



# Internet Telephony Gateway

VIP-160/VIP-260

User's manual

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# Home

System Information	
System Uptime:	0 days, 0h 1m 7s
LAN IP Address:	192.168.0.1 (Static)
MAC Address:	00:30:4f:32:34:b8
Security:	Password installed
Application Code Version:	SIP version 1.0 US (AC2S 0100)
Downloader Code Version:	1.0 US (NTRG VR33A)

**System Uptime:** specifies the amount of time, which the system has been up. This time is reset every time the system is reset.

**LAN IP Address:** indicates the IP Address of your LAN.

**MAC address:** MAC address is the address of your MAC.

**Security:** for your password, which is configured in the “System” section.

**Application Code Version:** tells the version of the application code which you are using.

**Download Code Version:** tells the version of the download code which you are using.

# Network

## Network status

The screenshot shows the PLANET SIP Internet Telephony Gateway Web Management interface. The left sidebar contains a navigation menu with the following items: Home, Internet Access Setup, SIP Setup, System Setup, Advanced Setup, Tools (with sub-items: Firmware Upgrade, Reset, Logout, System Status, Network Status), and Help. The main content area is titled "Network Status" and displays the following information:

Interface Status	
Enabled:	Yes
Service:	Routed
Protocol:	Ethernet
Interface Status:	<b>Up</b>

  

Network Settings	
Dynamic IP Assignment:	NO
IP Address:	<b>210.66.155.81</b>
MAC Address:	00:30:4f:32:34:b8
Subnet Mask:	255.255.255.0
Default Gateway:	210.66.155.94
DNS Address:	168.95.192.1
Domain Name:	
VLAN Tag:	Not set
Priority Tag:	Not set
Broadcast limit:	100% (of downstream bit rate)
Multicast limit:	100% (of downstream bit rate)

Below the network settings is a "REFRESH" button.

**Interface Status:** these are the details of your interface's status.

**Enabled:** "Yes", lets you know that your interface is enabled and ready to be used.

**Service:** either "Routed or Bridged", tells you the level of your interface's connection.

**Protocol:** refers to how you are transmitting data. (i.e. Ethernet)

**Interface Status:** either "Up" or "Down".

**Under Network Settings:** these are the details of your network settings.

**Dynamic IP Assignment:** "Yes" or "No", depending on whether or not you are using a dynamic IP.

**IP address:** your specified IP.

**MAC address:** Your specified MAC address.

**Subnet Mask:** indicates the IP address of your mask.

**Default Gateway:** is the IP address of the gateway. The gateway IP could be retrieved from DHCP offer in DHCP mode, or be set up manually in fixed IP mode.

**DNS address:** refers to the address of your dynamic name server, if applicable.

**VLAN:** VLAN tag value encoded in the Ethernet header in all outgoing packets

**Priority Tag:** Priority Tag value encoded in the Ethernet header in outgoing packets.

**Broadcast Limit & Multicast Limit:** Please see the **WAN Configuration**

# WAN Configuration

The screenshot shows the PLANET SIP Internet Telephony Gateway Web Management interface. The left sidebar contains a navigation menu with the following items: Home, Internet Access Setup, LAN, WAN, PPPoE, DHCP Server, MAC Cloning, SIP Setup, System Setup, Advanced Setup, Tools, Firmware Upgrade, Reset, Logout, System Status, Network Status, and Help. The main content area is titled "WAN Configuration" and features a "Device Operating Mode" dropdown menu set to "Router". Below this, there are two radio button options: "Obtain WAN configuration dynamically" (which is unselected) and "Specify fixed WAN configuration" (which is selected). Under the selected option, there are input fields for IP Address (210.66.155.81), IP Netmask (255.255.255.0), IP Gateway (210.66.155.94), and IP DNS Server (168.95.192.1). There are also empty input fields for Host Name and Domain Name. A "DONE" button is located at the bottom of the configuration area.

1. **Device Operating Mode:** you choose either “Router” or “Bridge”, depending on your operation.

2. You will check either “**Obtain WAN configuration dynamically**” or “**Specify fixed WAN configuration**”.

When you choose “**Obtain WAN configuration dynamically**”, the information is detected automatically through DHCP.

If you choose “**Specify fixed WAN configuration**”, you are required to enter the IP address, IP of the Sub mask, IP of the Gateway, and IP of the DNS Server, if applicable.

# WAN PPPoE Configuration

The screenshot shows the Planet SIP Internet Telephony Gateway Web Management interface. On the left is a navigation menu with the following items: Home, Internet Access Setup (LAN, WAN, PPPoE, DHCP Server, MAC Cloning), SIP Setup, System Setup, Advanced Setup, Tools (Firmware Upgrade, Reset, Logout, System Status, Network Status), and Help. The main content area is titled "PPPoE Configuration" and contains the following sections:

- Enable PPPoE:** A dropdown menu currently set to "No".
- Authentication:** Two input fields for "Username:" and "Password:".
- Connection Settings:** Four input fields: "Idle Timeout:" (with a "minutes" label), "Echo Timeout:" (with a "seconds" label), "Echo Count:", "Service Name:", and "AC Name:".

At the bottom of the configuration area is a "DONE" button.

1. **Enable PPPoE:** "Yes" or "No", to enable/disable PPPoE

2. Under "**Authentication**", you enter the username and password given by your ISP.

3. **Settings:**

**Idle Timeout:** Idle timeout before PPP connection is closed due to inactivity

**Echo Timeout:** the duration between PPP echo requests being sent to the server.

**Echo Count:** the number of unanswered PPP echo requests before the PPP connection is closed.

**Service Name:** PPPoE Service name

**AC Name:** PPPoE AC name

# MAC Cloning Configuration

The screenshot shows the Planet SIP Internet Telephony Gateway Web Management interface. The top header includes the Planet logo and the text "SIP Internet Telephony Gateway Web Management". A left-hand navigation menu is visible, listing various configuration options such as "Home", "Internet Access Setup", "LAN", "WAN", "PPPoE", "DHCP Server", "MAC Cloning", "SIP Setup", "System Setup", "Advanced Setup", "Tools", "Firmware Upgrade", "Reset", "Logout", "System Status", "Network Status", and "Help". The main content area is titled "MAC Address Cloning" and features a text input field labeled "WAN MAC Address:" with a "DONE" button below it. A note is displayed below the input field, stating: "NOTE : MAC address cloning is used on CABLE connection. In most circumstances, there is no need to configure this, but some ISPs require a particular value, often that of the PC initially used for Internet access."

## WAN MAC Address (Spoofed):

This is only available when devices are under the router mode. The spoofed MAC address to be used by the device's WAN interfaces, the Ethernet address of the outgoing packets from the WAN interface would be replaced with this address. If blank, the WAN interfaces will use the value of MAC.



# LAN

## LAN Configuration

The screenshot displays the PLANET SIP Internet Telephony Gateway Web Management interface. The top header features the PLANET logo and the text "SIP Internet Telephony Gateway Web Management". A left-hand navigation menu is visible, listing various configuration options under a "Home" section, including Internet Access Setup, LAN, WAN, PPPoE, DHCP Server, MAC Cloning, SIP Setup, System Setup, Advanced Setup, Tools, Firmware Upgrade, Reset, Logout, System Status, Network Status, and Help. The main content area is titled "LAN Configuration" and contains a "Network Settings" section. This section includes two input fields: "IP Address" with the value "192.168.0.1" and "Subnet Mask" with the value "255.255.255.0". Below these fields is a "DONE" button.

1. Under “**Network Settings**”, you enter the **IP address** and **subnet mask** of your network.

# DHCP Server Configuration

The screenshot shows the PLANET SIP Internet Telephony Gateway Web Management interface. The left sidebar contains a navigation menu with the following items: Home, Internet Access Setup (LAN, WAN, PPPoE, DHCP Server, MAC Cloning), SIP Setup, System Setup, Advanced Setup, Tools (Firmware Upgrade, Reset, Logout, System Status, Network Status), and Help. The main content area is titled "DHCP Server Configuration" and contains the following sections:

- Server Settings:** Includes radio buttons for "Enabled" (selected) and "Disabled", and a "Client IP Address Range" field set to "192.168.0." followed by input boxes for "100" and "131".
- DHCP Client Network Information:** Includes a "Domain Name" input field and "DNS Server 1:" and "2:" input fields.
- Static Address Assignments:** A table with columns "Identify Using", "Host Identifier", and "Internal Address". The first row shows "Hostname" in a dropdown, an empty "Host Identifier" box, and "192.168.0." followed by an empty "Internal Address" box. An "Add" button is to the right.

At the bottom of the configuration area are two buttons: "Save DHCP Settings" and "View DHCP Table".

These configuration parameters are for the device's internal DHCP server.

## 1. Server Setting: "Yes" or "No", to enable/disable DHCP

**Client IP Address Range:** Minimum and Maximum limit on the DHCP IP address pool

## 2. Client Network Information

**Domain Name:** LAN domain name provided to DHCP clients during the OFFER process.

**DNS Server:** This statically assigned DNS server IP address will be provided to clients during the OFFER process.

## 3. Static Address Assignment

Up to eight static DHCP address assignments can be configured. To add a static IP assignment, enter the LAN device's **host name** (must be unique in the private network) and/or **MAC address**. Specify the **Internal address** to be assigned and press the "Add" button.

# Port Forwarding Configuration

The screenshot shows the PLANET SIP Internet Telephony Gateway Web Management interface. The left sidebar contains a navigation menu with the following items: Home, Internet Access Setup, SIP Setup (with sub-items: SIP Configuration, Advanced SIP Conf, Voice Codec), System Setup, Advanced Setup (with sub-items: DTMF Setup, ToS/DiffServ Setup, Port Forwarding, Routing), Tools, and Help. The main content area is titled "Port Forwarding Configuration" and includes a "Reserved Ports" section with a note: "NOTE: following ports have been reserved by machine itself, and may not be forwarded to the LAN clients: 68, 5060-5070, 8000-8015, 5555, 80, 161". Below this is a "Port Forwarding" section with a table for configuration:

Port Range	Protocol	Destination Address
<input type="text"/> - <input type="text"/>	Both	192.168.0. <input type="text"/>

A "DONE" button is located at the bottom of the configuration area.

1. Under “**Reserved Ports**”, specified are the ports, which cannot be forwarded to the LAN.
2. Under “**Port Forwarding to LAN**”, you enter the specifications, which you will be forwarding to the LAN, including **port range**, **protocol**(Both, TCP or UDP), and **destination IP address**.

Click on “**DONE**” to save your configurations.

# SIP

## SIP Configuration

The screenshot displays the 'SIP Internet Telephony Gateway Web Management' interface. On the left is a navigation menu with categories like Home, Internet Access Setup, SIP Setup, System Setup, Advanced Setup, Tools, and Help. The main content area is titled 'SIP Configuration' and is divided into two sections: 'Proxy Server Settings' and 'Gateway Settings'.

**Proxy Server Settings** (Current Server: : 5060 ; Domain: proxy.addaline.com)

- \* Server Address: [ ] (IP address or URL)
- \* Port: [ 5060 ]
- Domain Name: [ proxy.addaline.com ]
- Send Registration Request with Expire Time [ 10 ]
- Outbound Proxy IP: [ ] (IP address or URL)
- Outbound Proxy Port: [ ]
- Outbound Proxy host: [ ]

**Gateway Settings**

- Dial Plan: [ xxxxxxx ]
- # use as a quick dial function
- To enable # to be recognized as dial number
- To enable \* to be recognized as dial number

	Phone Number	CallerID Name	Port	AEC On	User Name	Password
Line1:	[ 271977 ]	[ ]	[ 5060 ]	[ ON ]	[ 271977 ]	[ ..... ]
Line2:	[ ]	[ ]	[ 5061 ]	[ ON ]	[ ]	[ ]

1. Under “SIP Server Settings”, you enter the **server address**, **port**, **domain name**, and **expiration time** unit, if you choose to send a registration request with an expiration time.

### 2. Gateway Settings

- **Dial Plan:** refer to appendix A of this guide
- **# use as a quick dial function:** If this box is checked, the dialed digits would be sent out when the ‘#’ key is pressed.
- **Enable # to be recognized as dial number:** allow the ‘#’ key to appear in the INVITE request URI
- **Enable \* to be recognized as dial number:** allow the ‘\*’ key to appear in the INVITE request URI
- For the line on the endpoint, enter the **Line Phone Number**, **Caller-ID Name**, **signalling port value**, **authentication Username** and **Password**, and select if **AEC** is to be performed on this line.

# SIP Extensions

The screenshot shows the 'SIP Internet Telephony Gateway Web Management' interface. On the left is a navigation menu with the following items: Home, Internet Access Setup, SIP Setup (expanded), SIP Configuration, Advanced SIP Config, Voice Codec, System Setup, Advanced Setup (expanded), DTMF Setup, ToS/DiffServ Setup, Port Forwarding, Routing, Tools, and Help. The main content area is titled 'Advanced SIP Configurations' and contains the following options:

- Support PRACK method with provisional response reliability
- Encode SIP URI with user parameter
- Send INVITE with Timer header value:
- SIP Session Timer value:
- Conditional Call Forwarding Timer:
- Disable Call Waiting (Reject second incoming call)
- Disable Caller-ID Display
- Call Hold using c=0.0.0.0 in SDP
- send NOTIFY for REFER request

A 'DONE' button is located at the bottom of the configuration area.

1. **Support PRACK method:** enable SIP PRACK support.
2. **Encode SIP URI with user parameter:** encode user=phone parameter in SIP URI.
3. **Send INVITE with Timer header:** encode Timer header in all INVITE requests for ringing timeout
4. **SIP session timer:** enable SIP session timer function.
5. **Conditional Call Forwarding Timer:** Forward the call to the pre-configured number if the phone does not pick up within the timer.
6. **Disable Call Waiting:** disable the call waiting tone.
7. **Disable Caller-ID display:** disable the caller-id display of incoming calls.
8. **Call Hold using C=0.0.0.0:** using the call hold method described in RFC2543. If unchecked, the call hold would follow RFC3263 method
9. **Send NOTIFY:** send out NOTIFY request to transferor for unattended and attended call transfer.

# DTMF Configuration

The screenshot shows the Planet SIP Internet Telephony Gateway Web Management interface. The top header includes the Planet logo and the text "SIP Internet Telephony Gateway Web Management". A left-hand navigation menu lists various configuration categories such as Home, Internet Access Setup, SIP Setup, System Setup, Advanced Setup, Tools, and Help. The main content area is titled "DTMF Configuration" and contains the following settings:

- Send DTMF Events:
- RFC2833 signalling using payload value:
- Regenerate OOB DTMF tone

A "DONE" button is located at the bottom of the configuration area.

This sub-page allows configuration of the out-of-band signalling options for SIP. Select whether OOB telephone event signalling is to be done using the SIP INFO message, or to be done via RFC2833 RTP signalling. For additional information please refer to the RFC2833.

## ToS/DiffServ

The screenshot displays the PLANET SIP Internet Telephony Gateway Web Management interface. On the left is a navigation menu with categories like Home, Internet Access Setup, SIP Setup, System Setup, Advanced Setup, Tools, and Help. The main content area is titled "ToS/DiffServ Configuration" and contains two input fields: "Call Signalling Packets: C0 (2 Hex digit byte value)" and "RTP Packets: A0 (2 Hex digit byte value)". A "DONE" button is located below the input fields.

This sub-page is used to configure the Type-of-Service/Diffserv byte values, which are to be used in the IP header of all transmitted SIP signalling packets and RTP packets. The ToS/DiffServ byte values are entered as two-digit hexadecimal values. If no special ToS/DiffServ value is to be used for a particular traffic type, enter “00” or leave the setting empty.

Press “**DONE**” to save these new settings.

# CODEC

## Voice Codec Configuration

The screenshot shows the PLANET SIP Internet Telephony Gateway Web Management interface. The sidebar menu on the left includes: Home, Internet Access Setup, SIP Setup (SIP Configuration, Advanced SIP Configuration, Voice Codec), System Setup (User Password, Admin Password, Access Privilege, Time/Area Setup, SNMP Setup), Advanced Setup (DTMF Setup, ToS/DiffServ Setup, Port Forwarding, Routing), Tools, and Help. The main content area is titled "Voice Codec Configuration" and contains the following settings:

- Available Codec:** A table with columns "Selected" and "Silence Suppression".

Selected	Silence Suppression
<input checked="" type="checkbox"/> G711U	ON
<input checked="" type="checkbox"/> G711A	ON
<input type="checkbox"/> G723	ON
<input type="checkbox"/> G726	ON
<input checked="" type="checkbox"/> G729	OFF
- Packetization:** 10ms
- Jitter Buffer:**
  - Adaptive Jitter Buffer: 100ms (maximum playout delay in milliseconds)
  - Fixed Jitter Buffer: 40ms (fixed playout delay in milliseconds)
  - Automatically switch to Fixed Jitter Buffer upon fax/modem tone detection

A "DONE" button is located at the bottom of the configuration area.

1. **CODECS:** configure the silence suppression to your desired settings.

2. **Packetization:** configure the packet sending increments.

3. **Jitter Buffer:** configure the timing of the voice buffering.

Selection between adaptive or fixed jitter buffer. Default = ADAPTIVE

Set the adaptive jitter buffer maximum playout delay. Default = 100ms

or Fixed jitter buffer playout delay. Default = 40ms

Whether or not to automatically switch from an adaptive jitter buffer to a fixed jitter buffer upon fax/modem tone detection

Click on "**Save CODEC Configuration**" to save the configurations made.



# SYSTEM

## Set Administrator Password

The screenshot displays the PLANET SIP Internet Telephony Gateway Web Management interface. The top navigation bar includes the PLANET logo and the text "SIP Internet Telephony Gateway Web Management". A left-hand sidebar menu lists various configuration categories: Home, Internet Access Setup, SIP Setup (containing SIP Configuration, Advanced SIP Conf, and Voice Codec), System Setup (containing User Password, Admin Password, Access Privilege, Time/Area Setup, and SNMP Setup), Advanced Setup (containing DTMF Setup, ToS/DiffServ Setup, Port Forwarding, and Routing), Tools, and Help. The main content area is titled "Administrator Password Setup" and contains the following text and form elements:

Administrator Password Setup

---

Password is currently installed.

New password:

Confirm new password:

---

Configure a **password** for the system.

## Time /Area Setup

The screenshot displays the PLANET SIP Internet Telephony Gateway Web Management interface. The left sidebar contains a navigation menu with the following items: Home, Internet Access Setup, SIP Setup (with sub-items: SIP Configuration, Advanced SIP Conf, Voice Codec), System Setup (with sub-items: User Password, Admin Password, Access Privilege, Time/Area Setup, SNMP Setup), Advanced Setup (with sub-items: DTMF Setup, ToS/DiffServ Setup, Port Forwarding, Routing), Tools, and Help. The main content area is titled "Time/Area Setup" and contains the following configuration fields: "Country:" with a dropdown menu set to "United States", "NTP Server:" with an empty text input field, "Time Zone:" with a dropdown menu set to "GMT+08:00 Taipei Time", and an unchecked checkbox labeled "Adjust clock for daylight savings". A "DONE" button is located at the bottom of the configuration area.

Choose the correct country for a proper impedance match, as well as the NTP Server, and Time Zone. Check the “**Adjust clock for daylight savings**”, when applicable.

Click on “**Save Localization Settings**”, to save your configurations.

# SNMP Configuration

**SNMP Configuration**

**SNMP Trap Configuration**  
IP address:  Trap Community:

**SNMP Community Configuration**  
Read Community:  Write Community:

**SNMP System Configuration**  
System Description:   
System Objectid:

## 1. SNMP Trap Configuration

**IP address:** Trap host IP address

**Trap Community:** The community name used by the SNMP manager to verify traps. The default value is 'public'

## 2. SNMP Community Configuration

**Read Community:** The community name used by the SNMP manager when reading SNMP data items from a client MIB. The default value is 'public'

**Write Community:** The community name used by the SNMP manager when setting SNMP data items in a client's MIB. The default value is 'public'

## 3. SNMP System Configuration

**System Description:** Description of the unit (e.g. "John's phone")

**System Object Id:** A vendor's enterprise ID

# Access Privilege Configuration

The screenshot shows the Planet SIP Internet Telephony Gateway Web Management interface. The left sidebar contains a navigation menu with the following items: Home, Internet Access Setup, SIP Setup, System Setup (with sub-items: User Password, Admin Password, Access Privilege, Time/Area Setup, and SNMP Setup), Advanced Setup, Tools, and Help. The main content area is titled "Access Privilege" and includes the instruction: "Select which interfaces are allowed access to machine services". Below this is a table with columns for "LAN" and "WAN". The rows are "HTTP (Web access)", "SNMP", and "VoIP Discovery", each with checked boxes in both the LAN and WAN columns. A "DONE" button is located at the bottom of the configuration area.

	LAN	WAN
HTTP (Web access):	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
SNMP:	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
VoIP Discovery:	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>

Check the proper boxes enabling LAN and WAN for the **HTTP, SNMP, and VoIP Discovery**.

Click on "**DONE**", to save the configurations.

# Firmware upgrade

The screenshot displays the Planet SIP Internet Telephony Gateway Web Management interface. The left sidebar contains a navigation menu with the following items: Home, Internet Access Setup, SIP Setup, System Setup (with sub-items: User Password, Admin Password, Access Privilege, Time/Area Setup, and SNMP Setup), Advanced Setup, Tools (with sub-items: Firmware Upgrade, Reset, Logout, System Status, and Network Status), and Help. The main content area is titled "Firmware upgrade" and is divided into two sections. The first section, "Upgrade via TFTP Server", includes a sub-header "(Select remote TFTP server IP address and filename)", a "TFTP Server IP:" input field, a "Filename:" input field, and a "START DOWNLOAD" button. The second section, "Upgrade via web browser", includes a sub-header "(Firmware file should be located on local browser machine)", a "Filename:" input field with a "Browse..." button, and a "START DOWNLOAD" button. A warning message at the bottom of the page states: "Warning! During machine firmware upgrade. All network connections will be torn down, and be sure to keep stable power supply till upgrade process is completed."

For both **HTTP and TFTP methods**, the device will reboot itself into downloader mode if the main application is being executed, and proceed with the ROM file download and permanent write of the application to the device's flash memory. After the download is completed, the download status page will be displayed.

# Reset

The screenshot shows the Planet SIP Internet Telephony Gateway Web Management interface. The top header includes the Planet logo and the text "SIP Internet Telephony Gateway Web Management". A left-hand navigation menu lists various system management options such as "Home", "Internet Access Setup", "SIP Setup", "System Setup", "User Password", "Admin Password", "Access Privilege", "Time/Area Setup", "SNMP Setup", "Advanced Setup", "Tools", "Firmware Upgrade", "Reset", "Logout", "System Status", "Network Status", and "Help". The main content area is titled "Reset" and features a radio button next to the "Reboot machine" option, with a "RESET" button below it. A note in orange text states: "Note: Resetting the system will terminate all network connections and reset your browser connection".

Chose the “**Reboot machine**” option to reset the system.

## Appendix A. Dial Plans

The machine will allow provisioning (via web browser) of the dial plan. A dial plan gives the unit a map to determine when a complete number has been entered and should be passed to the gatekeeper for resolution into an IP address. Dial plans are expressed using the same syntax as used by MGCP NCS specification.

The formal syntax of the dial plan is described in the following notation:

Digit ::= "0" | "1" | "2" | "3" | "4" | "5" | "6" | "7" | "8" | "9"

Timer ::= "T" | "t"

Letter ::= Digit | Timer | "#" | "\*" | "A" | "a" | "B" | "b" | "C" | "c" | "D" | "d"

Range ::= "X" | "x" -- matches any digit

| "[" Letters "]" -- matches any of the specified letters

Letters ::= Subrange | Subrange Letters

Subrange ::= Letter -- matches the specified letter

| Digit "-" Digit -- matches any digit between first and last

Position ::= Letter | Range

StringElement ::= Position -- matches any occurrence of the position

| Position "." -- matches an arbitrary number of occurrences including 0

String ::= StringElement | StringElement String

StringList ::= String | String "|" StringList

DialPlan ::= String | "(" StringList ")"

A dial plan, according to this syntax, is defined either by a (case insensitive) string or by a list of strings. Regardless of the above syntax, a timer is only allowed if it appears in the last position in a string (12T3 is not valid). Each string is an alternate numbering scheme. The unit will process the dial plan by comparing the current dial string against the dial plan. If the result is under-qualified (partial matches at least one entry), then it will do nothing further. If the result matches or is over-qualified (no further digits could possibly produce a match), then it sends the string to the gatekeeper and clear the dial string. The Timer T is activated when it has all that is required to produce a match. The period of timer T is 4 seconds. For example, a dial plan of (xxxT|xxxxx) will match immediately if 5 digits are entered. It will also match after a 4 second pause when 3 digits are entered.

## Sample Dial Plans

### Simple Dial Plan

Allows the dialing of 7 digit numbers (e.g. 5551234) or an operator on 0. Dial plan is (0T|xxxxxxx)

### Non-dialed Line Dial Plan

As soon as the handset is lifted, the unit contacts the gatekeeper (used for systems where dtmf detection is done in-call). Dial plan is (x.) i.e. match against 0 (or more) digits. Note: the dot '.'

### Complex Dial Plan

Local operator on 0, long distance operator on 00, four digit local extension number starting with 3,4 or 5, seven digit local numbers are prefixed by an 8, two digit star services (e.g. 69), ten digit long distance prefixed by 91, and international numbers starting with 9011+variable number of digits.

The dial plan for this is:

(0T|00T|[3-5]xxx|8xxxxxxx|\*xx|91xxxxxxxxxx|9011x.T)