td>
</tr>
</tbody>
</table>

**Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:**

\[A/B\] + Indicator( \(\text{\ding{162}}\) \(\text{\ding{163}}\) \(\text{\ding{164}}\) \(\text{\ding{165}}\) ) + Caller-ID

User can press \([\Delta]\) and \([\nabla]\) navigation key to scroll soft keys on the bottom, where \([\text{\ding{175}}]/[\text{\ding{176}}]\) indicates that there are more soft keys, user may use \([\Delta]\) and \([\nabla]\) navigation key to scroll.

- \([\text{\ding{162}}A]\) – Pick up line 1 to answer this waiting call, where \(\text{\ding{162}}\) indicates a call waiting on A channel.
- \([B]\) – Pick up line 2 to seize an empty line for outbound call.
- \([\text{\ding{164}}]\) – Forward incoming waiting call to preconfigured target if available or reply it as busy if not available. This forwarded call will be recorded as a “received call”.

To configure the target number to forward to, please go to menu “4.Call Forward” / “Target Number.” (by TELNET or keypad); alternatively, you may refer to “Forward and DND” on this document for detail.

This may also be a network feature, handled from a network configuration by the system administrator.

- \([\text{Busy}]\) – Reject this incoming waiting call as busy and recorded it as a “received call”.
- \([\text{DSS}]: \) User defined programmable keys (ID-1 to ID-4 go to the 4 softkeys on the upper half of LCD; whereas others, from ID-5 to ID-16 will be listed on pressing this key).

To take this call, user may
- Turning on Speaker (Hands-free)
- Or pick up handset to take the call (handset)
- Or press related line key, \([\text{\ding{162}}A]\) or \([\text{\ding{162}}B]\), to take the call

If user presses the line key other than the incoming call line, the incoming call will be ignored, and can be taken by \([\text{\ding{162}}A]\) or \([\text{\ding{162}}B]\) later;

4.2 Incoming Call while Talking

<table>
<thead>
<tr>
<th>* IMPP</th>
<th>* Favorite</th>
</tr>
</thead>
<tbody>
<tr>
<td>* DTMF</td>
<td>* Register</td>
</tr>
</tbody>
</table>

**Screen Popup of incoming waiting call:**

User can press \([\Delta]\) and \([\nabla]\) navigation key to scroll soft keys on the bottom, where \([\text{\ding{175}}]/[\text{\ding{176}}]\) indicates that there are more soft keys, user may use \([\Delta]\) and \([\nabla]\) navigation key to scroll.

- \([\text{\ding{162}}A]\) – Answer the waiting call
- \([\text{\ding{162}}]\) – Forward incoming waiting call to preconfigured target if available or reply it as busy if not available. This forwarded call will be recorded as a “received call”.

To configure the target number to forward to, please go to menu “4.Call Forward” / “Target Number.”

*Soft-buttons for waiting call processing*

**Caller ID of active connected call.**

*Soft-buttons for waiting call processing*
(by TELNET or keypad); alternatively, you may refer to “Forward and DND” on this document for detail.

This may also be a network feature, handled from a network configuration by the system administrator.

- [Busy] – Reject this incoming waiting call as busy and recorded it as a “received call”.
- [DSS]: User defined programmable keys (ID-1 to ID-4 go to the 4 softkeys on the upper half of LCD; whereas others, from ID-5 to ID-16 will be listed on pressing this key).

User may press any keys (e.g., [0-9*#] or navigation keys) except for [Take], [FWD] and [Busy] to ignore this incoming call and continue his/her undergoing active call. Once those call-waiting context-sensitive soft-buttons ([Busy] and [FWD]) are disappeared in talking state, it will show:

| * IMPP      | * Favorite |
| * DTMF     | * Register |

[CB]  Marr@ISR.com

[ A ] 1 5 : 3 0
Joseph River
[ B ] D T M F
Info
Conf
DSS

- [ A B ] – Line switch to another channel and answer the waiting call.
- [Info] – Show channel information of current call.
- [Conf] – Toggle to switch between setting up and tearing down an ad hoc 3-way local conference.
- [DSS]: User defined programmable keys (ID-1 to ID-4 go to the 4 softkeys on the upper half of LCD; whereas others, from ID-5 to ID-16 will be listed on pressing this key).

You still could process the incoming waiting call by:
1. Press [ A B] to make a call-switch and answer the incoming waiting call.
2. Press [DSS] and execute [Forward] command to forward this waiting call.

```
1. Call History
2. My Presence Status
3. Call Return
4. Do Not Disturb (DND)
5. Network Status
6. Message
7. Register
8. DTMF List
9. Auto-Answer
10. DTMF Relay by INFO
11. Reject
```
```
12. Forward
```

This programmable command will forward an incoming ringing calls to a preconfigured forwarding target number. However, if this [Forward] command is executed when there is no waiting calls, it will enter “4 Call Forward” menu directly.

3. Press [DSS] and execute [Reject] command to reject this waiting call as “486 Busy”.

```
1. Call History
2. My Presence Status
3. Call Return
4. Do Not Disturb (DND)
5. Network Status
6. Message
7. Register
8. DTMF List
9. Auto-Answer
10. DTMF Relay by INFO
11. Reject
12. Forward
```

[45/117]
4.3 **Forward and DND**

You can configure the target number to forward to while this phone is busy or not answered within a predefined guarding interval. This forwarding number is also employed while the phone is engaged in Do Not Disturb (DND) mode or while the user presses [FWD] key on an incoming waiting call.

The system forwarding rules will check Do Not Disturb mode first, then All Calls Forward, Busy Forward, finally going to No Answer Forward while no-answer timer expires.

Note, your presence status will be ignored if it is set to On-Line (idle) state and you have enable the following features (in precedence):

<table>
<thead>
<tr>
<th>Terminal Features</th>
<th>Presence Status</th>
<th>RPID Activities</th>
</tr>
</thead>
<tbody>
<tr>
<td>All-Calls-Forward is enabled</td>
<td>open</td>
<td>Away</td>
</tr>
<tr>
<td>Logout of all SIP Service Domains</td>
<td>open</td>
<td>Away</td>
</tr>
<tr>
<td>Do-Not-Disturb (DND) is enabled</td>
<td>open</td>
<td>Busy</td>
</tr>
</tbody>
</table>

That is, this terminal will respond a “open” presence state with corresponding RPIDF activity (which is “Away” for All-Calls-Forward and logout, and “Busy” for DND) if ever got Presence SUBSCRIBEd from other terminals. Refer to 【】 => “1.Address Book” / “IMPP” / “State” on “YV3 Administration” or “IMPP” section on this document for detail.

4.3.1 **Do Not Disturb (DND)**

You can block incoming calls by invoking the Do Not Disturb, DND feature from programmable softkeys. Once enabled all incoming calls will be forwarded unconditionally to the forwarded target number if applicable; otherwise, incoming calls will be turned down as “486 Busy Here”. Blocked calls are logged in the Missed Calls.

i. Press [DSS] to activate programmable menu.

```
* IMPP * Favorite
* DTMF * Register
```

Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:

```
[A/B] + Indicator(...) + Caller-ID
```

<table>
<thead>
<tr>
<th>Tue</th>
<th>09</th>
<th>26</th>
<th>Status Indicator ( ) 1 5 : 5 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>7</td>
<td>7</td>
<td>5</td>
<td>4</td>
</tr>
</tbody>
</table>

```
DSS
```

ii. Execute “DND” to toggle Do Not Disturb feature.

```
1. Call History
2. My Presence Status
3. Call Return
4. Do Not Disturb (DND) Notice
5. Network Status
6. Message
7. Register
8. DTMF List
9. Auto Answer
10. DTMF Relay by INFO
11. Reject
12. Forward
```

4.3.2 **Call Forward**

To configure the target number to forward to while this phone is busy or not answered within a predefined guarding interval:
By keypad or TELNET:
- Press 【Menu】 to activate menu.
- Go to submenu “4. Call forward”.

<table>
<thead>
<tr>
<th>Target Setting</th>
<th>Number</th>
<th>Target Setting</th>
<th>Number</th>
</tr>
</thead>
<tbody>
<tr>
<td>2. All Calls Forward</td>
<td></td>
<td>3. Busy Forward</td>
<td></td>
</tr>
<tr>
<td>4. No Answer Forward</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

By web browser:
- Go to 『Call Forward』 page, and then click the “Contacts” on the right panel to pick an entry from the address book to set it as “Target Number”.
- Click “Clear” button to remote target number.

Note, This target forwarding number is also employed while the phone is engaged in Do Not Disturb (DND) mode or while the user presses [FWD] soft-button (or invoke via DSS programmable command, “Forward”) on an incoming waiting call.

4.3.2.1 All Calls Forward

You can configure to unconditionally forward all incoming calls by enable the All Calls Forward feature from menu “4. Call Forward” / “All Calls Forward” or on 『Call Forward』 web page. Forwarded calls are logged as “Missed Calls”. If this feature is enabled, LCD will display  as status indicator and show “FWD:TargetAoR” to remind user that all incoming calls will be forwarded unconditionally thereafter.

4.3.2.2 Busy Forward

You can configure to forward incoming waiting calls when the system is busy, on which time all lines are occupied, from from menu “4. Call Forward” / “Busy Forward” or on 『Call Forward』 web page. Forwarded calls are logged as “Missed Calls”. Default is disabled.
4.3.2.3 No Answer Forward

You can configure to forward incoming waiting calls after ringing for a predefined interval from menu “4. Call Forward” / “No Answer Forward” or on 『Call Forward』 web page. Forwarded calls are logged as “Missed Calls”.

Default is disabled

4.3.3 Forwarding Rules

<table>
<thead>
<tr>
<th>Priority</th>
<th>Conditions</th>
<th>Results</th>
</tr>
</thead>
<tbody>
<tr>
<td>All Calls Forward</td>
<td>[DND] is on</td>
<td>Forward the incoming call to the target forwarding number.</td>
</tr>
<tr>
<td></td>
<td>Forwarding number is available</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Forwarding number is unavailable</td>
<td>1. Reply as “486 Busy”.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>2. Recorded as a missed call.</td>
</tr>
<tr>
<td></td>
<td>Both</td>
<td></td>
</tr>
<tr>
<td></td>
<td>1. All calls forward feature is on</td>
<td>Forward the incoming call to the target forwarding number.</td>
</tr>
<tr>
<td></td>
<td>2. Forwarding number is available</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Others</td>
<td>Normal incoming call.</td>
</tr>
<tr>
<td>Busy Forward</td>
<td>System call capacity is full loaded</td>
<td>Forward the incoming call to the target forwarding number.</td>
</tr>
<tr>
<td></td>
<td>Both</td>
<td></td>
</tr>
<tr>
<td></td>
<td>1. Busy forward feature is on</td>
<td></td>
</tr>
<tr>
<td></td>
<td>2. The forwarding number is available</td>
<td></td>
</tr>
<tr>
<td></td>
<td>others</td>
<td>1. Reply as busy.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>2. Record as a missed call.</td>
</tr>
<tr>
<td>No Answer Forward</td>
<td>Press [FWD]/[Busy] or [DSS] =&gt; [Forward]/[Reject] on a ringing call.</td>
<td>Forward the incoming call to the target forwarding number.</td>
</tr>
<tr>
<td></td>
<td>Forwarding number is available</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Forwarding number is unavailable</td>
<td>1. Reply as busy.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>2. Record as a received call.</td>
</tr>
<tr>
<td></td>
<td>Both</td>
<td></td>
</tr>
<tr>
<td></td>
<td>1. No answer forward feature is on</td>
<td></td>
</tr>
<tr>
<td></td>
<td>2. No-answer timer expires.</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Forwarding number is available</td>
<td>Forward the incoming call to the target forwarding number.</td>
</tr>
<tr>
<td></td>
<td>Forwarding number is unavailable</td>
<td>1. Reply as no-answer.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>2. Recorded as a missed call.</td>
</tr>
</tbody>
</table>

Note: Those not-forwarded calls will be recorded as “Missed Calls” and retrievable at a later time to review. The status indicators will include a icon to inform user that there are un-reviewed missed calls; and clear this icon after the user has reviewed them.

* IMPP                                      * Favorite
* DTMF                                      * Register

Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:

[A/B] + Indicator(⁀ ⁀ ⁀ ⁀ ) + Caller-ID

<table>
<thead>
<tr>
<th>T u e</th>
<th>0 9 - 2 6</th>
<th>Status Indicator (⁀ ⁀ ⁀ ⁀ )</th>
<th>1 5 : 5 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>M y</td>
<td>D i s p l a y</td>
<td>N a m e</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>A</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>B</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>M  A B</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>C I D</td>
<td></td>
</tr>
</tbody>
</table>

Besides, whenever there are un-reviewed missed calls, if you press [DSS] to activate programmable menu and execute “Call History” (CID) command, it will enter “2.Call History” / “Missed Call” menu directly; otherwise, it will enter “Call History” menu.

1. C a l l H i s t o r y S t a t u s
2. M y P r e s e n c e S t a t u s
3. C a l l R e t u r n
4. D o N o t D i s t u r b ( D N D )
5. N e t w o r k S t a t u s
6. M e s s a g e
7. R e g i s t e r

[48/117]
<table>
<thead>
<tr>
<th>8</th>
<th>D T M F  L i s t</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>9</td>
<td>A u t o - A n s w e r</td>
<td></td>
</tr>
<tr>
<td>10</td>
<td>D T M F  R e l a y  b y  I N F O</td>
<td></td>
</tr>
<tr>
<td>11</td>
<td>R e j e c t</td>
<td></td>
</tr>
<tr>
<td>12</td>
<td>F o r w a r d</td>
<td></td>
</tr>
</tbody>
</table>

Back
5 Connected

<table>
<thead>
<tr>
<th></th>
<th>IMPP</th>
<th>Favorite</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>DTMF</td>
<td>Register</td>
</tr>
</tbody>
</table>

Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:

\[ [A/B] + \text{Indicator}( \text{Call-ID} ) + \text{Caller-ID} \]

5.1 Mute

During a conversation, you can mute the voice such that you can hear voice from the peer but the peer cannot hear you.

To mute a active call:

- Set up a connected call.
- Press \([\text{Mute}]\) to toggle your voice transmission. On muting state, the red LED of \([\text{Mute}]\) will be on, and you can hear still the peer’s voice; however they cannot hear you.

5.2 Hold

During a conversation, you can put a call on hold then put the handset back to on-hook, and this call will be kept on the background without lost connection.

i. To put the peer on hold, press \([\text{Hold}]\) once:

1. if you are in hand-set mode, it will show:

<table>
<thead>
<tr>
<th></th>
<th>IMPP</th>
<th>Favorite</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>DTMF</td>
<td>Register</td>
</tr>
</tbody>
</table>

   Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:

   \[ [A/B] + \text{Indicator}( \text{Call-ID} ) \]

i. To put the peer on hold, press \([\text{Hold}]\) once:

2. if you are in hands-free mode, it will goes into idle mode.

<table>
<thead>
<tr>
<th></th>
<th>IMPP</th>
<th>Favorite</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>DTMF</td>
<td>Register</td>
</tr>
</tbody>
</table>

   Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:

   \[ [A/B] + \text{Indicator}( \text{Call-ID} ) \]

\(\Rightarrow[A] \text{ Jason: Screen popup for holding call.}\)
\(\Rightarrow[A] \text{ Holding call on channel A.}\)
ii. To retrieve the call from holding state, you could:
   1. Press 【 助助 】 again to unhold the peer.
   2. Press 【 A】 or 【 B】 context-sensitive soft-button to unhold the holding call.
   3. On idle mode, press 【 助助 】 to unhold the last holding call.

iii. If you are put on hold by the peer, your terminal will start playing music-on-hold and LCD will show:

   | Screen Popup for call-waiting, holding calls, Conference or Auto-Redial: |
   |\[A/B\] + Indicator( .attributes ) + Caller-ID |
   |      J a s o n        3 : 1 2              |
   | B         Info       *Conf   DSS             |

   Note: there will be hold recall timer to alert the user while some calls are holding. For example, if you put [A Call] on hold for more than one minute (which is configurable through 【 助助 】 => “5. Preferences” / “Hold Recall Timer” by keypad, or on web page『YV3』/『Preferences』 => “Hold Recall Alerting Timeout (s)”):

   • If you are in on-hook (idle) state:
     i. The phone will start ringing as an incoming call arrives.
     ii. The recall state continues ‘till user picks up the call. However, if user does not answer the call within 1 minute, the call will be disconnected.

   • If you are in off-hook state, either engaged in other calls or about to making calls:
     i. No ringing tone will ever be played.
     ii. Play hold recall tone once to remind the user that some calls are in hold.
     iii. Reset the hold recall timer for next time alarm.

   Alternatively, you may configure this “Hold Recall Timer” on web page『YV3』/『Preferences』 => “Hold Recall Alerting Timeout(s)”.
5.3 Transfer

Call Transfer allows a user, on any existing two-party call, to place the existing call on hold and originate another call to a third party. The user may consult privately or connect the original call to the third party.

There are two types of transfers:

- **Blind Transfer**—With blind transfer you transfer the caller to another party without announcing the caller to the called party. To blind transfer a caller, you would press the [Hold] button to place the caller on hold. You will hear a dial tone and can dial the number you want to transfer the call to. It will immediately transfer the call and you could hang up as soon as finishing dialing or wait until the 3rd party picks up then disconnect from the other party.

- **Consultative Transfer**—With consult transfer you consult the other party before transferring the call. To consult transfer a caller, you would press the [Hold] button to place the caller on hold. You will hear a dial tone and can dial the number you want to transfer the call to. Wait until the other party answers the call and inform them you are transferring the call. Press the [Hold] button on the phone set to complete the transfer or just hangs up the call it will also perform the transfer.

Note, you can not transfer two inbound calls to each other. Instead, you can blindly transfer each call to a third party.

5.3.1 Consultative Transfer

Follow the following steps to do a **consultative transfer: announce it first before actually transfer the call.**

- Set up a connected call to party-A
- Press the [Hold] button to put A on hold and gain dial-tone.
- Dial the target number of party-B and press [Call] soft-button to finish dial.
- Wait 'till party-B picks up, announcing the call from party-A (consultation).
- If party-B is unwilling to take this call, you should ask party-B to hangs up first, then press line
soft-button, [A] or [B] to take back party-A. Since then, you may redo the transfer procedure from step-1.

- **Just hang up the call to B or press [ ] again** to finish transferring party-A to party-B. Note, you may do so even when B has not picked up the consultation call yet (such as it is in ringing back phase) to perform a call-transfer; and this is the so-called Semi-attended transfer or ring-transfer.

- (Optional) Alternatively, you may [Conf] context-sensitive soft-button to hold a 3-way conference with both party-A and party-B instead of transferring party-A to party-B.

Alternatively, you may do a consultative transfer as follows:

- Take an incoming call from party-A on [A] channel.
- Get an empty channel by pressing context-sensitive soft-button [B] to perform a call switch (effectively put A on hold), then make an outbound call to party-B.
- After talking to party-B, you decide to transfer A to B, press [ ] while talking to party-B to perform a call-transfer. Note, you may do so even when B has not picked up the consultation call yet (such as it is in ringing back phase) to perform a call-transfer; and this is the so-called Semi-attended transfer or ring-transfer.

- However, if you decide to host a conference among party-A and party-B rather than transfer from party-A to party-B, you should press [Conf] context-sensitive soft-button to conference both parties in.

### 5.3.2 Blind Transfer

Follow the following steps to do a **blind transfer: transfer the call without asking the transferring target first**.

- Set up a connected call to party-A
- Press [ ] to put A on hold and gain dial-tone.
- Dial the target number of party-B then just hang up (on B-channel) as soon as finishing dialing B’s number without pressing [Call] soft-button to blindly transfer party-A to party-B. Alternatively, you may press [ ] again after finishing dialing B’s number without pressing [Call] soft-button to blindly transfer party-A to party-B

**Note**, if you press [Call] context-sensitive soft-button after finishing dialing to party-B, it behaves just the same as consultative transfer.

Alternatively, you may do a blind transfer as follows:

- Take an incoming call from party-A on [A] channel.
- Get an empty channel by pressing context-sensitive soft-button [B] to perform a call switch (effectively put A on hold), and gain dial tone.
- Dial the target number of party-B without pressing [Call] soft-button; instead, you SHOULD press [ ] (on B-channel) as soon as finishing dialing B’s number to blindly transfer party-A to party-B.

**Note**, you can not transfer two inbound calls to each other. Instead, you can blindly transfer each call to a third party.

### 5.4 Conference

**YV3** supports ad hoc 3-way local conference. The context-sensitive soft-button [Conf] during connected call set allows the user to set up a conference call with a number of people.

Press [Conf] context-sensitive soft-button to set up a conference call when you have 2 connected calls, which one is active and the other is put on hold on the background:

<table>
<thead>
<tr>
<th>*</th>
<th>IMPP</th>
<th>*</th>
<th>Favorite</th>
</tr>
</thead>
<tbody>
<tr>
<td>*</td>
<td>DTMF</td>
<td>*</td>
<td>Register</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>3</th>
<th>:</th>
<th>1</th>
<th>2</th>
</tr>
</thead>
<tbody>
<tr>
<td>[A]</td>
<td>Mary</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

[53/117]
If the conference could be set up, it will enter the following mode:

| * IMPP | Favorite |
| * DTMF | Register |

\[A\]  Mary

\[B\]  Conference

\[C\]  3 : 1 2

\[D\]  Jason

On the contrary, you may follow the following procedure to set up a 3-way local conference if you are starting from scratch:

- Call the first person
- Press the [Conf] context-sensitive soft-button to put current call on hold, and gain dial tone.
- Call the second person and it will enter 3-way local conference mode whenever getting connected with 2nd party.
- Repeat until all callers are added to the call. To conference in additional callers, the last person called with an YV3 can call an additional person, that new person can then call someone else, and so on. This is called “daisy-chaining.

Note: During a conference, an auditable tone will be played regularly from the hosting phone to notify all parties that a conference is undergoing. The alerting tone is as the same as the hold-recall tone and the alerting interval is identical to the hold-recall interval set in [DSS] => "5. Preferences" / "Hold Recall Timer". You may disable this conference altering tone from [DSS] => "5. Preferences" / "Conference Alert". Alternatively you may configure them on web page [YV3] / [Preferences] => “Hold Recall Alerting Timeout(s)” and “Play Alerting Tone during Conference”.
5.4.1 Constraints

When a conference is undergoing, YV3 will take or receive no more calls (just like in busy state).

5.4.2 Conference Tips

- Once the conference call initiator disconnects, the calls are terminated between all parties (because your terminal is serving as a media mixer such that the other two parties may hear each other). Under some circumstances, you may want to keep the other two parties connected even after you have exited this conference. By enabling `YV3` / `Preferences` => “Transfer after Conference Teardown” (or `Preferences` => “5.Preferences” / “Xfer on Exit Conf”), this terminal will carry out an attended transfer on exiting ad hoc 3-way local conference. As a result, the other two parties may keep conversation without your involvement.
To disconnect a specific party in a three-way local conference, you may ask the party to whom you want to disconnect to hang up first.

To place a conference call on hold, press the 【 】 key. The other parties cannot talk among themselves. To avoid disrupting the other callers, consider muting the call instead. To mute the call, press the 【 】 button.

To place a conference call on speakerphone, press 【 】.

Press 【 】 to mute the speakerphone. The conference parties cannot hear you but you can hear them.

The phone only allows three parties in a conference call.

Call waiting calls cannot be conferenced in; the conferencing node must establish both legs of the 3-Way conference.

A holding call will be unhold automatically before adding into a conference. Besides, you could put all parties on hold after setting up conference.

Muting calls will be un-mute automatically before being conferenced in.

You cannot perform transfer on conference calls.

To conference in additional callers, the last person called with an YV3 can call an additional person, that new person can then call someone else, and so on. This is called “daisy-chaining.”
5.5 Group Listening

Group-listening (a.k.a. open listening or loud-sounder), which is to facilitate meeting where there are many
listeners at the same room but only one person is talking at any time. If the talker is far away from the phone-set,
then it may be inappropriate to employ hands-free mode since the volume heard by the peer will be too small to
tell (otherwise the talker has to shout). Instead, the talker should talk by handset to make the voice more clear to
the peer and turn on the speaker to make the conversation heard by all by-listeners.

The group-listening mode supports in YV3 works as follow: if you switch to hands-free mode with handset
lifted (off-hooked), the phone will be in group-listening mode, which it continues transmitting voice from handset,
(not from microphone), and played voice to both speaker and handset receiver simultaneously. The phone will
activate microphone for voice transmission only when it is on hands-free mode and the handset is on hook. This
applies to ear-phone mode as well: in group-listening mode with ear-phone enabled, the ear-phone could hear
voice but not speak-able unless the handset is on hook.

To enable group-listening mode in YV3:

- Press 【】 to switch between speaker-phone and handset. The steady green LED of 【】 indicates whether speaker-phone is on or off.
- However, if you are in ear-phone mode, then press 【】 will switch between handset and ear-phone.
  The flashing green LED of 【】 indicates it is now in ear-phone mode.
- To switch to group-listening, you should turn the speaker on while keeping handset lifted (off-hooked).
  Then you can continue talking by handset while all by-listeners in the same room could hear your
  conversation from the speaker.
6 Terminated

6.1 Terminated in Calling State

* IMPP  * Favorite
* DTMF  * Register

Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:

<table>
<thead>
<tr>
<th>[A/B] + Indicator(  ⌐  ⌐  ⌐  ⌐ ) + Caller-ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>[A ]</td>
</tr>
<tr>
<td>Released</td>
</tr>
<tr>
<td>7 7 5 0 @ ISP.com</td>
</tr>
</tbody>
</table>

[B] – Line Switch to another channel
[Auto] – Toggle-switch to enable/disable auto-redial
[DSS]: User defined programmable keys (ID-1 to ID-4 go to the 4 softkeys on the upper half of LCD; whereas others, from ID-5 to ID-16 will be listed on pressing this key).

You can press [Auto] after finishing dialing while making a phone call but before connected or hanging up. After activating auto-redial, the system will launch the auto-redial process and re-dial the target number regularly till “connected” (see below). To manually cancel auto-redial, press [Stop].

Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:

<table>
<thead>
<tr>
<th>[A/B] + Indicator(  ⌐  ⌐  ⌐  ⌐ ) + Caller-ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>[A ]</td>
</tr>
<tr>
<td>Auto-Renal – Trying</td>
</tr>
<tr>
<td>7 7 5 0 @ ISP.com</td>
</tr>
</tbody>
</table>

[B] – Line Switch to another channel
[Stop] – Cancel Auto-Redial
[DSS]: User defined programmable keys (ID-1 to ID-4 go to the 4 softkeys on the upper half of LCD; whereas others, from ID-5 to ID-16 will be listed on pressing this key).

6.2 Terminated in Connected State

6.2.1 Peer Disconnected

* IMPP  * Favorite
* DTMF  * Register

Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:

<table>
<thead>
<tr>
<th>[A/B] + Indicator(  ⌐  ⌐  ⌐  ⌐ ) + Caller-ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>[A ]</td>
</tr>
<tr>
<td>Released</td>
</tr>
<tr>
<td>7 7 5 0 @ ISP.com</td>
</tr>
</tbody>
</table>

[B] – Line switch to another channel
[Stop] – Cancel Auto-Redial
[DSS]: User defined programmable keys (ID-1 to ID-4 go to the 4 softkeys on the upper half of LCD; whereas others, from ID-5 to ID-16 will be listed on pressing this key).

Where “mm:ss” indicates the duration of the call. If the user is on speaker phone mode, it will return to IDLE state in 3 seconds.

[58/117]
6.2.2 User Hangs Up

1. Ringing calls are waiting => Back to Ringing State

<table>
<thead>
<tr>
<th></th>
<th>IMPP</th>
<th></th>
<th>Favorite</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>DTMF</td>
<td></td>
<td>Register</td>
</tr>
</tbody>
</table>

[ A ] 0 0 : 0 3

From Scott

A B FWD Busy

2. Answered calls are holding:

<table>
<thead>
<tr>
<th></th>
<th>IMPP</th>
<th></th>
<th>Favorite</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>DTMF</td>
<td></td>
<td>Register</td>
</tr>
</tbody>
</table>

[B] Jeffery <91888@ISP.com>

Tu e 0 9 - 2 6 Status Indicator ( ) 1 5 : 5 2

7 7 5 4

A B AB CID

The “Holding Calls Recall Timer” will be triggered on expiry to alert user that there are still some holding calls. User could retrieve the holding call by

i. Picking up handset
ii. Or turning on speaker
iii. Or pressing 【 】 to retrieve it;

Otherwise the holding call will be disconnected after ringing for 1 minute.
7 Terminal Settings

7.1 Management Accounts

You need to provide account/password to log into YV3 by a web browser, telneting into YV3, and you need to provide a password when you want to unlock your terminal (see below “Phone Lock”).

There are 2 levels of management accounts on YV3:

i. Administrative account, which default account/password is “admin”/”0000”, “administrator”/”0000” or “root”/”0000”, excluding the double quotes.

ii. User-level account, which default account/password is “user”/”0000”, excluding the double quotes.

YV3 utilizes HTTP digest authentication before authorizing web browser access to configuration. Login user names, [“admin” | “administrator” | “root”] are recognized as administrative account and “user” is recognized as user-level account.

To change those passwords, you may:

• Press 【】to active menu.
• Go to submenu “3. Terminal Settings” / “Password”

   1. Password
   2. Programmable Keys
   3. Date/Time
   4. Ring
e
   5. Alert - Info
   6. LCD Brightness
   7. Language
   8. Alarm
   9. Phone Lock

   • [Admin]: Modify password to administrator’s account, whose login name could be “admin”, “administrator” or “root” (excluding the double quotes, ‘”’).
   • [User]: Modify password to user’s account, whose login name is “user” (excluding the double quotes, ‘”’). User-level account has limited access to some configuration to prevent accidental user wrong configurations.
   • [Back]: Return.
   • 【】and 【】: Navigate through menu items.

   • Terminal password consists of all digits, 0-9, only, and display star sign, ‘*’, for security reasons.

Note, if you have ever forgotten your terminal password (administrative and user-level) after locking your terminal, you may enter hardware reset password, “4273927373738#” (HardwareReset#) to restore everything back to shipping values then auto-reboot. Whenever this string “4273927373738#” is entered as dial-string or as password to unlock the terminal, it will trigger a hardware reset. After properly boot up, the default passwords to both user-level and privileged-level are reset to “0000”. This, however, will clear all settings, including personal information, such as address book, call history, instant messages, etc. Moreover, such hardware reset password can only be entered via keypad applicable in neither telnet nor HTTP web page) for security reasons.

Alternatively, you may reset terminal password by pointing your web browser to YV3 and goes to page YV3 / Terminal Settings / Account:
Note, if you forget user-level password, fill in the password of administrative account as the old password of "User-Level Account" and fill in new password accordingly to reset your user-level password; otherwise, you must reset it via keypad UI on the terminal.

7.2 Phone Lock

You can lock your terminal to prevent from kids misuse or for door-phone usage at the lobby (such that visitors may only dial phone numbers but not be able to check call-history and change configurations).

To lock you terminal, you may goes to

- Press 【🔄】 to active menu.
- Go to submenu “3. Terminal Settings” / ”Phone Lock”.

1. Off: The terminal is not in security mode.
2. Function Keys: This terminal is locked on security level. In security mode, the following keys will be protected (key press will be ignored):
   - 4 programmable hard keys located on the upper half of LCD.
   - Menu Key: 【🔄】
   - Redial key: 【🔄】
   - Message Waiting Indication: 【��🔄】
   - Context-sensitive soft-keys:
     - [CID] (Call History)
3. Take Call Only: Lock the phone set such that no outbound calls could be made and the phone
could not be configured either. That is, the phone would leave in “receive only” mode (No dial
tone could be heard).

Once locked in either Function Keys (security) mode or take call only mode, you need either user-level or
privileged password to unlock it.

**Note**, if you have ever forgotten your terminal password (administrative and user-level) after locking your
terminal, you may enter hardware reset password, “423927373738#” (HardwareReset#) to restore everything
back to shipping values then auto-reboot. After properly boot up, the default passwords to both user-level and
privileged-level are reset to “0000”. This, however, will clear all settings, including personal information, such
as address book, call history, instant messages, etc. Moreover, such hardware reset password can only be
entered via keypad applicable in neither telnet nor HTTP web page) for security reasons.

Alternatively, you may point web browser to YV3 and goes to page [YV3]/[Terminal Settings] => “Phone
Lock”:

![YV3 - [192.168.3.93] - Microsoft Internet Explorer](image)

### 7.3 Ear-Phone

YV3 will auto-detect ear-phone to switch between speaker-phone and ear-phone. That is if both microphone
and audio output jacks of ear-phone are connected (plugged in), YV3 will route audio I/O to ear-phone instead of
speaker-phone in hands-free mode. As a result, you may press 【】 to toggle between handset and ear-phone;
otherwise, press 【】 to toggle between handset and speaker-phone.

Unplug either of the ear-phone I/O jacks will disable ear-phone and switch back to speaker-phone mode.
7.4 **Volume Adjustment**

7.4.1 **Ringer**

You could adjust the ringing volume either during the phone set is ringing or from the menu.

- **Press** 【↑】volume adjustment keys on the lower-right corner of your terminal to adjust the ringing volume while the phone set is ringing.

  ![Ringer Volume](image)

- **To adjust ringer volume from menu:**
  - Press 【Esc】to active menu.
  - Go to submenu “3. Terminal Settings” / "Ringer" => [Vol] context-sensitive soft-button.
  - Adjust the ringing volume by 【↑】volume adjustment keys on the lower-right corner of your terminal, and it will play the ringing tone on adjustment.

Besides, you may pick another ringing tone to play on calls arrival:

- Press 【Esc】to active menu.
- Go to submenu “3. Terminal Settings” / "Ringer" => [Ring] context-sensitive soft-button.
- Pick your favorite ringing tone to use when incoming calls arrive.

To configure it from web page, go to 【YV3】/『Terminal Settings』:
7.4.1.1 Alert-Info

YV3 supports “Alert-Info” header in the first SIP INVITE message as per RFC3261, “Alert-Info” header dictates the phone to use an alternative ringing tone, which is specific for that call. The header should be in a format similar to “Alert-Info: <http://MediaServer.ISP.com/Announce.pcmu>;AnyParameter=xxxx” or “Alert-Info: xxxx”, where “xxxxx” (case-insensitive) is the tone tag for one of the available ringing tones (0~10). This is useful to distinguish calls, for example local calls from calls coming from PSTN. Besides, this header is ignored for re-INVITE. If the specified tone is out of range, the current ringer is used.

Besides, you may pick another ringing tone by to play on calls arrival:

- Go to submenu “3. Terminal Settings” / "Ringer" => [Ring] contest-sensitive soft-button.
- Pick your favorite ringing tone to use when incoming calls arrive.

You may re-map which tone should be played when “xxxx” tone tag is specified. For example, if “Alert-Info: ringtone-0” is mapped to “Ringer7”, then “Ringer7” will be played whenever “Alert-Info: ringtone-0” or “Alert-Info: <http://SIP.ISP.com/file.pcmu>;anyParameter=ringtone-0” is specified in the initial INVITE.

- Go to submenu “3. Terminal Settings” / "Alert-Info”.
- Remap the matrix of Alert-Info by ringtone-X.

Alternatively, you may point your web browser to [YV3] / [Terminal Settings] / [Alert-Info]. From this page, user may re-map which tone should be played when “xxxx” tone tag is specified. For example, if “Alert-Info: ringtone-0” is mapped to “Ringer0”, then “Ringer0” will be played whenever “Alert-Info: ringtone-0” or “Alert-Info: <http://SIP.ISP.com/file.pcmu>;anyParameter=ringtone-0” is specified in the initial INVITE.

Default is to map “ringtone-0” to Ringer0, “ringtone-1” to Ringer1, and so on.

7.4.2 Handset

Handset Volume:

Activate when speaker phone is off and the phone set is hooked off or at least one call is engaged. Use [;]
volume adjustment keys on the lower-right corner of your terminal to adjust to adjust handset volume.

Note, if you adjust volume in group-listening (a.k.a. open-listening or loud-speaker) mode, it adjust only hands-free volume rather than that of handset.

### 7.4.3 Hands-Free (Speaker-Phone and Ear-Phone)

**Speaker Phone:**

Activate when hands-free mode is on or while at least one call is engaged. Use volume adjustment keys on the lower-right corner of your terminal to adjust hands-free volume.

If you are in speaker-phone mode (either of the ear-phone I/O jacks is disconnected), you can adjust speaker-phone volume as:

**Speaker Phone:**

On the contrary, if you are in ear-phone mode (both of the ear-phone I/O jacks are connected), you can adjust ear-phone volume as:

**Ear Phone:**

### 7.5 Call-Progress Tones

*YV3* supports various call-progress tones and you can configure them on web page *[YV3]* / *[Terminal Settings]* / *[Tones]*. (Note, reconfigure Call-Progress Tones demands a terminal reboot to make your new changes applied!):

- **Dial Tone:** default value is (350, 0, 0, 440, 0, 0, 0)
- **Busy Tone:** default value is (480, 500, 500, 620, 500, 500, 0)
- **Ring-back Tone:** default value is (440, 1200, 3000, 480, 1200, 3000, 0)
- **Call Waiting Tone:** default value is (350, 100, 100, 440, 100, 100, 3)
- **Alert Tone:** default value is (440, 200, 0, 440, 200, 0, 1)
- **Reorder Tone:** default value is (480, 250, 250, 620, 250, 250, 0)
- **Stutter Dial Tone:** default value is (350, 100, 100, 440, 100, 100, 3)

- There is an incoming call waiting for answer during connected mode.
- Auto-redial feature is activated.
- Regularly hold-recall alerting tone.

*Note, this tone will be played regularly which is controlled by *[Call Settings]* / *[Call Waiting Alarm]*.
Interval (s) /.

- Alert Tone: default value is (440, 200, 0, 440, 200, 0, 1)
  - In conference mode, play to all parties to remind users of an undergoing meeting.
- Reorder Tone: default value is (480, 250, 250, 620, 250, 250, 0)
  - This tone is played when the called number is not available ("404 Not Found") or the external circuit is busy.
- Stutter Dial Tone: default value is (350, 100, 100, 440, 100, 100, 3)
  - Unconditional forward feature is on.

You may re-configure them to fit various countries’ telecommunication regulations (note, reconfigure Call-Progress Tones demands a terminal reboot to make your new changes applied!).

Each tone is specified by 7 integers as follows:

- \( Frequency1, OnTimeMs1, OffTimeMs1, Frequency2, OnTimeMs1, OffTimeMs2, Repetition \)

- \( Frequency1, Frequency2: \)
  - Specify the frequencies of the first and second frequencies, respectively.
  - To configure a single frequency tone, you should set \( Frequency2 \) to zero.

- \( OnTimeMs1, OnTimeMs2: \)
  - Specify the lengths of time measured in milliseconds the tone is played for the first and second on-off pairs of a cadence, respectively (see below for graphic representation).

- \( OffTimeMs1, OffTimeMs2: \)
  - Specify the lengths of time measured in milliseconds that silence is played for the first and second on-off pairs of a cadence, respectively (see below for graphic representation).

- \( Repetition: \)
  - Control the count of one-off pairs the tone is played. If this value is set to 0, the tone will play until another call event stops the tone.

- \( Tone \rightarrow OnTimeMs1 \rightarrow OnTimeMs1 \rightarrow \) on-off pairs.

- \( Silence \rightarrow OffTimeMs1 \rightarrow OffTimeMs1 \rightarrow \) on-off pairs.

Note, reconfigure Call-Progress Tones demands a terminal reboot to make your new changes applied!
8 Call Processing

8.1 Call Pickup

YV3 supports directed call pickup as well as group call pickup to pick up ringing calls not destined to your own terminal.

For example, if Bob and Peter are part of a work group at example.com that can pick up each others calls. Alice calls Bob who does not answer. Peter wishes to pick up the call then he could dial this specified “Call Pickup” code followed by Bob’s number, such as “*01Bob”, where “*01” is the assumed “Call Pickup” code. Then Peter’s terminal will send a SUBSCRIBE to Bob to retrieve the early (ringing) dialog information. Peter then generates an INVITE with a Replaces to Alice. Alice answers the INVITE and sends a CANCEL to stop Bob’s phone ringing. Note that the order of the CANCEL/ACK sequence in message-7 through message-8 is not significant.

To specify the group pickup code to pick up a ringing call of the same group (RFC4235-Dialog Event Page, RFC4462-Event Notify for Resource List, RFC2387-Multipart-Related MIME type, and draft-ietf-sipping-service-examples-10.txt):

- Press 【 】 to active menu.
- Go to submenu “5. Preferences” / “Dial Plan” / “Call Command” => “Call Pickup”.

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<tbody>
<tr>
<td>1</td>
<td>M S A C :</td>
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</tr>
<tr>
<td>2</td>
<td>Anonymous Call :</td>
<td>* 6 7</td>
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<tr>
<td>3</td>
<td>CLIP :</td>
<td>* 8 2</td>
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<td>4</td>
<td>M W I :</td>
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<td>5</td>
<td>Server Hold :</td>
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<tr>
<td>6</td>
<td>Call Pickup :</td>
<td>* 0 1</td>
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<tr>
<td>7</td>
<td>Join :</td>
<td>* 0 2</td>
<td></td>
<td></td>
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<tr>
<td>8</td>
<td>Feature Code 0 :</td>
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</table>
8.2 Join

You can join an existing conversation to have a conference among all joined parties, or ask somebody to join your ongoing conversation to have a 3-way local conference among them as per RFC3911.

For example, if Bob is talking to Alice and Bob wants his assistant to join this conversation, then Bob could request his assistant to “join” his current conversation by a gesture, via MSN or directly ask her to do so. Then the assistant should dial this “Join” feature activation code followed by Bob’s number, such as “*02Bob”, where “*02” is the assumed “Join” code. Then the assistant’s terminal will send a SUBSCRIBE to Bob to retrieve the confirmed (talking) dialog information. The assistant then generates an INVITE with a Join to Bob. Bob auto-answers the INVITE and set up a 3-way local conference among Alice, the assistant and himself, which results in his assistant “Join” an existed conversation in his terminal.
To specify the “Join” feature activation code:

- Press 【】 to active menu.
- Go to submenu “5. Preferences” / "Dial Plan" / "Call Command" => “Join”.

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<table>
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<tr>
<td>1.</td>
<td>M S A C :</td>
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<td>2.</td>
<td>A n o n y m o u s C a l l : * 6 7</td>
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<tr>
<td>3.</td>
<td>C L I P : * 8 2</td>
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<td>4.</td>
<td>M W I :</td>
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<tr>
<td>5.</td>
<td>S e r v e r H o l d :</td>
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<tr>
<td>6.</td>
<td>C a l l P i c k u p : * 0 1</td>
<td></td>
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<td></td>
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<tr>
<td>7.</td>
<td>J o i n : * 0 2</td>
<td></td>
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<tr>
<td>8.</td>
<td>F e a t u r e C o d e 0 :</td>
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<tr>
<td>9.</td>
<td>F e a t u r e C o d e 1 :</td>
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</tr>
<tr>
<td>10.</td>
<td>F e a t u r e C o d e 2 :</td>
<td></td>
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</tbody>
</table>

- Default Call Pickup feature invocation code is “*02”.

To configure it from web page, go to 『YV3』 / 【Preferences】 / 【Call Command】 => “Join”:
8.3 DTMF Transmission and Prefix Dial

You may specify a sequence of DTMF sequences to dial in pre-dial phase or to transmit after call setup. This is shortcut to transmit a sequence of preconfigured DTMF keys, which aims to facilitate IVR system interaction. For example, you may configure your personal ID or banking account, and just activate corresponding DTMF entry to generate those pre-configured DTMF tones whenever appropriate (such as inquired by tele-banking system).

If a DTMF entry is activated during digit-collecting phase, then those DTMF keys will be collected in a manner identical to those manually pressed by user, which is useful if some prefixes (such as a password for long-haul call) is required on making outbound calls (a.k.a. prefix-dial or preset-dial).

To configure DTMF sequences, you may:

- Press  to active menu.
- Go to submenu “5. Terminal Settings” / “Programmable Keys” => [DTMF] context-sensitive soft-button.

### DTMF keypad

<table>
<thead>
<tr>
<th>City</th>
<th>Bank</th>
<th>Ant</th>
<th>Master Card</th>
<th>PIN</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>1</td>
</tr>
<tr>
<td>0</td>
<td>1</td>
<td>0</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>0</td>
<td>2</td>
<td>0</td>
<td>3</td>
<td>3</td>
</tr>
<tr>
<td>0</td>
<td>3</td>
<td>0</td>
<td>4</td>
<td>4</td>
</tr>
<tr>
<td>1</td>
<td>0</td>
<td>0</td>
<td>5</td>
<td>5</td>
</tr>
<tr>
<td>1</td>
<td>1</td>
<td>0</td>
<td>6</td>
<td>6</td>
</tr>
<tr>
<td>1</td>
<td>2</td>
<td>0</td>
<td>7</td>
<td>7</td>
</tr>
<tr>
<td>1</td>
<td>3</td>
<td>0</td>
<td>8</td>
<td>8</td>
</tr>
<tr>
<td>2</td>
<td>0</td>
<td>0</td>
<td>9</td>
<td>9</td>
</tr>
</tbody>
</table>

### Command interface

<table>
<thead>
<tr>
<th>Dial Add Del Back</th>
</tr>
</thead>
<tbody>
<tr>
<td>[Dial]: Dial selected DTMF string.</td>
</tr>
<tr>
<td>[Add]: Add new DTMF entry.</td>
</tr>
<tr>
<td>[Del]: Remove selected entry list</td>
</tr>
<tr>
<td>[Back]: Return.</td>
</tr>
<tr>
<td>【・】: Edit selected entry.</td>
</tr>
</tbody>
</table>
**8.3.1 Inband and Outband (RFC2833 and SIP INFO)**

Depending on the configuration, such DTMF sequence may be transmitted to the peer either inbandly (DTMF tones are mixed with normal voice and transmitted over RTP stream) or outbandly (rfc2833 or INFO).

**8.3.1.1 DTMF Relay by SIP INFO**

YV3 supports DTMF relay over SIP signaling channel by INFO method (RFC2976) in addition to DTMF over RTP (by either RFC2833 or mixed with normal voice stream). Once enabled, all DTMF keys, 0-9*#&, will be sent by SIP INFO method; otherwise they will be transmitted by DTMF over RTP.

```
INFO sip:3101@SIP.isp.com SIP/2.0
Via: SIP/2.0/UDP 192.168.3.51:5060;branch=abc7801
From: 7751 <sip:7751@SIP.isp.com>;tag=22516
```
Default is disabled.

To enforce DTMF generated during after call setup to be transmitted over SIP signaling channel by INFO method, you need to enable "9. Advanced" / "CODEC" / "DTMF Relay by INFO".

1. Preferences
2. Packetization
3. Comfort Noise
4. RFC 2833 Payload Type: 101
5. DTMF Relay by INFO

- [On] / [Off]: Toggle between enable and disable this feature. Show check symbol ‘√’ in-line if enabled!
- [Back]: Return.
- 【⋀】 and 【⋁】: Scroll menu items.
- 【↲】: Return.

Alternatively, you may configure it from web page, go to "YV3" / "Advanced" / "CODEC" => "DTMF Relay by INFO":

### 8.3.1.2 DTMF over RTP (RFC2833)

If “DTMF Relay by SIP INFO” is disabled, the method of DTMF sequences generated in conversation is negotiated as per RFC3264 (Offer/Answer Model) during call setup phase. You can specify the RTP payload type
for the transmission of out-of-band DTMF over RTP as per RFC2833 in "Advanced" / "CODEC" / "RFC2833 PT":

<table>
<thead>
<tr>
<th>Payload Type:</th>
</tr>
</thead>
<tbody>
<tr>
<td>1 0 1</td>
</tr>
</tbody>
</table>

- [Del]: Delete one character
- [Back]: Return without any changes.
- [>] : Save changes and return.
- [<] and [> ] : move cursor one position in edit mode.

Valid value MUST be between 96 and 127 in decimal. Specify 128 to disable RFC2833 and transmit DTMF inbandly (i.e., mixing the DTMF tones with voice and transmitted in audio RTP stream).
Default is 101.

Alternatively, you may configure it from web page, go to YV3 / Advanced / CODEC => "RFC2833 Payload Type":

8.4 Blocked Calls

YV3 can add entries on address book into blocking list and filter out calls from those parties. On receiving calls from those parties, YV3 will drop them silently with a response “480 Temporarily unavailable”. Currently the max blocking list capacity is 100.

To check blocking list:

a. Press 【】 to activate menu.
b. Go to submenu “1. Address book” / “Call Screen”. It will show call screen list in alphanumeric order and you may use keypad to jump to the first contact prefixed with the entered alphanumeric character (or use navigation key 【】 and 【】 to scroll contact list).
• **[Call]**: Dial to selected contact. *Alternatively, You may lift the handset (offhook) or turn speaker on by pressing \( \text{【} \text{□} \text{】} \) to make a call to selected contact as well.*

• **[Unblk]**: Remove selected contact from black list and return (Unblock, revoke).

• **[Add]**: Add a contact from address book into black list.

<p>| | | | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Call</td>
<td>Unblk</td>
<td>Add</td>
<td>Back</td>
</tr>
</tbody>
</table>

<p>| | | | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

- [Back]: return without any changes.
- 【\( \text{□} \)】: Add selected contact into black list.
- 【\( \text{∧} \)】 and 【\( \text{∨} \)】: Navigate through address book.

• **[Back]**: Return.

• 【\( \text{∧} \)】 and 【\( \text{∨} \)】: Navigate through black list.

• 【\( \text{↓} \)】: Check selected record (Read only)

```
1 4 : 3 7 : 2 1 1 0 / 2 9 / 2 0 0 6
At t e m p t s : 2 0 1
A o R : D e v i l W e a r i n g P r a d
a < s i p : b o s s @ I S P . f o o . c o m
>```

<p>| | | | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

- Show
  - last time-of-call
  - Accumulated call attempts originated from this contact.
  - Address-of-record.
  - **[Call]**: Dial to this contact. *Alternatively, You may lift the handset (offhook) or turn speaker on by pressing \( \text{【} \text{□} \text{】} \) to make a call to selected contact as well.*
  - **[Unblk]**: Remove this contact from black list (Unblock, revoke).
  - **[Back]**: Return.
  - 【\( \text{∧} \)】 and 【\( \text{∨} \)】: Scroll Line.

Alternatively, you may point your web browser from you PC to the phone, then go to \( YV3 \) / \( \text{Address Book} \) and click on the contact which you want to block out.
Then click “Add to Call Screen” to add this contact into black list.

Besides, You may point your web browser from your PC to the phone, then go to `YV3 / Address Book` / `Call Screen` to view and remove contacts from screening list.
8.5 Auto-Answer

If auto-answer feature is enabled when an incoming call arrives, the phone-set will play distinguished auditable tone shortly, then answer the call by turning on speaker-phone (or headset if it is enabled) on idle mode; otherwise proceed as a normal incoming call.

By idle, it means the phone-set is either engaging in no calls or putting a call on hold.

YV3 implements various ways to support auto-answer features:

i. Server-side invoked auto-answer for urgent calls:
   Use of a private header “P-Auto-answer” in the initial INVITE message for the server-side invoked auto-answer extension:
   a. *P-Auto-answer: normal*
      - Respond a “486 Busy Here” if all lines are occupied.
      - Otherwise, auto-answer this incoming call.
   b. *P-Auto-answer: urgent*
      - Dropped an inactive call if all lines are occupied.
      - Put any on-going calls to hold and auto-switch to an available line if not idle.
      - auto-answer the newly arrived call.
   c. *P-Auto-answer: imperious*
      Identical to “P-Auto-answer: urgent” except that the phone-set will auto-answer this call even the do-not-disturb (DND) or all-calls-forward feature has been activated.
   d. *P-Auto-answer: silent*
      Identical to “P-Auto-answer: imperious” except that the phone-set will not play distinguished auditable tone on answering this incoming call. This is for supervised monitor/announce features.

Note, the header, “P-Auto-Answer”, and its keyword, “normal”, “urgent” or “imperious”, are all case-insensitive.

ii. Phone-set locally configured to auto-answer all incoming calls unconditionally (suitable for attendant in call center service).
   a. Unconditionally auto-answer all incoming calls on idle mode can be achieved by either configuring the global phone-set setting and/or map the auto-answer feature as a programmable key.
   b. To configure the global phone-set setting for auto-answer:
      - Press 【 】 to activate menu.
      - Go to submenu 『 5. Preferences / 『 Auto-Answer 』.

[76/117]
2. Control List

- [On] / [Off]: Toggle between enable and disable this feature. Show check symbol ‘√’ in-line if enabled!
- [Back]: Return
- 【↲】: Edit control list if selected otherwise return.

System default is disabled
c. Alternatively, you may configure this system-wide feature from web page „YV3“ / „Preferences“ => “Auto Answer”.
d. If you have mapped “Auto-Answer” (which is “Auto-Ans” if you have programmed “Auto-Answer” as one of the 4 hardware programmable keys on upper half of the LCD panel) as one for the programmable features, you may press [DSS] context-sensitive soft-button to execute this command to toggle auto-answer mode. Once enabled, the corresponding red LED of mapped DSS key will be on, and the phone will auto-answer all incoming calls arrive on idle mode. Besides, unmap this DSS feature will NOT turn off auto-answer configuration.

iii. Auto-Answer incoming calls arrive on specific registered SIP address-of-records.
For each service domain, you can configure to auto-answer all incoming calls to this SIP account (<To> header in the initial INVITE message). This is particularly useful for IP-PBX to implement broadcast feature. Alternatively this feature is also useful when the user have 2 SIP accounts (or more), one for public and one for private use only, and he/she wants to auto-answer all calls to his/her private account.
To configure the account-specific auto-answer feature:
- Press 【Go】 to activate menu.
● Go to submenu 『7.SIP Settings』 / 『Service Domain』 / 『N-th Realm』 / 『Auto-Answer』.

1. Activation
2. Authentication
3. Address-of-Record
4. Proxy Server
5. Registrar Server
6. Auto-Answer
7. Keep NAT Alive

Default is disabled.

● Alternatively, you may configure this account-specific feature from web page 『YV3』 / 『SIP Settings』 / 『N-th Domain』 => “Auto Answer”.

a. Once enabled, all calls destined to this specific service account will be auto-answered on idle mode.

b. This works even when the system-wide auto-answering is off.

● auto-answer\(^{(1)}\) operation:
  i. On idle mode, the phone-set will play a distinguished audible tone shortly, then answer the call by turning on speaker-phone (or headset if applicable). By idle, it either takes no call or is putting a call on hold.
  ii. Otherwise, proceed as normal incoming calls

● Incoming call processing rules (by precedence):
  i. If “silent” or “Imperious” is specified, then auto-answer it. Besides, if “silent” is specified, then no “distinguished audible tone” will be played.
  ii. If the phone is engaged in do-not-disturb mode, then DND wins
  iii. Otherwise, if the phone has turned on unconditionally forward feature, then all incoming calls are forwarded.

---

\(1\) auto-answer operation:

i. On idle mode, the phone-set will play a distinguished audible tone shortly, then answer the call by turning on speaker-phone (or headset if applicable). By idle, it either takes no call or is putting a call on hold.

ii. Otherwise, proceed as normal incoming calls
iv. Check for auto-answer feature:
   a. Check for server-side auto-answer feature.
   b. Check for the global switch for auto-answer feature.
   c. Match this call against all enabled service domains for auto-answering feature, if a match is located, auto-answer it.

v. Otherwise, proceed as normal incoming calls.

### 8.6 Hot Line

This terminal supports hot-line: dial to a specific number immediately whenever user turns on the speaker phone or picks up handset.

For example, if you enable this feature, and specify the number: “sip:888@ISP.com” or “888@ISP.com” or “888”, then the terminal will dial to the specified number whenever user turns on the speaker phone or picks up handset without waiting for user input.

To configure hot line:

a. Press 【】 to activate menu.
b. Go to submenu “5. Preferences” / “Dial Plan” / “Hot Line”:

<table>
<thead>
<tr>
<th>On/Off</th>
<th>AoR</th>
<th>Back</th>
</tr>
</thead>
<tbody>
<tr>
<td>[On]/[Off]: Toggle between enable and disable this feature. Show check symbol ‘√’ in-line if enabled!</td>
<td></td>
<td></td>
</tr>
<tr>
<td>[Back]: Return.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>[AoR]: Specify the number to dial out whenever the user hooks off.</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

   Address - off - Record:

<table>
<thead>
<tr>
<th>Del</th>
<th>Abc../123..</th>
<th>Clear</th>
<th>Back</th>
</tr>
</thead>
<tbody>
<tr>
<td>[Del]: Delete one character.</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>[Abc../123..]: Toggle between digits and alphanumeric input, where. [Abc..] indicates current input method is alphanumeric and [123..] indicates digits input.</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>[Clear]: Clear all input.</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>[Back]: Return without any changes.</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>【】: Save changes and return.</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>【&lt;】 and 【&gt;】: move cursor one position in edit mode.</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>【▲】 and 【▼】: move cursor per line in edit mode.</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Alternatively, you may go to web page 『YV3』/『Preferences』 / “Preferences” => “Hot Line” to configure it:
9 Service

9.1 Message Waiting Indication (MWI)

Voice mail allows you to access messages left by callers when you are unavailable to take their calls. Voice mail is an optional feature configured by your system administrator. Your particular phone setup might not support accessing voice mail in this way.

YV3 supports messages waiting indication as per RFC3842. It accepts NOTIFY messages with or without prior SUBSCRIBE (note, by [http://pentest.tele-consulting.com/advisories/05_07_06.voip-phones.txt](http://pentest.tele-consulting.com/advisories/05_07_06.voip-phones.txt), there is a weakness of unsolicited NOTIFY for MWI. You may turn off unsolicited MWI NOTIFY, thus effectively disable “Out-of-Dial MWI Notification”).

**MWI Subscription**

If you have configured the voice mailbox link, YV3 will SUBSCRIBE to this link by sending the SUBSCRIBE message to SIP server. Please refer to the following message flow (suppose your SIP AoR is sip:7700@SIP.isp.com and the voice mailbox you configured is sip:vms7700@SIP.isp.com):

```plaintext
SUBSCRIBE sip:vms7700@SIP.isp.com SIP/2.0
Via: SIP/2.0/UDP 192.168.3.50
From: John <sip:7700@SIP.isp.com>;tag=17542c1
To: John <sip:7700@SIP.isp.com>
Call-ID: 0c1c7a67461@ipr.SIP.isp.com
CSeq: 281 SUBSCRIBE
Contact: sip:7700@192.168.3.50
Event: message-summary
Accept: application/simple-message-summary
Expires: 3600
User-Agent: SIP-Phone /1.1
Content-Length: 0

SIP/2.0 202 Accepted
Via: SIP/2.0/UDP 192.168.3.50
Record-Route: <sip:192.168.3.1;lr=1>
Contact: sip:192.168.3.1:6060
Expires: 3600
User-Agent: ISP Soft-Switch/5.1.0
To: John <sip:7700@SIP.isp.com>;tag=980765
From: John <sip:7700@SIP.isp.com>;tag=17542c1
Call-ID: 0c1c7a67461@ipr.SIP.isp.com
CSeq: 281 SUBSCRIBE
Content-Length: 0
```

**Out-of-Dial MWI Notification**

YV3 accepts unsolicited MWI NOTIFY messages as well. The sample NOTIFY message is as follows (suppose “sip.isp.com” is one of your active service domains and your SIP AoR is sip:1234@sip.ISP.com):

```plaintext
NOTIFY sip:1234@192.168.3.50 SIP/2.0
Via: SIP/2.0/udp 192.168.0.1;branch=z9hG4bKcfb4
Content-Type: application/simple-message-summary
Contact: sip:192.168.0.1:6060
User-Agent: ISP Soft-Switch
Event: message-summary
Subscription-State: active
To: <sip:1234@sip.isp.com>
From: <sip:1234@192.168.0.1>;tag=d8370cb
Call-ID: d07b59da8e
CSeq: 224493566 NOTIFY
Content-Length: 39
Max-Forwards: 70
```

[81/117]
9.1.1 Set Up Voice Mailbox URI

To set up voice mail access number by TELNET or keypad:

a. Press 【】 key to activate menu.

b. Go to submenu “8. Service” / “MWI” / “Voice Mailbox AoR”

```
Voicemail Mailbox AoR
sip: vms@isp.foot.com
```

- [Call]: Dial to voice mailbox address-of-record if available. Alternatively, You may lift the handset (offhook) or turn speaker on by pressing 【】 to make a call to selected contact as well.
- [Reset]: Reset voice mailbox address-of-record from address book.

```
Michael
Mike
Nick
Patrick
Paul
Richard
```

- [Back]: Return without any changes.
- 【】: Pick selected contact.
- 【▲】 and 【▼】: Navigate through the list.
- [Del]: Remove currently configured voice mailbox address-of-record (no MWI subscription).
- [Back]: Save changes and return.
- 【↲】: Return.

To set up voice mail access number from web browser, go to 「YV3」 / 「Preferences」 / 「Voice Mailbox」, then click the “Contacts” on the right panel to pick an entry from your address book.
9.1.2 Access Voice Mailbox

To access voice mail, press the [ ]] button and follow the voice instructions. The red LED adjunct to the [ ]] button also will be flashing whenever you have unread and/or new voice messages (Message Waiting Indication) in your voice mailboxes.

Note, when the MWI LED is off while pressing [ ], it will dial to the voice mailbox your configured. However, when the MWI LED is flashing and you press [ ], it will make a call to the Message-Account stipulated on the latest NOTIFY message it received (if this field is absent or is not a SIP AoR, the AoR in request is used instead). If there are unsolicited out-of-dialog NOTIFY messages received from different service domains, those voice mailboxes will be called in turn (in circular fashion) each time [ ] is pressed.

9.2 Time Synchronization

You could synchronize the time on your phone with external time server by Simple Network Time Protocol, SNTP.

- To configure NTP / SNTP server on your phone:
  a. Press [Menu] key to activate menu.
  b. Go to submenu “8. Service” / “SNTP”.

<table>
<thead>
<tr>
<th>1. Mode</th>
<th>2. Server</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
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<tr>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>

- [Back]: Return.
- [ ]: Edit selected item
- [ ] and [ ]: Scroll menu items.

c. Set up SNTP mode to meet your demand.

<table>
<thead>
<tr>
<th>1. Unicast</th>
<th>2. Multicast</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
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<td></td>
<td></td>
</tr>
</tbody>
</table>

- [Back]: Return.
- [ ]: Save changes and return.
- [ ] and [ ]: Scroll menu items.

- To configure NTP / SNTP from web browser, go to \[ YV3 \] / \[ Terminal Settings \] / \[ Date / Time \]
The Simple Network Time Protocol is used to synchronize time with YV3. If you set SNTP server to Anycast mode, the phone will send SNTP query to LAN broadcast address. Otherwise, it sends a request to the specified SNTP / NTP server, extracting the reported time from the reply, and overwrites the phone’s time. Typically, SNTP / NTP servers operating in broadcast mode send update messages every 64 to 1024 seconds. The default time on system starting up is 00:00, January 1, 2007, GMT.

<table>
<thead>
<tr>
<th>SNTP</th>
<th>Unicast</th>
<th>Multicast</th>
<th>Anycast</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sends</td>
<td>SNTP request to the specified SNTP server if available.</td>
<td>Nothing</td>
<td>SNTP packet to the local network broadcast address, 224.0.1.1. After the first SNTP response is received, the phone switches to unicast mode with the server being set as the one who first responded.</td>
</tr>
<tr>
<td>Receives</td>
<td>SNTP response from the SNTP server and ignores responses from other SNTP servers.</td>
<td>SNTP data via the SNTP / NTP multicast address from the local network broadcast address from any server on the network. The default multicast address is 224.0.1.1.</td>
<td>Unicast SNTP data from the SNTP server that first responded to the network broadcast request.</td>
</tr>
</tbody>
</table>

By “SIP Register”, this terminal will synchronize its local time based on the “Date” header present in successful 2xx SIP responses to REGISTER method. Per rfc3261, it restricts the time zone in SIP-date to “GMT”, in a format similar to (case-sensitive)

**Date: Sat, 13 Nov 2010 23:29:00 GMT**

As a result, the parsed time will be converted to local time based on time-zone and daylight saving time (DST) adjustment as well (refer to “3.Terminal Settings” / “Date/Time”).

By “SIP Register”, this terminal will synchronize its local time based on the “Date” header present in successful 2xx SIP responses to REGISTER method. Per rfc3261, it restricts the time zone in SIP-date to “GMT”, in a format similar to (case-sensitive)

**Date: Sat, 13 Nov 2010 23:29:00 GMT**

As a result, the parsed time will be converted to local time based on time-zone and daylight saving time (DST) adjustment as well (refer to “3.Terminal Settings” / “Date/Time”).
Assign the IP of NTP/SNTP server.

```
S N T P S e r v e r :
c l o c k . p s u . e d u
```

You can use either a dotted IP address or a DNS name. For available SNTP/NTP servers near your location, please refer to Appendix C – “Available NTP Servers” on “YV3 Administration”.

Note: If you enable DHCP, the NTP server may be acquired by DHCP option 42.

### 9.3 Instant Message

YV3 supports instant short message (255 characters). Its operation is very similar to the short message on modern mobile phones.

- To take advantage of instant messaging by TELNET or keypad:
  a. Press 【X】 key to activate menu.
  b. Go to submenu “8. Service” / “Message”.

```
1. W r i t e M e s s a g e
2. I n b o x
3. O u t b o x
4. D r a f t
```

- 【Back】: Return.
- 【↓】: Enter submenu.
- 【↑】 and 【↓】: Scroll menu items.

- To take advantage of instant messaging by web browser, go to “Instant Message” page.

Please refer “Instant Message” on “YV3 Web Administration” for detail.

By executing “Message” command from [DSS] context-sensitive soft-button, you will get into INBOX.
directly whenever there are new (unread) messages in your INBOX; otherwise, it will bring you into 『Service』/『Message』 menu.

Besides, there will be a status indicator to remind you that there are new (unread) messages in the INBOX.

1. Favorite
2. IMPP
3. AddrBook
4 SIP Realm

Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:
[A/B] + Indicator( ▲▼▲▼) + Caller-ID

T U e 0 9 - 2 6
7 7 5 4

A B ☐ AB

Status Indicator (☒ ☐ ☐ ☐ ☐)

Note, YV3 supports “flashing short message” to display received out-of-dialog message on LCD without user interaction (or any alert) for 2~3 seconds. Such flashing messages are for server-side notification, and they will not be saved (thus “flashing”). To activate such feature, the received out-of-dialog instant message must carry a proprietary header “P-Flash-SMS: on” (case-insensitive) in received MESSAGE messages.

P-Flash-SMS

1 MESSAGE
P-Flash-SMS: on

Content-Length: 17
Content-Type: text/plain
…(content)

2 200 OK

<table>
<thead>
<tr>
<th>#</th>
<th>Content</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>MESSAGE sip:<a href="mailto:7741@ISP.com">7741@ISP.com</a> SIP/2.0</td>
</tr>
<tr>
<td></td>
<td>Via: SIP/2.0/UDP 192.168.3.55:5060;rport;branch=z9hG4bK45d4b54</td>
</tr>
<tr>
<td></td>
<td>From: “Billing”<a href="">sip:billing@ISP.com</a>;tag=452d3453</td>
</tr>
<tr>
<td></td>
<td>To: “Call Shop”<a href="">sip:7741@ISP.com</a></td>
</tr>
<tr>
<td></td>
<td>Call-ID: 034fe184-ad961d26-b57b05e8-9d766d81@192.168.3.55</td>
</tr>
<tr>
<td></td>
<td>CSeq: 22585 MESSAGE</td>
</tr>
<tr>
<td></td>
<td>User-Agent: Billing-AS/1.0</td>
</tr>
<tr>
<td></td>
<td>Max-Forwards: 70</td>
</tr>
<tr>
<td></td>
<td>Contact: &quot;Billing&quot;<a href="">sip:billing@192.168.3.55:5060</a></td>
</tr>
<tr>
<td></td>
<td>Content-Length: 17</td>
</tr>
<tr>
<td></td>
<td>Content-Type: text/plain</td>
</tr>
<tr>
<td></td>
<td>P-Flash-SMS: on</td>
</tr>
<tr>
<td></td>
<td>Balance $9.99</td>
</tr>
</tbody>
</table>
On receiving this out-of-dialog SIP MESSAGE, YV3 will show the following message content on LCD for 2~3 seconds then restore to idle state.

<table>
<thead>
<tr>
<th>Balance</th>
<th>$9.99</th>
</tr>
</thead>
</table>

---

SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.3.55:5060;rport=5060;branch=z9hG4bK45d4b54;received=192.168.3.55
From: “Billing”<sip:billing@ISP.com>;tag=452d3453
To: “Call Shop”<sip:7741@ISP.com>;tag=1bd9d1a
Call-ID: 034fe184-ad961d26-b57b05c8-9d766d81@192.168.3.55
CSeq: 22585 MESSAGE
Server: YV3/0.9.10
Contact: "Call Shop"<sip:7741@192.168.3.41:5060>
Content-Length: 0
10 Multi-Domains Registration

You could register to multiple domains simultaneously such that you could receive calls from those registered domains and make calls to users belonging to different realm. For example, you may have several SIP addresses from different ISPs. Your configuration may be similar to the following:

To configure SIP service domains, you may go to \[7. SIP Settings\] / "Service Domain" / "N-th Domain".

Alternatively, you may configure SIP service domains by a web browser (page: \[YV3\] / [SIP Settings] / [N-th Domain] :
By default, you can gain access to active Service-Domain-N by line-N whenever applicable. That is, the default service domain of [B Call] is 2nd active service domain if available; otherwise, it will use the 1st active service domain as default.

Moreover, you could change the target service domain while making calls by pressing [Realm] context-sensitive soft-button to circle those active service domains (no matter whether they have registered or not).

The active service domain will appear on the upper-right corner.
In addition to those service domains, you could also specify the call as “Auto-locating”. That is, if you have registered 2 service domains, you have 3 items to pick from:

a. foo.net  

```
[A]-----------------------------------
foo.net
```

b. private.ISP.biz  

```
[A]-----------------------------------
private.ISP.biz
```

c. Auto-locate  

```
[A]-----------------------------------
Auto-locate
```

To check register status of each activated SIP service domains, you may execute “SIP Domain Status” (which is “SIP Realm” if mapped to one of the 4 hardware programmable keys on upper half of the LCD) command from [DSS] context-sensitive soft-button to invoke programmable features:

1. √ Answer
2. √ Delay by INFO
3. DN
4. √ SIP Domain Status
5. Reject
6. Call Detail
7. Network Info

Show a symbol indicates:

a. ‘√’ to indicate *successfully registered to all active SIP service domains.*
b. ‘x’ to indicate *none of the active domains are registered.*
c. No symbol to indicate registered to any of the active SIP service domain succeeded.

If pressed (activated), it will show registration status of each active SIP service domain.

```
1 √ ISP . foo . net
2 x ISP . ISP . com
3 . ISP . private . biz
```

Show symbol preceding each active SIP service domain:

a. ‘√’ to indicate a register success state
b. ‘x’ to indicate a register failed state
c. ‘.’ To indicate a registering state.

Alternatively, you may point a web browser to your terminal and check the register status of each domain after signing in:
### Service Domain Status

<table>
<thead>
<tr>
<th>Domain</th>
<th>Status</th>
<th>IP Address</th>
</tr>
</thead>
<tbody>
<tr>
<td>1st</td>
<td>Registering</td>
<td>7705@<a href="mailto:isp7705@isp.biz">isp7705@isp.biz</a></td>
</tr>
<tr>
<td>2nd</td>
<td>N/A</td>
<td></td>
</tr>
<tr>
<td>3rd</td>
<td>N/A</td>
<td></td>
</tr>
</tbody>
</table>

### Active Network Status

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>LAN Link</td>
<td>Up 100M Half Duplex</td>
</tr>
<tr>
<td>PC Link</td>
<td>Down</td>
</tr>
<tr>
<td>Host IP</td>
<td>192.168.3.95</td>
</tr>
<tr>
<td>Network Mask</td>
<td>255.255.0.0</td>
</tr>
<tr>
<td>MAC</td>
<td>0000C3AB4D10</td>
</tr>
<tr>
<td>Primary DNS Server</td>
<td>192.168.3.253</td>
</tr>
<tr>
<td>Alternate DNS Server</td>
<td></td>
</tr>
</tbody>
</table>

### System Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>CPU</td>
<td>AC495</td>
</tr>
<tr>
<td>DSP</td>
<td>AC495</td>
</tr>
<tr>
<td>Memory</td>
<td>16 MB</td>
</tr>
<tr>
<td>Flash ROM</td>
<td>4 MB</td>
</tr>
<tr>
<td>Serial Number</td>
<td>YV3-RD-095</td>
</tr>
</tbody>
</table>

### Version

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>Product</td>
<td>YV3 v0.9.7</td>
</tr>
<tr>
<td>Application</td>
<td>1.9.7.3</td>
</tr>
<tr>
<td>Driver</td>
<td>1.9.5.1</td>
</tr>
<tr>
<td>Hardware</td>
<td>PDA</td>
</tr>
<tr>
<td>Factory Value</td>
<td>1.9.6.12</td>
</tr>
<tr>
<td>Code Name</td>
<td>YV3</td>
</tr>
</tbody>
</table>
11 Instant Message and Presence Protocol (IMPP)

YV3 supports the following specifications and integrated them as Instant Message and Presence Protocol, IMPP, for rich server features integration:

- **Presence**
  
  Monitor the presence state (availability or reachability to take calls) of a contact or a list of contacts. Refer to RFC3265 for SIP Event Notification, RFC3856 for Presence Event Package and RFC3863/RFC2387/RFC4662/RFC4880 for “application/pidf+xml”, ”MIME Multipart-Related”, “Event Notify for Resource List” and “Rich PIDF” data format in NOTIFY.

<table>
<thead>
<tr>
<th>LCD</th>
<th>Web</th>
<th>Status</th>
<th>Presence Status</th>
<th>RPID Activities</th>
</tr>
</thead>
<tbody>
<tr>
<td>(None)</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>☒</td>
<td>☒</td>
<td>Failed to SUBSCRIBE to the presence state of chosen contact!</td>
<td></td>
<td></td>
</tr>
<tr>
<td>☒</td>
<td>☐</td>
<td>Group (a list of users are received, RFC4662)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>☒</td>
<td>☐</td>
<td>On-Line</td>
<td></td>
<td></td>
</tr>
<tr>
<td>☐</td>
<td>☐</td>
<td>Away</td>
<td></td>
<td>&lt;away /&gt;</td>
</tr>
<tr>
<td>☐</td>
<td>☐</td>
<td>Be Right Back</td>
<td>At least one of the entities of the SUBSCRIBED presence contact is available/on-line (NOTIFY status is “open”)</td>
<td>&lt;breakfast /&gt;, &lt;dinner /&gt;, &lt;meal /&gt;, &lt;worship /&gt; or &lt;other&gt; Be Right Back/other&gt;</td>
</tr>
<tr>
<td>☐</td>
<td>☐</td>
<td>Busy</td>
<td></td>
<td>&lt;appointment /&gt;, &lt;busy /&gt;, &lt;meeting /&gt;, &lt;performance /&gt;, &lt;presentation /&gt;, &lt;steering /&gt; or &lt;working /&gt;</td>
</tr>
<tr>
<td>☐</td>
<td>☐</td>
<td>On the Phone</td>
<td></td>
<td>&lt;on-the-phone /&gt;</td>
</tr>
<tr>
<td>☐</td>
<td>☐</td>
<td>Out to Lunch</td>
<td></td>
<td>&lt;out-to-lunch /&gt;</td>
</tr>
<tr>
<td>☐</td>
<td>☐</td>
<td>Off-Line</td>
<td>All entities of the SUBSCRIBED presence contact number are unavailable/off-line (NOTIFY status is “closed”).</td>
<td>Don’t care.</td>
</tr>
</tbody>
</table>

- **Busy Lamp fileld (BLF)**

  This feature is to monitor the status of a contact or a list of contacts, including early (ringing)/confirmed (conversation)/terminated (idle) states. Refer to RFC3865 and RFC4235 for detail. Besides, please refer to RFC2387 for Multipart-Related MIME data type and RFC4662 for “Event Notify for Resource List”.

<table>
<thead>
<tr>
<th>LCD</th>
<th>Web</th>
<th>State</th>
<th>Dialog-Info Status</th>
<th>Note</th>
</tr>
</thead>
<tbody>
<tr>
<td>(None)</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>☒</td>
<td>☒</td>
<td>SUBSCRIBE failed</td>
<td>Failed to SUBSCRIBE to the dialog state of chosen contact!</td>
<td></td>
</tr>
<tr>
<td>☒</td>
<td>☐</td>
<td>Group (a list of users are received, RFC4662)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>☐</td>
<td>☐</td>
<td>Idle</td>
<td>Terminated (no-dialogs)</td>
<td>SUBSCRIBE to the dialog state of chosen contact succeeded and in idle mode.</td>
</tr>
</tbody>
</table>
| ☒     | ☐     | Busy     | confirmed, trying or preceeding | the SUBSCRIBED contact number is busy.  
  o One of the dialogs of any dialog-info entity is in confirmed | trying | proceeding state.  
  o User off-hook is in trying state. |
| ☒     | ☐     | Ringing  | early              | the SUBSCRIBED contact number is ringing.  
  o Not busy  
  o And one of the dialogs of dialog-info entity is in early state. |
• Shared-Line Appearance (SLA), a.k.a. Shared Call Appearance (SCA), Bridged Line Appearance (BLA) or Bridged Call Appearance (BCA)  

<table>
<thead>
<tr>
<th>Icon</th>
<th>Web</th>
<th>SLA Status</th>
<th>Line Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>(None)</td>
<td></td>
<td>SUBSCRIBE to the call-info state of chosen contact succeeded but has not been NOTIFY yet.</td>
<td></td>
</tr>
<tr>
<td>✗</td>
<td>✗</td>
<td>SUBSCRIBE to the chosen shared line failed!</td>
<td></td>
</tr>
<tr>
<td>⬠</td>
<td>⬠</td>
<td>Idle</td>
<td>this appearance of chosen shared line is idle</td>
</tr>
</tbody>
</table>
| ⬁ | ⬁ | Busy | this appearance of chosen shared line is:  
- Seized  
- Progressing: making outbound calls.  
- Active  
- Held-private: be Held and only can be retrieved by the endpoint which held it involved in this call. |
| ✗ | ⬠ | Ringing | this appearance of chosen shared line is:  
- Alerting: receiving an incoming call. |
| ⬁ | ✗ | Held | this appearance of chosen shared line is:  
- Held: Holding the peer and can be retrieved by other endpoints. |

The Shared Line Appearance (SCA), a.k.a. Shared Call Appearance (SCA), Bridged Line Appearance (BLA), or Bridged Call Appearance (BCA), feature allows the administrator to add multiple locations to a given line. Any of the locations can be used to originate or receive calls.

When a call comes in to an idle line, all the provisioned locations for that line are alerted. The first location to answer the call is connected to the originator. If the line is already active in a call, only the active location is alerted.

A subscriber can originate calls from any of the configured locations. All other locations are unable to originate calls until all calls are released.

It is recommended that the phone number plus an index (<phoneNumber>_<index>) is used when provisioning the unique address of record (AoR) for each shared line. For example: 2405511111_2. If a phone number does not exist, the MAC address plus an index could be used (<macAddress>_<index>).

For example in the configuration below:

Bob and Joe each have two lines and that Bob shares a line with Joe and Joe shares a line with Bob. The
The figure also shows the applicable Subscriber Profile and Shared Line Appearance Configuration data for subscribers Bob and Joe.

When Bob (2405551111) is called, Bob’s first line and Joe’s second line will ring. When Joe (2405552222) is called, Joe’s first line and Bob’s second line will ring.

Please refer to “BROADWORKS SIP ACCESS SIDE EXTENSIONS INTERFACE SPECIFICATIONS” release 13.0 version 1 from BroadSoft Inc. for detailed implementation.

If you make an outbound call by click [Call] to this “Shared-Line Appearance” contact, it will:

1. Gain an empty channel 【A】/【B】 and perform line-seize SUBSCRIBE-NOTIFY transaction before sending INVITE.

2. On receiving INVITE with Call-Info and “answer-after” parameter present, such as:
   
   Call-Info: <sip:ProxyDNSorIP.com>;appearance-index=3;answer-after=0

   ◆ Auto Answer this call if
      • “answer-after” parameter is present in Call-Info header
      • From Header is a recognized Shared-Line AoR
      • appearance-index is configured
   ◆ Mapping Rule
      • answer-after=0: silent
      • answer-after=1: imperious
      • answer-after<3: urgent
      • Others and present: normal

### 11.1 Terminal Presence Status

You can configure the Presence state of this terminal as per RFC4480 (Rich Presence Extension to the Presence Information Data Format). YV3 will indicate user’s status in status line during idle as in:

<table>
<thead>
<tr>
<th></th>
<th>Favorite</th>
<th>Register</th>
</tr>
</thead>
<tbody>
<tr>
<td>* IMPP</td>
<td></td>
<td></td>
</tr>
<tr>
<td>* DTMF</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

* Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:
To configure the Presence status of your own terminal, press 【 】 to activate menu mode and go to “1.Address Book” / “IMPP” => [State] context-sensitive soft-button (alternatively, you may execute “My Presence Status” (acronym “My Status”  on 4 hardware programmable keys if mapped) command from [DSS] programmable features):

- [State]: Set terminal’s state.
- [Back]: Return.
- 【↲】: Enter sub-menu.
- 【⋀】 and 【⋁】: Navigate through menu items.

### Terminal Presence Status indicator:

<table>
<thead>
<tr>
<th>Icon</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>N/A</td>
<td>On-Line (idle)</td>
</tr>
<tr>
<td>✈</td>
<td>Off-Line</td>
</tr>
<tr>
<td>☯</td>
<td>Busy / On-the-Phone</td>
</tr>
<tr>
<td>☯</td>
<td>Away / Be-Right-Back / Out-to-Lunch</td>
</tr>
</tbody>
</table>

### State

<table>
<thead>
<tr>
<th>State</th>
<th>Back</th>
</tr>
</thead>
<tbody>
<tr>
<td>On-Line</td>
<td></td>
</tr>
<tr>
<td>Away</td>
<td></td>
</tr>
<tr>
<td>Be Right Back</td>
<td></td>
</tr>
<tr>
<td>Busy</td>
<td></td>
</tr>
<tr>
<td>On the Phone</td>
<td></td>
</tr>
<tr>
<td>Out to Lunch</td>
<td></td>
</tr>
<tr>
<td>Off-Line</td>
<td></td>
</tr>
</tbody>
</table>

- 【↲】 and 【⋀】: Set and return.
- 【⋁】 and 【⋀】: Navigate through menu items.

Note, your presence status will be ignored if it is set to On-Line (idle) state and you have enable the following features (in precedence):

### Terminal Features

<table>
<thead>
<tr>
<th>Terminal Features</th>
<th>Presence Status</th>
<th>RPID Activities</th>
</tr>
</thead>
<tbody>
<tr>
<td>All-Calls-Forward is enabled</td>
<td>open</td>
<td>Away</td>
</tr>
<tr>
<td>Logout of all SIP Service Domains</td>
<td>open</td>
<td>Away</td>
</tr>
<tr>
<td>Do-Not-Disturb (DND) is enabled</td>
<td>open</td>
<td>Busy</td>
</tr>
</tbody>
</table>

Note, your presence status will be ignored if it is set to On-Line (idle) state and you have enable the following features (in precedence):
That is, this terminal will respond a “open” presence state with corresponding RPIDF activity (which is “Away” for All-Calls-Forward and logout, and “Busy” for DND) if ever got Presence SUBSCRIBEd from other terminals.

11.2 IMPP Manipulation by Keypad

You can invoke IMPP list by pressing 【】 => “1.Address Book” / “IMPP”; alternatively, you may press “IMPP” programmable keys (or run it from [DSS] context-sensitive soft-button) to show IMPP list.

<table>
<thead>
<tr>
<th>Bill (off line)</th>
<th>Call</th>
<th>Add</th>
<th>Del</th>
<th>Back</th>
</tr>
</thead>
<tbody>
<tr>
<td>Fox</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Jason (on the phone)</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Maria (active)</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Nick</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Patrick (ringing)</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Richard (Away)</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

- [Call]: Dial to selected contact. Alternatively, You may lift the handset (offhook) or turn speaker on by pressing 【↲】 to make a call to selected contact as well.
- [Add]: Add a new contact into IMPP list.
- [Del]: Remove selected contact from IMPP list
- [Back]: Return.
- 【↲】: review selected IMPP contact.
- 【⋀】 and 【⋁】: Scroll IMPP list.

IMPP contacts will be listed as:
1. Show all subscribed Presence, Busy Lamp Filed (BLF), and Shared-Line Appearance (SLA) status.
2. Show display or user-part of the email-like address whenever possible and listed in alphanumeric order.
3. User may use keypad to jump to the first contact prefixed with entered alphanumeric character. Alternatively, user may use navigation key 【⋀】 and 【⋁】 to scroll contact list.
5. Presence (RPIDF):

<table>
<thead>
<tr>
<th>LCD</th>
<th>Web</th>
<th>Status</th>
<th>Presence Status</th>
<th>RPID Activities</th>
</tr>
</thead>
<tbody>
<tr>
<td><img src="None" alt="None" /></td>
<td></td>
<td></td>
<td>Subscribed OK but not received NOTIFY for presence event yet.</td>
<td></td>
</tr>
<tr>
<td><img src="X" alt="X" /></td>
<td></td>
<td>SUBSCRIBE failed</td>
<td></td>
<td></td>
</tr>
<tr>
<td><img src="X" alt="X" /></td>
<td></td>
<td>Group (a list of users are received, RFC4662)</td>
<td></td>
<td></td>
</tr>
<tr>
<td><img src="On-Line" alt="On-Line" /></td>
<td></td>
<td>open</td>
<td>N/A, &lt;other&gt;idle&lt;/other&gt; or all others.</td>
<td></td>
</tr>
<tr>
<td><img src="Away" alt="Away" /></td>
<td></td>
<td>open</td>
<td>&lt;away /&gt;</td>
<td></td>
</tr>
<tr>
<td>![Be Right Back](Be Right Back)</td>
<td></td>
<td>open</td>
<td>&lt;breakfast /&gt;, &lt;dinner /&gt;, &lt;meal /&gt;, &lt;worship /&gt; or &lt;other&gt;Be Right Back&lt;/other&gt;</td>
<td></td>
</tr>
<tr>
<td><img src="Busy" alt="Busy" /></td>
<td></td>
<td>open</td>
<td>&lt;appointment /&gt;, &lt;busy /&gt;, &lt;meeting /&gt;, &lt;performance /&gt;, &lt;presentation /&gt;, &lt;steering /&gt; or &lt;working /&gt;</td>
<td></td>
</tr>
<tr>
<td>![On the Phone](On the Phone)</td>
<td></td>
<td>open</td>
<td>&lt;on-the-phone /&gt;</td>
<td></td>
</tr>
<tr>
<td>![Out to Lunch](Out to Lunch)</td>
<td></td>
<td>open</td>
<td>&lt;out-to-lunch /&gt;</td>
<td></td>
</tr>
<tr>
<td><img src="Off-Line" alt="Off-Line" /></td>
<td></td>
<td>closed</td>
<td>Don’t care.</td>
<td></td>
</tr>
</tbody>
</table>

6. Busy Lamp Field (Dialog-Info)

<table>
<thead>
<tr>
<th>LCD</th>
<th>Web</th>
<th>State</th>
<th>Aggregated Dialog-Info Status</th>
</tr>
</thead>
<tbody>
<tr>
<td><img src="None" alt="None" /></td>
<td></td>
<td>Subscribed OK but not received NOTIFY for dialog-info event yet.</td>
<td></td>
</tr>
<tr>
<td><img src="X" alt="X" /></td>
<td></td>
<td>SUBSCRIBE failed</td>
<td></td>
</tr>
</tbody>
</table>
7. **Shared Line Appearance (Call-Info)**

<table>
<thead>
<tr>
<th>Icon</th>
<th>Web</th>
<th>Line Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>(None)</td>
<td></td>
<td>Subscribed OK but not received NOTIFY for dialog-info event yet.</td>
</tr>
<tr>
<td><img src="144x766" alt="crossed" /></td>
<td><img src="162x784" alt="crossed" /></td>
<td>SUBSCRIBE failed</td>
</tr>
<tr>
<td><img src="114x753" alt="circle" /></td>
<td><img src="121x763" alt="circle" /></td>
<td>Idle</td>
</tr>
<tr>
<td><img src="114x741" alt="star" /></td>
<td><img src="121x750" alt="star" /></td>
<td>Seized, progressing, active or held-private</td>
</tr>
<tr>
<td><img src="144x722" alt="bell" /></td>
<td><img src="162x752" alt="bell" /></td>
<td>Alerting</td>
</tr>
<tr>
<td><img src="58x554" alt="sun" /></td>
<td><img src="116x78" alt="sun" /></td>
<td>Held</td>
</tr>
</tbody>
</table>

7. **Shared Line Appearance (Call-Info)**

<table>
<thead>
<tr>
<th>Icon</th>
<th>Web</th>
<th>Line Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>(None)</td>
<td>Subscribed OK but not received NOTIFY for dialog-info event yet.</td>
<td></td>
</tr>
<tr>
<td><img src="144x766" alt="crossed" /></td>
<td><img src="162x784" alt="crossed" /></td>
<td>SUBSCRIBE failed</td>
</tr>
<tr>
<td><img src="114x753" alt="circle" /></td>
<td><img src="121x763" alt="circle" /></td>
<td>Idle</td>
</tr>
<tr>
<td><img src="114x741" alt="star" /></td>
<td><img src="121x750" alt="star" /></td>
<td>Seized, progressing, active or held-private</td>
</tr>
<tr>
<td><img src="144x722" alt="bell" /></td>
<td><img src="162x752" alt="bell" /></td>
<td>Alerting</td>
</tr>
<tr>
<td><img src="58x554" alt="sun" /></td>
<td><img src="116x78" alt="sun" /></td>
<td>Held</td>
</tr>
</tbody>
</table>

11.3 **IMPP Manipulation on Web**

If configured by a web browser, point the browser to your terminal and goes to „YV3“, „Address Book“, „IMPP“:

![YV3 - [192.168.3.95] - Microsoft Internet Explorer](image)

IMPP contacts will be listed as:

i. Show all subscribed Presence, Busy Lamp Filed (BLF), and Shared-Line Appearance (SLA) status.
ii. Show display or user-part of the email-like address whenever possible and listed in alphanumeric order.
iii. User may use keypad to jump to the first contact prefixed with entered alphanumeric character. Alternatively, user may use navigation key 【A】and 【V】to scroll contact list.
iv. Max entries: 50.

v. Presence (RPIDF):

<table>
<thead>
<tr>
<th>LCD</th>
<th>Web</th>
<th>Status</th>
<th>Presence Status</th>
<th>RPID Activities</th>
</tr>
</thead>
<tbody>
<tr>
<td>(None)</td>
<td>Subscribed OK but not received NOTIFY for presence event yet.</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td><img src="144x766" alt="crossed" /></td>
<td><img src="162x784" alt="crossed" /></td>
<td>SUBSCRIBE failed</td>
<td></td>
<td></td>
</tr>
<tr>
<td><img src="114x753" alt="circle" /></td>
<td><img src="121x763" alt="circle" /></td>
<td>Group (a list of users are received, RFC4662)</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
### vi. Busy Lamp Field (Dialog-Info)

<table>
<thead>
<tr>
<th>LCD</th>
<th>Web</th>
<th>State</th>
<th>Aggregated Dialog-Info Status</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>N/A, idle</td>
<td>N/A, idle or all others.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>&lt;away /&gt;</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>&lt;breakfast /&gt;, &lt;dinner /&gt;, &lt;meal /&gt;, &lt;worship /&gt; or &lt;other&gt; Be Right Back</td>
<td>&lt;breakfast /&gt;, &lt;dinner /&gt;, &lt;meal /&gt;, &lt;worship /&gt; or &lt;other&gt; Be Right Back</td>
</tr>
<tr>
<td></td>
<td></td>
<td>open</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>confirmed/trying/preceeding</td>
<td>confirmed/trying/preceeding</td>
</tr>
<tr>
<td></td>
<td></td>
<td>terminated/no-dialogs</td>
<td>terminated/no-dialogs</td>
</tr>
<tr>
<td></td>
<td></td>
<td>idle</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>Don’t care.</td>
<td>Don’t care.</td>
</tr>
</tbody>
</table>

### vii. Shared Line Appearance (Call-Info)

<table>
<thead>
<tr>
<th>Icon</th>
<th>Web</th>
<th>Line Status</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>(None)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>N/A, idle</td>
</tr>
<tr>
<td></td>
<td></td>
<td>SUBSCRIBE failed</td>
</tr>
<tr>
<td></td>
<td></td>
<td>SUBSCRIBE failed</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Seized, progressing, active or held-private</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Alerting</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Held</td>
</tr>
</tbody>
</table>

- New: click “New” on the top toolbar to add new IMPP contacts into your address book.
You can add max 5 new contacts into IMPP list each time.
  o Type:
    ▪ Presence
    ▪ Busy Lamp Field
    ▪ Shared-Line Appearance
  o Contact:
    ▪ Focus on the text input field by clicking on it
    ▪ click the interested contact on “Contacts” panel on the right-hand side to select it into current text input.
    ▪ You may click “Clear” button to remove the wrongly selected contact.
  o Add: add and return
  o Cancel: abort operation and return.

- Delete: Remove selected items from IMPP list.
  o Check those contacts you want to delete then click “Delete” on the top toolbar to delete it
  o Delete All:
    ▪ Check the “Select All” button to select all records.
    ▪ Click “Delete” on the top toolbar to remove all selected contacts.

- Call: click-to-call to selected contact.
Check the contact you want to call to then click “Call” on the top toolbar. Then the terminal will make an outbound call to this contact. If the terminal is on hook, then it will enter hands-free mode automatically.

If you make an outbound call by click [Call] to this “Shared-Line Appearance” contact, it will:

i. Gain an empty channel \([A] / [B]\) and perform line-seize SUBSCRIBE-NOTIFY transaction before sending INVITE.

ii. On receiving INVITE with Call-Info and “answer-after” parameter present, such as:

\[
\text{Call-Info: } \text{<sip:ProxyDNSorIP.com>};\text{appearance-index=3;answer-after=0}
\]
◆ Auto Answer this call if
  • “answer-after” parameter is present in Call-Info header
  • From Header is a recognized Shared-Line AoR
  • appearance-index is configured

◆ Mapping Rule
  • answer-after=0: silent
  • answer-after=1: imperious
  • answer-after<3: urgent
  • Others and present: normal

• My Status: Configure the Presence state of this terminal as per RFC4480 (Rich Presence Extension to the Presence Information Data Format). YV3 will indicate user’s status in status line during idle as in:

```
<table>
<thead>
<tr>
<th>Status Indicator</th>
<th>Other Statuses</th>
</tr>
</thead>
<tbody>
<tr>
<td>* IMPP</td>
<td>Favorite</td>
</tr>
<tr>
<td>* DTMF</td>
<td>Register</td>
</tr>
</tbody>
</table>

Screen Popup for call-waiting, holding calls, Conference or Auto-Redial:
[A/B] + Indicator( (!_)  ) + Caller-ID
```

```
Tue 09-26 15:52
7 4: Not Registered
```

Terminal Presence Status indicator:

<table>
<thead>
<tr>
<th>Icon</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>N/A</td>
<td>On-Line (idle)</td>
</tr>
<tr>
<td>8</td>
<td>Off-Line</td>
</tr>
<tr>
<td>🕴</td>
<td>Busy / On-the-Phone</td>
</tr>
<tr>
<td>🕴  📞</td>
<td>Away / Be-Right-Back / Out-to-Lunch</td>
</tr>
</tbody>
</table>

Available Presence status:

<table>
<thead>
<tr>
<th>Icon</th>
<th>My Status</th>
<th>Presence Status</th>
<th>RPID Activities</th>
</tr>
</thead>
<tbody>
<tr>
<td>N/A</td>
<td>On-Line (idle)</td>
<td>open</td>
<td>&lt;other&gt;idle&lt;/other&gt;</td>
</tr>
<tr>
<td>🕴</td>
<td>Away</td>
<td>open</td>
<td>&lt;away /&gt;</td>
</tr>
<tr>
<td>🕴  📞</td>
<td>Be Right Back</td>
<td>open</td>
<td>&lt;other&gt;Be Right Back&lt;/other&gt;</td>
</tr>
<tr>
<td>Presence Status</td>
<td>Status</td>
<td>RPID Activity</td>
<td></td>
</tr>
<tr>
<td>--------------------</td>
<td>---------</td>
<td>---------------</td>
<td></td>
</tr>
<tr>
<td>Busy</td>
<td>open</td>
<td>&lt;busy /&gt;</td>
<td></td>
</tr>
<tr>
<td>On the Phone</td>
<td>open</td>
<td>&lt;on-the-phone /&gt;</td>
<td></td>
</tr>
<tr>
<td>Out to Lunch</td>
<td>open</td>
<td>&lt;out-to-lunch /&gt;</td>
<td></td>
</tr>
<tr>
<td>Off-Line</td>
<td>closed</td>
<td>None</td>
<td></td>
</tr>
</tbody>
</table>

Note, your presence status will be ignored if it is set to On-Line (idle) state and you have enable the following features (in precedence):

<table>
<thead>
<tr>
<th>Terminal Features</th>
<th>Presence Status</th>
<th>RPID Activities</th>
</tr>
</thead>
<tbody>
<tr>
<td>All-Calls-Forward is enabled</td>
<td>open</td>
<td>Away</td>
</tr>
<tr>
<td>Logout of all SIP Service Domains</td>
<td>open</td>
<td>Away</td>
</tr>
<tr>
<td>Do-Not-Disturb (DND) is enabled</td>
<td>open</td>
<td>Busy</td>
</tr>
</tbody>
</table>

That is, this terminal will respond a “open” presence state with corresponding RPIDF activity (which is “Away” for All-Calls-Forward and logout, and “Busy” for DND) if ever got Presence SUBSCRIBEd from other terminals.
12 NAT Traversal

If this terminal locates within a local area network and you want to place a call to public internet, you must configure your terminal to traverse the NAT and firewall it currently behind.

To learn determine whether you resides on public internet or local area network, please click `YV3` to show the current Host IP (alternatively, you may call to “#*47” (“#*IP”) to show terminal’s IP on LCD):

If your host IP is within any of the listed ranges, then your terminal resides on LAN, otherwise, it locates on public internet.

- 10.0.0.0 ~ 10.255.255.255 (10.xxx.xxx.xxx).
- 172.16.0.0 ~ 172.31.255.255 (172.16.xxx.xxx ~ 172.31.xxx.xxx).

Note: if your host IP is 127.0.0.1, you should reconfigure your network to acquire a valid IP.

12.1 Public Internet Configuration
There are basically three options for CPE to traverse NAT and Firewall:

- Option 1: Set up a static route in the NAT gateway (Recommend).
- Option 2: Use STUN to measure out ports.
- Option 3: Use UPnP to dynamically map UDP/TCP ports on UPnP-capable NAT/Firewall/Gateway co-locate on your LAN.

Please adopt suitable option based on your network configuration.

**Note:** some SIP ISPs may provide SIP-aware routers (NAT/Firewall) for their customers. When you use a SIP-aware router, NAT detection should be set to "Off" as if you were on the public internet and the configuration is the same as “Public Internet Configuration”: set [UDP Traversal] to be [Full Access (Public Host IP)], which is to say that you do not have to worry about the NAT, nor firewall.

### 12.2.1 Static NAT Route

If you have access to your NAT/Firewall, you may contact your network administrator to configure your NAT/Firewall route.

- Contact your network administrator and acquire consecutive UDP ports (at least 5 UDP ports in case of YV3) mapped from the NAT to your terminal. For example:
  - NAT IP = 218.81.107.51 (you may detect your NAT IP by clicking [Diagnose NAT])
  - The network administrator has mapped 7 consecutive UDP ports, 45700 ~ 5706, from NAT to your terminal, which the terminal IP is 192.168.3.57.

  **Note 1:** Since the network administrator has to configure the NAT/firewall to map those UDP ports to your terminal statically, thus you should use static IP instead of DHCP as your network configuration. Otherwise you take the risk that the terminal would get a different IP from the currently set into the mapping when it reboot and re-get its IP by DHCP.

  **Note 2:** If there are several terminals reside under the same NAT, their NAT port mappings must not be overlapped since they all share the same NAT resource!

- Configure RTP ports
RTP Port Base: 45700 (Must be an even number and between 2 and 65534).
RTP Port Range: 6 (This value must be an even number, and larger than or equal to 2); for YV3, it should be at least 4 to support two concurrent calls.
Such RTP setting effectively designates 6 consecutive UDP ports ranging from 45700 to 45705 (inclusive) for media transaction.

Note: RTP/RTCP ports are used for media transmission. Those ports must be consecutive and start with an even number. Besides, the total allocated UDP ports SHOULD at least double the max concurrent calls supported by your terminal since each active call takes 2 ports for RTP/RTCP.

- Configure SIP service signaling port:
  Take the scenario above as an example:
Transport: UDP and TCP (or “UDP”, you must include UDP anyway).
- SIP Listen port: 45706

Assign static NAT IP:
- Diagnose NAT (optional): you may detect your NAT IP by clicking 『Set and Diagnose NAT』. Note, this diagnosis utilize STUN server, you must have assigned a valid/viable STUN server first.
- Static NAT IP: Fill in the acquired NAT IP from network administrator, such as 218.81.107.51 mentioned above.
- UDP traversal: Static NAT IP/UDP Map
  Note, if your NAT equipped with no fixed IP, such as those NATs dial into WAN by PPPoE, then you must synchronize the NAT IP currently set into YV3 whenever your NAT IP changed (such as NAT
re-dialup). By default, the system will regularly (every 10 minutes) auto-detect the NAT IP (if the STUN server set a valid and viable server) and notify you on the YV3 status page whenever it determines a mismatch between the static NAT IP you configured and the one it auto-discovered by STUN protocol.

Alternatively, you may enable Auto Detect & Update Static NAT IP by STUN to auto-refresh the newly changed NAT IP into Static NAT IP field whenever it detects an inconsistency.

### 12.2.2 NAT Traversal by STUN

Setting up the NAT router is impossible in many cases, and new equipment may be too expensive. For these environments, "Simple Traversal of UDP through NATs" (STUN) has come to rescue. To enable STUN, you must acquire the IP of your served STUN server and diagnose your NAT to see if STUN is viable.

- **To determine whether you are currently NATed or behind a firewall:**
  - STUN server: configure your STUN server, such as “YourISP.com” (YV3 will try to locate the STUN server of “YourISP.com” domain via DNS SRV query (_stun._udp.YourISP.com) or you may explicitly specify the STUN server as “STUN.YourISP.com” or 61.222.26.124. You can enter both dotted IP and DNS record.
  - Click Set and Diagnose NAT to detect your NAT type. If the diagnosis is one of the following, you could consider enabling STUN as a way to traverse NAT; otherwise you should adopt option 1- Static NAT Route.
    (a) Public internet (neither NAT nor Firewall is involved).
    (b) Symmetric UDP Firewall
    (c) Full cone NAT
    (d) Restricted cone NAT
    (e) Port restricted cone NAT.

**Note:** The NAT bundled with Windows 2000/XP/2003 can NOT be traversed by STUN protocol, even though it may wrongly detected by STUN as Port-restricted NAT.

- **Activate STUN Mode**

![YV3 NAT Traversal by STUN](image)

- STUN server: Enter a functional and reachable STUN server IP for STUN to work.
- UDP Traversal: Enable STUN
12.2.3 NAT Traversal by UPnP

If your terminal is behind an UPnP-capable (Universal Plug & Play) NAT/Gateway/Firewall devices (support either WANIPConnection or WANPPPConnection), you may turn on UPnP, then detect whether there is such UPnP NAT available on your LAN.

By “Set and Diagnose UPnP”, it will search for UPnP devices and show its external IP. If such devices existed, this terminal will open UDP/TCP mapping on NAT device for media streaming and SIP signaling by UPnP as necessary; Otherwise if there is no such devices co-exist (or turn off UPnP explicitly), it may show “Not found UPnP device”. Consult to your LAN administrator for further information.

Note: the UPnP mode must have been enabled before you can diagnose UPnP device on your LAN.
13 Web-Specific System Administration

13.1 Export Personal Data and Configuration

Export terminal configuration data in plaintext. The exported file is as the same as the one got by auto-provision procedure (refer to “Auto-Provision” on YV3 Administration) which can be imported (by upload configuration data or http/tftp auto-provision) again to configure other terminal. You can either download the exported data on PC or uploaded it from your terminal to TFTP/HTTP server by TFTP or HTTP protocol.

- System Configuration
  - Network, VLAN and RTP: export network related settings.
  - SIP Settings and Port User: export SIP-related settings.
  - Others except Mentioned Above

- Personal Data
  - Address Book: export address book contacts.
  - Call Screen: export call screen list.
  - Speed Dials: export public speed dial and port-specific speed dial records.
  - Voice Mailbox: export voice mailboxes entries
  - Call Forward Settings: export call forward settings.
  - Favorite: export favorite list
  - IMPP: export IMPP list
  - DTMF list: export pre-configured DTMF list.

- File Extension:
  Specify the file exported file extension, such as txt. By default the exported filename is the Ethernet MAC address represented in hexadecimal string appended with the file extension. For example, “000D0AC0F9.txt”. Pick “csv” extension for easy procession by Micosoft Excel.
Security:
Specify whether the exported data should be ciphered by Advanced Encryption Standard, AES, of 192-bit key length.
- When security option is checked, you can specify your own encryption key (password) to protect your configuration data. If you leave it as blank, a built-in system default AES 192-bit key will be employed to encrypt the exported data.

Operation Method
- Export:
  Export selected (checked) configuration data as a plain text file and download to PC.
  Specify whether the exported data should be ciphered by Advanced Encryption Standard, AES, of 192-bit key length.
  - When security option is checked, you can specify your own encryption key (password) to protect your configuration data. If you leave it as blank, a built-in system default AES 192-bit key will be employed to encrypt the exported data.
- TFTP Upload:
  Export selected (checked) configuration data as a plain text file and upload the exported file to Auto-provision server specified in YV3/Auto-Provision/Provision Server by TFTP protocol.
- HTTP Upload:
  Export selected (checked) configuration data as a plain text file and upload the exported file to Auto-provision server specified in YV3/Auto-Provision/Provision Server by HTTP protocol.

13.2 System Administration

- Re-REGISTER
  Re-REGISTER all activated SIP address-of-records to SIP service again. If the user has gone off-line (see below) explicitly, restart the auto-registration process to let user go online again.
- Un-REGISTER (Offline)
  Unregister all SIP address-of-records to all activated SIP service immediately and stop the regular...
auto-registration scheduling to keep user offline. The phone will keep unregistered until either the “Register” command is executed by invoking [DSS] programmable features or the “Re-REGISTER” command is issued to go online explicitly. This is useful when you are off work and want all calls go directly to ring your home lines.

Note, reboot the terminal will clear this status and register all activated SIP address-of-records after startup.

- **Reboot:**
  Save all configuration data back to NVRAM and restart (reboot) YV3.

- **Reset System Settings:**
  Restore all configuration data back to factory default values (such as restore both privileged password and user-level password to “0000”), but leaving general network settings, and personal information intact. Those unchanged user data include:
  - **General Network Settings:** Mode, Static Settings, PPPoE Settings, DNS Settings, and IP-TTL.
  - **Address Book**
  - **Speed Dials**
  - **Call Screen**
  - **Favorite List**
  - **IMPP List**
  - **DTMF List**
  - **Call History:** Missed, Received and Dialed Calls.
  - **Instant Messages:** Inbox, Outbox and Draft
  - **Call Statistics**

- **Factory Value:**
  Restore everything, including network settings and personal information, back to factory values, then auto-reboot the terminal.

- **Shut down:**
  Shut down this terminal for power off.
  After shutdown, all indicators (LEDs) and status LED will be off, then you could power off your terminal safely.

- **Logout:** Log out HTTP this session.

### 13.2.1 Issue Commands by HTTP Get

You can remotely issue those command, such as:

- **http://terminal_ip_address/reboot** => reboot this terminal.
- **http://terminal_ip_address/reset** => restore all configuration back to factory values.
- **http://terminal_ip_address/shutdown** => shut down this terminal.
- **http://terminal_ip_address/register** => register to registrars of all activated service domains immediately and re-schedule regular auto-registration as necessary.
- **http://terminal_ip_address/unregister** => un-register to registrars of all activated service domains immediately and cease regular auto-registration scheduling to keep user offline.
- **http://terminal_ip_address/export** => download all configuration data in provision format.

where terminal_ip_address is the IP address of your terminal. Those command web pages are password protected. Once the service provider issues one of these commands, the password page will prompt for authorized password to proceed.
### 13.3 SIP Status Code

Show 20 records of the most recent received and transmitted SIP responses in the format of:

```
"hh:mm:ss Tx/Rx (CSeq Method): Status-Code Reason",
```

where 1xx provisional status codes are in black font, 2xx success codes in blue font, 3xx redirect codes are in blown font, whereas 4xx, 5xx, 6xx error codes are in red font.

This is runtime information and those records will be cleared on startup.

<table>
<thead>
<tr>
<th>Time</th>
<th>Type</th>
<th>Content</th>
</tr>
</thead>
<tbody>
<tr>
<td>05:40:50</td>
<td>Tx (35996)</td>
<td>INVITE; 486 Busy Here</td>
</tr>
<tr>
<td>05:40:49</td>
<td>Tx (35996)</td>
<td>INVITE; 180 Ringing</td>
</tr>
<tr>
<td>05:40:43</td>
<td>Rx (41712)</td>
<td>INVITE; 488 Busy Here</td>
</tr>
<tr>
<td>05:40:39</td>
<td>Rx (41712)</td>
<td>INVITE; 180 Ringing</td>
</tr>
<tr>
<td>05:40:38</td>
<td>Rx (41712)</td>
<td>INVITE; 100 Trying</td>
</tr>
<tr>
<td>05:40:38</td>
<td>Rx (41711)</td>
<td>INVITE; 407 Proxy Authentication Required</td>
</tr>
<tr>
<td>05:40:35</td>
<td>Rx (20832)</td>
<td>SUBSCRIBE; 408 Request Timeout</td>
</tr>
<tr>
<td>05:40:33</td>
<td>Rx (31096)</td>
<td>BYE; 200 OK</td>
</tr>
<tr>
<td>05:40:32</td>
<td>Rx (31095)</td>
<td>INVITE; 200 OK</td>
</tr>
<tr>
<td>05:40:31</td>
<td>Rx (31095)</td>
<td>INVITE; 100 Trying</td>
</tr>
<tr>
<td>05:40:31</td>
<td>Rx (31094)</td>
<td>INVITE; 407 Proxy Authentication Required</td>
</tr>
<tr>
<td>05:40:30</td>
<td>Rx (31093)</td>
<td>INVITE; 200 OK</td>
</tr>
<tr>
<td>05:40:29</td>
<td>Rx (31093)</td>
<td>INVITE; 100 Trying</td>
</tr>
<tr>
<td>05:40:29</td>
<td>Rx (31092)</td>
<td>INVITE; 407 Proxy Authentication Required</td>
</tr>
<tr>
<td>05:40:20</td>
<td>Rx (31091)</td>
<td>INVITE; 200 OK</td>
</tr>
<tr>
<td>05:40:19</td>
<td>Rx (31091)</td>
<td>INVITE; 180 Ringing</td>
</tr>
<tr>
<td>05:40:18</td>
<td>Rx (31091)</td>
<td>INVITE; 100 Trying</td>
</tr>
<tr>
<td>05:40:18</td>
<td>Rx (31090)</td>
<td>INVITE; 407 Proxy Authentication Required</td>
</tr>
<tr>
<td>05:39:53</td>
<td>Rx (1379)</td>
<td>REGISTER; 200 OK</td>
</tr>
<tr>
<td>05:39:53</td>
<td>Rx (1379)</td>
<td>REGISTER; 100 Trying</td>
</tr>
</tbody>
</table>
13.4 Firmware Upgrade

- **Music on Hold:**
The acceptable format is header-less 8-bit raw data in a sampling rate of 8,000 samples per second, and encoded as compressed G.711μ or G.711A. Only the beginning 128 Kbytes (at most) would be saved. We recommend the uploaded tone size should be less than 128K bytes to facilitate the upload process.

- **Compressed Image:**
Image for firmware upgrade. The largest upload size is 4 MB, and only image released by authorized by companies could be uploaded. **Note**, after upgrading all file(s) or/image, **you MUST reboot YV3 to start running with this new image (image upload will not trigger an auto-reboot).**

- **Factory Value:** upload factory default values.
The data is used whenever the user performs a 「Restore to Factory Value」operation to restore this terminal’s configurations back to factory default value. Only configuration files released by authorized companies could be uploaded.

- **Configuration Data:**
Upload a provision file to configure this terminal. The file format should be as the same as the one got by auto-provision procedure (refer to “Auto-Provision” on YV3 Administration.).

---

For example, to import personal address book, you may enter the following information and update it as plain text:

```
// 1. everything after double slashes, "//", are ignored
// 2. The tag, AB[x], where x ranges between 0~999, and no spaces are allowed for tag
// 3. If the specified contact has been existed (comparing only protocol and email-like
//    address), then it will not be added into terminal address book; otherwise it will
//    be added into address book. For example, "Michael Wu"<sip:200@ISP.com> is
//    the same as "michael<sip:200@ISP.com>".
AB[0]=Michael <sip:200@isp.com>
AB[1]=Horace Fu <sip:Horace@somewhere> //
AB[2]=sip:9100@voip.net
AB[3]=5100 // same as <sip:5100>
AB[4]=alber@isp.com // this is same as <sip:albert@ISP.com>
AB[123]=sales@ibm.com // the index can be randomly chosen between 0~999.
AB[999]=mary<sip:1237@biz.com> // this is the max index
```
Decryption Key:
Specify whether the uploaded configuration data should be deciphered by Advanced Encryption Standard, AES, of 192-bit key length.

- As Plain Text (Not encrypted)
- Decipher by Import Key: Use the key specified in 「YV3」 / 「Auto-Provision」 / 「Security」 / 「Decryption Key」
- Decipher by Export Key: Use the AES key specified in 「YV3」 / 「Advanced」 / 「Export」 / 「Security」 / 「Encryption Key」
- Decipher by Built-in System Key
Appendix A – Trouble Shooting

1. To verify your network, please go to [Advanced] , then type a domain name to ping its reachability and/or aliveness, like “yahoo.com”,”iptel.org” or “fwd.pulver.com”. If the response is “Host unreachable”:
   i. Check your network link, make sure it works normally (RJ-45 jack plugs into the right hole and the LAN indication LED should be on.
   ii. Check your IP, DNS, and gateway settings on [Network] setting page. Note: if you do not statically assign domain name server by picking [Static DNS Server] , you must enable either [DHCP] or [PPPoE] ; otherwise, it will use the static DNS IP configured into the [Primary DNS Server] and [Alternate DNS server] item.
   iii. If you reside on LAN without gateway, you should specify “0.0.0.0” as your gateway IP to disable gateway routing rather than assign a non-existent or an invalid IP; otherwise the network packets may not be routed correctly (which may result in no voice packets could be sent from this phone)! This constrain applies to DHCP and PPPoE as well: DHCP and PPPoE server should not designate a non-existent or an invalid gateway.
   iv. Go to [YV3] page to make sure that those [Active Network Status] matches those configured. Specifically, if the active DNS is 0.0.0.0, then you may have wrongly configured to [Static DNS Server] without setting a valid DNS IP in either [Primary DNS server] or [Alternate DNS Server] (see point-2 for detail).

2. To verify SIP settings:
   i. Go to [SIP Settings] page:
      a. Your [Transport] setting should include [UDP]. Verify the network settings to see whether it works normally.
      b. Your [SIP Listen Port] setting must be less than 65536 and greater than 0. We suggest a value greater than 5000 to avoid accidentally conflicting with system service ports. System default is 5060.
   ii. Go to [Network] / [RTP Settings] :
      a. Your [RTP Port Base] should be an even number ranges between 2 and 65534.
      b. Your [RTP Port Range] setting should be an even number and larger than or equal to 2. At least 4 ports should be configured for YV3 since it has a maximum capacity of two concurrent calls and each call consumes two consecutive UDP port pair (one for RTP and the other for RTCP). We suggest a range value of 6
      c. The sum of [RTP Port Base] and [RTP Port Range] must be less than 65536, and must not overlap with [SIP Settings] / [SIP Listen Port].
   iii. Go to [N-th Domain] and configure
      a. Authentication
      b. SIP Address-of-Record
      c. Proxy Server
      d. Registrar
   Configure SIP domain applied from service provider as appropriate. To apply for a public domain account, you may go to www.freeworlddialup.com or www.iptel.org.

3. Check for NAT and Firewall settings:
   Go to [Network] / [NAT & firewall] , enter the STUN server IP provided by your service provider. For example, you may enter “stun.YourISP.com” or “YourISP.com” or “larry.gloo.net” for test purpose. Please refer to chapter 6 – [NAT Traversal] to determine the best way to traverse your network address translator or firewall deployed by your ISP or company.

4. Experience long-delay or poor response time on making calls (roughly 10 seconds):
   The most possible cause is that you have enabled STUN to traverse NAT, but the STUN server is either down or unreachable.
   To verify the exact reason, please go to [Network] / [NAT & firewall] and click [Set and Diagnose NAT] to figure it out. If the result is [Firewall blocks UDP] and you are pretty sure that you are not under such a network condition, you should check for the DNS and STUN server, making sure they works normally by Pinging for their viability. Otherwise, you should pick [Static NAT IP/UDP Map] as the way to traverse NAT.

[115/117]
5. After setting up a call, it always disconnects automatically after 32 seconds. This may be due to the fact that either the involved SIP proxy servers and / or the terminals (either the caller or callee) are behind NAT, such that the ACK signal cannot be transmitted through the signaling path. If only the terminals are behind the NAT, please refer to previous parts for NAT & firewall traversal. On the contrary, if it is the SIP proxy that behind NAT, which has a private unroutable IP address, then it hardly leaves room to solve this problem except for direct IP dialing (because most proxy servers demands record-route, but it fails if any of the involved servers bear private IP).

6. Experience one-way voice (or no voice could be heard by either party) after call setup. This is because one of the terminals resides behind NAT and it does not specify a way to traverse it reliably. Normally, the one who cannot hear voice mainly because it specifies a private IP address to receive voice media on call setup phase; as a result, the peer starts to send voice media to this unroutable private IP address and the voice packets will get lost since private IP are not routable. To verify the real cause:
   i. If both parties have connected and engage in calls already, please go to 「Call Statistics」/「Channel Info」 page to view the following information and check whether they convey private IPs (alternatively, you may activate by alternatively, you may press DSS key 【F10】，which default maps to “Channel Info”, to activate 「Channel Info」):
      a. From(Contact)
      b. Media Session => both local and remote RTP session.
   ii. If you have hanged up the call, please go to 「Call Statistics」/「Call Detail」 to check for the following information and figure out whether they convey private IPs (alternatively, you may activate by alternatively, you may press DSS key 【F11】，which default maps to “Call Detail”, to activate 「Call Detail」):
      a. From(Contact)
      b. Media Traffic / RTP Session => both local and remote RTP session.
   iii. The format of 「Contact」is xxx.xxx.xxx.xxx:port, which specifies the peer’s contact information for SIP signaling.
   iv. The format of RTP/RTCP session comes in the format “xxx.xxx.xxx.xxx:RTP/RTCP”, which specifies the IP of CPE and the UDP port used for RTP and RTCP session.

For YV3, if either side is behind NAT and conveying its private IP, please specify a way to solve this problem from page 「Network」/「NAT & firewall」，and refer to chapter 6 – 「NAT Traversal」 to determine the best way to traverse your network address translator or firewall deployed by your ISP or company.

7. Sometimes there would be only one party be held successfully on conference mode when the conference master presses 【HOLD】:

This happens only on some SIP proxy server implementations, such as the VOCAL server (both 1.4 and 1.5) from www.vovida.com. The problem is due to the fact that those implementations will increase their “nonce” on authorization (www-authORIZATION), therefore forbid multiple concurrent transactions originated from the same host.

The same problem happens on those servers when you have multiple calls on hold!

8. Direct IP dialing or LAN dialing cannot work:

Most often it is due to the SIP service port does not match. For example, if your dial string is something like “*100”, then the default SIP service port of the called party is UDP 5060. If the called party does not run its SIP service port on UDP 5060, then it definitely will not ring. To solve it, please dial the port as well if the called party listens on port other than UDP 5060. For example, dial “192*168*1*200**6666” to reach “192.168.1.200:6666” (you first dial the IP dial prefix to activate IP dial plan as appropriate).

YV3 by default will always listen on UDP port 5060 in addition to the user configured SIP Listen port on 「SIP Settings」 page.

9. Received “Unsupported Media” or “Not Acceptable” while making calls.

This message will be shown if both sides fail to negotiate a CODEC for media session. The failure cause is due to the mutual exclusion of both parties’ CODEC capabilities. For example, if you specify explicitly to use only G.723.1 for voice stream whereas the peer is only capable of G.711, then the conversation cannot proceed.

To ensure the phone will gracefully fall back to G.711, either party of the call should not disable G.711 CODEC. To check CODEC settings on YV3, please go to 「Advanced」/「CODEC」 page to check for
[Preference] section. If you prefer to use G.729 or G.723.1 for voice compression, consider lowering the priority of G.711 rather than disabling them.

10. I can call others, but some of them or all others can not ring me.

Possible cause:

i. You are in All Call Forward mode (You could hear stuttering dial tone while picking up handset).

ii. Both of you are under the same NAT, and either one of you employs [Enable STUN] or [Static NAT IP/UDP Map] to traverse NAT. This is largely because some NATs will not loop packets back if the source and destination are on the same NAT.

iii. You use [Static NAT IP/UDP Map] to traverse NAT but the assigned static NAT IP on [NAT & Firewall] page does not match the real IP of your NAT. (This is possible especially when your NAT does not come with a fixed IP but employ PPPoE or DHCP to gain access to WAN instead). By default, YV3 will auto-detect the NAT IP (if the STUN server configured into [NAT & Firewall] page is viable) and notify you whenever you hit the [Network] / [NAT & Firewall] page if it detects a mismatch between the static NAT IP you configured and the one it auto-discovered by STUN protocol. Alternatively, you enable the option – [Auto Update Static NAT IP by STUN] to auto-refresh the newly changed NAT IP whenever it detects an inconsistency.