

# SETU VP236 System Manual



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## **Section 1: Introduction**



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## Welcome

Thanks for purchasing a Matrix product.

We want you to get the maximum performance from our product. If you run into technical difficulties, we are here to help. But please consult this system manual first.

If you still can't find the answer, gather all the information or questions that apply to your problem and with the product close to you, call your dealer. Matrix dealers are trained and ready to give you the support you need to get the most from your Matrix product. In fact, most problems reported are minor and can be easily solved over the phone.

In addition, technical consultation is available from Matrix engineers every business day. We are always ready to give advice on application requirement or specific information on installation and operation of our products.

The system manual is divided in following sections:

Section 1: Introduction

Section 2: Features and Facilities

Section 3: Appendices

We suggest the first time users to read this system manual in the following sequence:

- Section 1
- Section 2 (In the sequence given below for few topics)
  - Web Jeeves 168
  - Menu 100
  - Status 138
  - Configuring the Phone Menu 75
  - Key Assignment 87
  - Auto Configuration 44
  - Change Password 73
  - Call Progress Tone Generation 65
  - LCD Settings 92
  - Network Port 102

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- Time Settings 148
- OG Call Route Selection 110
- Making a Call 94
- Receiving a Call 119
- SIP Parameters 128
- Call Logs 60

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## Packing List

- The ideal sales package contains:

<b>Sr.</b>	<b>Item</b>	<b>Qty.</b>
01	SETU VP236S/P	1
02	System Manual	1
03	Handset	1
04	Adaptor 5V, 2.0 Amp.	1
05	RJ45 Cable	1
06	4P4C Spring Cable	1
07	Wall Mount Bracket	1
08	Warranty Card Set	1
09	Support Card	1
10	Telephone Message Pad	1
11	SETU VP236 CD	1

- Please make sure that these components are present.
- In case of short supply or damage detection, contact the source from where you have purchased the system.

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## Warranty Statement

Matrix Telecom Pvt. Ltd. (Matrix) warrants to its consumer purchaser any of its products to be free of defects in material, workmanship and performance for a period of 15 months from date of manufacturing or 12 months from the date of installation which ever is earlier.

During this warranty period, Matrix will at its option, repair or replace the product at no additional charge if the product is found to have manufacturing defect. Any replacement product or part/s may be furnished on an exchange basis, which shall be new or like-new, provided that it has functionality at least equal to that of the product, being replaced. All replacement parts and products will be the property of Matrix. Parts repaired or replaced will be under warranty throughout the remainder of the original warranty period only.

***This limited warranty does not apply to:***

1. Products that have been subjected to abuse, accident, natural disaster, misuse, modification, tampering, faulty installation, lack of reasonable care, repair or service in any way that is not contemplated in the documentation for the product, or if the model or serial number has been altered, tampered with, defaced or removed.
2. Products which have been damaged by lightning storms, water or power surges or which have been neglected, altered, used for a purpose other than the one for which they were manufactured, repaired by customer or any party without Matrix's written authorization or used in any manner inconsistent with Matrix's instructions.
3. Products received improperly packed or physically damaged.
4. Products damaged due to operation of product outside the products' specifications or use without designated protections.

***Warranty valid only if:***

- Primary protection on all the ports are provided
- Mains supply is within limit and protected
- Environment conditions are maintained as per the product specifications

**Warranty Card:**

- When the product is installed, please return the warranty card with:
  - Date, signature and stamp of the customer.
  - Date, signature and stamp of the channel partner.
- Matrix assumes that the customer agrees with the warranty terms even when the warranty card is not signed and returned as suggested.

The Purchaser shall have to bear shipping charges for sending product to Matrix for testing/rectification. The product shall be shipped to the Purchaser at no-charge if the material is found to be under warranty. The Purchaser shall have to either insure the product or assume liability for loss or damage during transit.

Matrix reserves the right to waive off or make any changes in its warranty policy without giving any notice.

If Matrix is unable to repair or replace, as applicable, a defective product which is covered by Matrix warranty, Matrix shall, within a reasonable time after being notified of the defect, refund the purchase price of the product provided the consumer/purchaser returns the product to Matrix.

In no event will Matrix be liable for any damages including lost profits, lost business, lost savings, downtime or delay, labor, repair or material cost, injury to person, property or other incidental or consequential damages arising out of use of or inability to use such product, even if Matrix has been advised of the possibility of such damages or losses or for any claim by any other party.

Except for the obligations specifically set forth in this Warranty Policy Statement, in no event shall Matrix be liable for any direct, indirect, special, incidental or consequential damages whether based on contract or any other legal theory and where advised of the possibility of such damages.

Neither Matrix nor any of its distributors, dealers or sub-dealers makes any other warranty of any kind, whether expressed or implied, with

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respect to Matrix products. Matrix and its distributors, dealers or sub-dealers specifically disclaim the implied warranties of merchantability and fitness for a particular purpose.

This warranty is not transferable and applies only to the original consumer purchaser of the Product. Warranty shall be void if the warranty card is not completed and registered with Matrix within 30 days of installation.

All legal course of action subjected to Vadodara (Gujarat, India) Jurisdiction only.

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## Care for Your VP236

### Power

- To disconnect power from the device the electric plug and socket must be easily accessible at all times.
- The standard power adaptor is supplied with a Phone. The phone should only be operated with this universal power adaptor. Damages to the phone caused by using other power adaptors are not covered by the manufacturer's warranty.
- This phone can be powered if Power on Ethernet (PoE) is available, without using Adaptor.

### Shock Hazard

- Avoid operating this device during periods of lightning.
- Insert carefully the adaptor in the 230VAC socket.

### Protection from Water, Heat, Sun and Cold

- Do not expose the phone to water, as this will create a shock hazard.
- The phone is an electronic device and should not be exposed to excessive heat, sun or cold.

### Cleaning Your VP236 Phone

- Please use a lightly water moistened paper or cloth towel.
- Do not spray or pour cleaning solution directly onto the phone.

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## Introducing the System

- Thank you for purchasing Matrix SETU VP236. You made an excellent choice and we hope you will enjoy all its capabilities.
- The SETU VP236 is a SIP based VoIP phone. Sending voice signals over Internet is called Voice Over IP (VoIP). And Session Initiated Protocol (SIP) is an internationally recognized standard for implementing VoIP.
- This SIP SETU VP236 is the innovative enterprise IP telephone that offers a rich set of functionality and superb sound quality. They are fully compatible with SIP industry standard and can inter operate with many other SIP compliant devices and software available in the market.
- These specialized phones look just like normal phones with a handset, cradle and buttons. But instead of having the standard RJ-11 phone connector, SETU VP236 have an RJ-45 Ethernet connector. It can be connected directly to router and have all the required hardware and software right onboard to handle the IP call.
- Dialing a number from the phone and Receiving the call is just like normal phone. Operating different features is user friendly.
- LCD display gives information about SIP account and other details regarding active call and features set. It can also display CLI of Incoming call.
- All important features are programmable through Web pages as well as features in Menu on the Phone. Some call features can be assigned to the keys on the phone so that features can be directly used. For this the phone has a key pad with 36-keys. Some keys are with LED's also for the indication of features set currently.
- Matrix SETU VP236 supports maximum of 2-SIP Accounts, and 4-Call Appearances.
- Some terminology used in this Manual are described as below:
  - **User:** The person who is using the SETU VP236.
  - **Remote User:** The person with whom the user wants to interact/interacts.
  - **Transfer Target:** The remote user to whom the call is to be transferred.
  - **Transferor:** The user who transfers the call.
  - **Transferee:** The user who is to be transferred.

- 
- **Line:** The SIP1 or SIP2 Trunk.

*The words **SETU VP236**, **IP Phone**, **Phone** and **system** are used interchangeably in this manual.*

*Before proceeding further, please ensure that you have an Internet access and a SIP Account already set up.*



**Relevant Topics:**

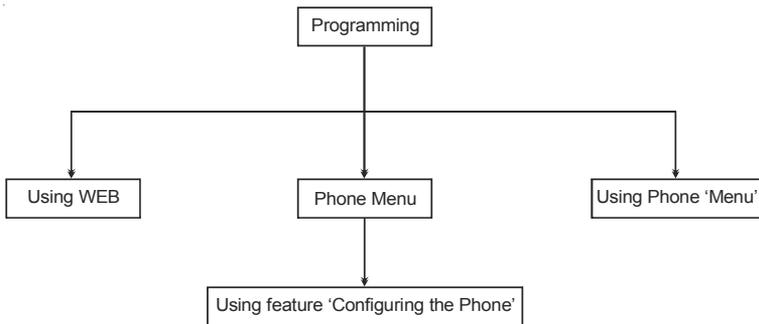
1. VoIP Basics 162
2. Glossary 179

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## Configuring SETU VP236

- Generally user will come across three varieties, while programming different features using the Matrix IP-Phone.
- Accordingly these can be grouped as mentioned below:
  - One-time programmable parameters (e.g. Ring Type, CPTG Type, etc.).
  - Less Frequently Used parameters (e.g. DND List, Black Callers' List, etc.).
  - Frequently Used parameters (e.g. Set/cancel Call Mute, Set/Cancel DND, etc.).
- The parameters which need to be programmed once only, can be programmed through Web Interface only.
- Less frequently used parameters like Auto Answer Timer, etc. can be programmed both through Web as well as Phone. Programming through Phone can be done using an option called 'Configuring the Phone' provided in the Menu of the Phone.
- Frequently used parameters like Set/Cancel Hotline are settable from Web as well as from Phone. Programming through Phone in this case is directly through Menu of the Phone.



Refer chapter '[Know Your SETU VP236](#)' for list of following parameters:

- Features programmable from 'Configuring Phone Menu'.
- Features programmable from 'Menu'.
- Features which can be assigned a 'Key'.

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## Know Your SETU VP236

Matrix SETU VP236 has the facility to use keys and LCD as explained in following sections:

- LCD Display
- Keypad and Ringer LED
- Key assignment
- Function specific key
- Using Menu key
- Options of feature 'Configuring the Phone'
- Using Dial pad key Interface

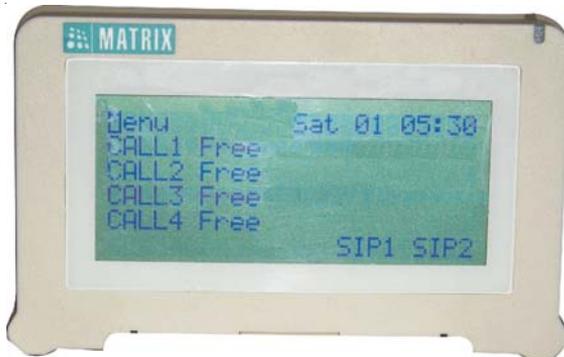
### **LCD Display:**

Matrix supports two varieties of SETU VP236:

- SETU VP236S (2x24 Line LCD)
- SETU VP236P (6x24 Line LCD)
  
- These Phones support similar features and can be used any one directly without any change. The only difference is the display capacity. Type-S comes with 2-line, 24-characters and Type-P comes with 6-line, 24-characters LCD display.
- While dialing a number, all 24 characters of LCD display are used to display the number string. First 18 characters are used to display the number string, 19<sup>th</sup> character is blank and last five characters are used to display the call progress status.
- User can dial number string with maximum of 40 characters. If dialed number string has more then 40 char, the phone will dial only first 40 char of number string.
- User should program Max. 24 characters from the phone configuration even if the maximum field length allowed for the number string/ address is more then 24.
- User should configure the Number string/ address having more then 24 characters using Web Jeeves.
- Display strings are same for 2-line LCD display and 6-line LCD display. The content of the strings (text messages) are so selected that single line is sufficient to convey the message.

**For Example:**

- If the user enters wrong password, the display string selected is 'Invalid P/w, Try again'.
- Display pattern are also same for 2-line LCD display and 6-line LCD display.
- For the normal display which consists of 6 lines the content of 'display' is as described below:
  - **First Line:**  
It will display Menu and Day, Date and Time which is not affected by Incoming or Outgoing call.
  - **Second Line to Fifth Line:**  
It is used to display status of CA1 to CA4 (Call Appearance). When a CA is free, a text string 'CALLx Free' is displayed.
- The CA row on the LCD can be treated as CA key. Doing so, following is possible:
  - In idle state, if the user presses 'Enter' key on any of the CA row, it is treated as OFF-Hook. If speaker key is pressed, user gets dial tone and 'Dial....' is displayed, in place of 'CALLx free'
  - While talking to remote user if the user presses 'Enter' key on the CA row on which the current call is going through, the call is held. The display will now show the dialed number/name and a string 'Hold' in the call status characters.
  - Likewise for all other functions which are possible from the CA keys are possible from CA rows.
- **Sixth Line:**  
It displays notification/feature settings. These strings are displayed only when the notification is received or a feature is set. Otherwise that character will display space (blank).
- The 2-line LCD user can just scroll, to know the status of other CA as only the first CA is displayed on the second line of the LCD and can select the option by pressing enter key. Whereas the 6-line LCD user will be able to view all the 6 lines of the normal display.
- **Idle Condition:**
- Normal parameter display on Phone will be as shown below:



- When the phone is in idle state the LCD display will now be displaying all the Call Appearances Free as shown above.
- When Menu is accessed, all the features are displayed in alphabetical order. Refer topic 'Menu' for more details.
- When the user makes a selection, the feature title is displayed in the first row whereas its options are displayed in subsequent rows.

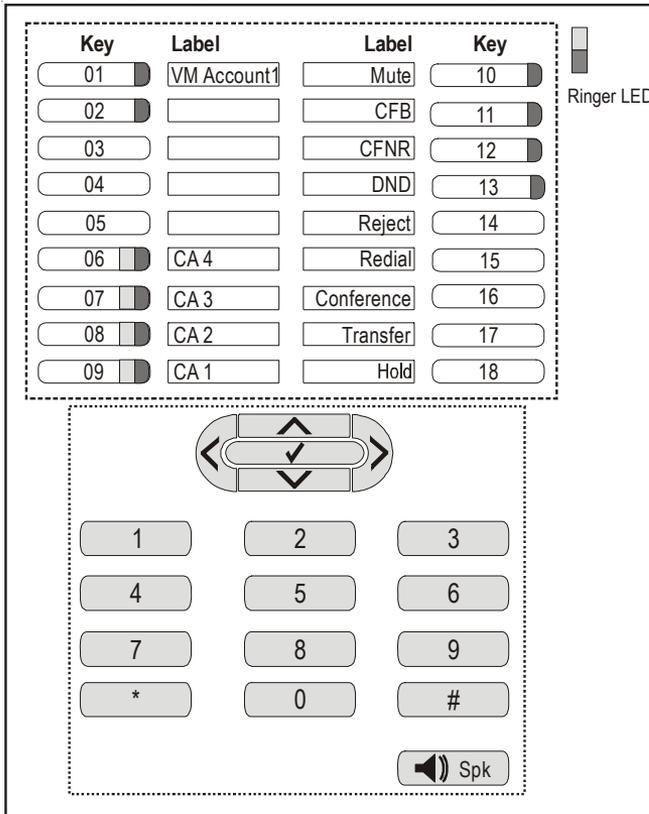
#### **For Example:**

- When the user selects 'Call Forward-Busy', CALL FORWARD-BUSY is displayed in the first row whereas Set, Cancel options are displayed in subsequent rows.

#### **Keypad and Ringer LED:**

- Matrix SETU VP236 supports a keypad with dial pad keys, programmable keys, Speaker/Headset key and feature keys as explained below:
  - 12 Dial pad keys (Non-programmable)-Fixed keys
  - 5 Navigational keys (Non-programmable)
  - 1 Speaker key (Non-Programmable) with 'Red' LED for speaker ON indication.
  - 18 feature keys (Programmable)-Keys 06 to 09 have dual color LEDs (Green and Red), keys 01, 02 and 10 to 13 have single color LEDs (Red) whereas remaining keys do not have a LED.
  - Ringer LED (Single Color) is provided on right-top side of the phone to indicate the IC call.
  - Keypad can be locked.

- Default, Key Assignment is as shown in diagram:



**Key assignment:**

- Matrix IP Phone provides the feature, “Key Assignment”.
- With this feature, user can access the feature just by pressing the key assigned.

**For Example:**

If a user uses a feature ‘Hot Line’ feature, repeatedly and he wishes to use the programmable key for this purpose, then this feature is useful. User can label that key and use every time.

- Table showing features which can be assigned to programmable key, using web page:

<b>Feature to which key canbe assigned</b>	<b>Any LED Activity on the key?</b>
None	N
Alphanumeric Dialing	Y
Auto Answer	Y
Auto Call Back	Y
Call Appearance 1	Y (Dual Colour)
Call Appearance 2	Y (Dual Colour)
Call Appearance 3	Y (Dual Colour)
Call Appearance 4	Y (Dual Colour)
Call Forward-Busy	Y
Call Forward-No Reply	Y
Call Hold	N
Call Logs	N
Call Mute	Y
Call Transfer	N
CLIR	Y
Conference	Y
DND-Set/Cancel	Y
Headset	N
Hotline-Set/Cancel	Y
Keypad Lock	Y
Last Number Redial	N
Menu	N
Phone Book (Speed Dial)	N
Reject	N
Search Contact	N
SIP1	Y
SIP2	Y
VM Account 1	Y
VM Account 2	Y

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- Key assignment can be done using Web only.
- Please note that features needing dual color LEDs (like Call Appearance) should be assigned keys 06 to 09. Likewise features needing single color LEDs should be assigned keys 01, 02 or 10-13. Whereas features not needing a LED should be assigned to a key with no LED.

**BE CAREFUL TO SELECT A KEY WITH LED OF REQUIRED NUMBER OF COLOURS FOR A FEATURE WHICH IS TO BE ASSIGNED.**

- 'None' option is to indicate that key is not assigned a feature.

### **The Function Specific Keys:**

- spk : This key is fixed for 'Speaker' mode.
- '<' : This key is used to decrease the Rx Gain of Call Progress Tones in all port states (Dial, Speech, etc.) or to go to Main Menu from Sub-menu.
- '>' : This key is used to increase the Rx Gain of Call Progress Tones in all port states (Dial, Speech, etc.).
- '√' : This key is used as selection or Enter key.
- '^' : This key is used to navigate parameters of 'Menu' to upward.
- 'v' : This key is used to navigate parameters of 'Menu' to downward.

### **Using Menu key:**

- User can access certain features using 'Menu' feature if that feature is not assigned a key. For example 'CLIR' can be enabled from the Phone just by navigating the 'Menu' and selecting 'CLIR'. Menu can be accessed by following ways:
  - Normal Display on the LCD. Select the line displaying 'Menu' and press 'Enter' key.
  - Using the specific key assigned for 'Menu'. The key assignment can be done only from the Web-Page.
- For more information refer Chapter "Menu".
- Following features can be accessed by 'Menu' key.
  - Access SIP1 trunk
  - Access SIP2 trunk
  - Alphanumeric dialing

- 
- Auto Answer
  - Auto Call Back
  - Call forward-busy
  - Call forward-no reply
  - Call hold
  - Call logs
  - Call mute
  - Call transfer
  - Call VM1 Server
  - Call VM2 Server
  - CLIR
  - Conference
  - Configuring the Phone
  - Do Not Disturb
  - Headset-OnHook/OffHook
  - Hotline
  - Keypad Lock
  - Last Number Redial
  - Network Status
  - Rejecting a Call
  - Search Contact

**Options of the feature: 'Configuring the Phone' of the Menu key:**

- User will not have to enter Web Jeeves every time for configuring the Phone. As a short cut it is possible to enter 'Config Phone' by navigating from Menu feature and add or modify SIP and other required parameters.
- User can select this option to access 'not frequently used' features.
- Thus, using the 'Config. Phone' menu, it is possible to program few parameters of the Phone from Phone itself.
- The feature 'Configuring the Phone' is one of the options of the Menu.
- Phone will ask for the 'Programming Password' (PW), when entering this option.

Remember that this Config. Phone Password is NOT same as Keypad-Unlock password or Web-Jeeves password.

- Following options are displayed when 'Configuring Phone Menu' is scrolled with up/down key:

- Anonymous Call Rejection
- Auto Answer Timer
- Black Listed Callers' List
- Call Forward Timer
- Change Configuring Password
- Change User Password
- DND List
- Hotline
- Keypad Lock Setting
- LCD Settings
- Network Parameters
- Number of Call Appearances
- Phone Book
- SIP1 Parameters
- SIP2 Parameters
- Volume Setting

### **Using Dial Pad key Interface:**

- SETU VP236 supports 12 dial pad keys, 18 feature keys, 1 speaker key and 5 navigation keys.
- 12 Dial pad keys and 5 navigation keys are used to dial a number/SIP ID, enter a number/SIP ID or edit a number/SIP ID.
- By repeatedly pressing various keys, different characters can be selected as shown in table below: (when you have selected 'Alphanumeric Dialing').

### **Table of characters:**

Key No.	Characters selected								
	First time pressed	Two times pressed	...	...	...	...	...	...	Nine times pressed
1	1	.	@	,	-	?	!	'	:
2	2	a	b	c	A	B	C		
3	3	d	e	f	D	E	F		
4	4	g	h	i	G	H	I		
5	5	j	k	l	J	K	L		
6	6	m	n	o	M	N	O		
7	7	p	q	r	s	P	Q	R	S
8	8	t	u	v	T	U	V		
9	9	w	x	y	z	W	X	Y	Z
0	0	+	-	*	/	=	%		
*									
#	#	Sp.							

- User can easily enter non numeric characters as the method of entering is same as that used by Mobile Phones while writing an SMS except that the first character in our case is a numeric when key is pressed 'first time'.
- Following examples show how to enter/edit a number/SIP ID.

### Examples:

- Select first 'Alphanumeric Dialing'.
- If user presses "1" digit key once, "1" is displayed and after waiting for 1 second (aprox):
  - If user again presses "1" within 1 second, phone will display "." and again wait for 1 second.
  - If user does not press any key within 1 second or enter other digit key, an entry of "." is done and the cursor is moved to the next position.
- If user presses "2" digit key once, "2" is displayed and wait for 1 second.
  - If user does not presses any key within 1 second, an entry of "a" is done and the cursor is moved to the next position.
- Same method is applicable for digit keys "3", "4", "5", "6", "7", "8", "9", "0" and "\*" as shown in above Table.

### **How to delete a character? (Not applicable when dialing)**

- If user presses “#” once, “#” is displayed and after waiting for 1 second:
  - If user does not press any key, an entry of “#” is displayed and the cursor is moved to the next position.
  - If user again presses “#” key within 1 second, the character is deleted on that position and the cursor is moved to the next position.
- For Example, if amit is written, user wants to delete “mit”.
  - Move the cursor to “m”, “amit”
  - Press “#” key twice within 1 second, display will be “a it”.
  - If again “#” is pressed twice within 1 second, display will be “a t”.
  - If again “#” is pressed twice within in 1 second, display will be “a “.

### **How to Override characters?**

- The IP Phone supports rewriting the numbers without first deleting the entire number.
  - For Example, if user wants to replace “12345” by “76890”.
    - Move the cursor to “1” as “12345”, now if user enters “76890”, then, “12345” is replaced by “76890”.

### **Important Points:**

- ‘Menu’ feature can be assigned a key.
- While in Menu if the user does not perform any activity, phone goes to the normal display mode after 15 seconds (fixed).
- While in option ‘Configuring the Phone’ if the user does not perform any activity, phone goes to the normal display mode after 60 seconds (fixed).
- When the SETU VP236 prompts for the password and if the user does not enter the password, phone goes to normal display mode after 15 seconds.

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## Getting Started

### Overview

The Matrix SETU VP236 allows you to use Packet Switching technique of IP network for normal communication at a very low cost of the call-rate. This is a different technology than circuit switching deployed by POTS. User needs only a subscribed line of Internet through a DSL Modem or Router and a PC with Internet Explorer (V6.0) installed in it.

This chapter gets the SETU VP236 up and running quickly. The details which we have skipped to make this brief can be found elsewhere in the user's guide. It is divided into four sections:

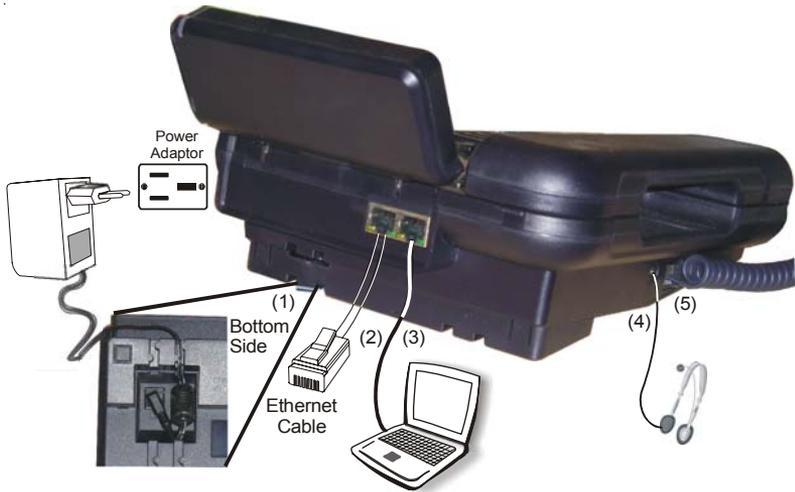
- Getting to know the SETU VP236
- Instruction for connecting the SETU VP236
- Basic steps for configuration
- Making phone calls

### Getting to know the Phone

- The Matrix SETU VP236 interfacing ports are located on side panel and LED's are on few keys. Both are explained below:
  - Connection of ports
  - LED's on Phone keys

### ***Connection of Ports:***

Following view shows how the ports are connected to various interfaces:



No.	Port Name	Connector	Description
1	Power	DC Jack	Power Adaptor, 5V 2.5Amp. DC. If ethernet switch supports PoE (Power on Ethernet), do not connect Adaptor.
2	Network Port	RJ45	Connection to cable modem/ADSL modem. Ethernet activity LED (Green) is glowing. It is a 10/100 Switch LAN port PoE (802.3agf) connection is supported, on this port.
3	PC Port	RJ45	It is a 10/100 Switch port for connecting PC LAN cable connection for co-located PC. Also connect a cable from SETU VP236 to the PC. You can also use this port as network port if PoE is not be used.
4	Headset	Audio Jack	It is a headset port. To connect head phone with suitable impedance.
5	Handset	RJ11	To connect the handset with speaker and MIC inbuilt.
6	MIC	--	This is inbuilt port. User can speak to MIC for handsfree application.

**LED's on Phone keys:**

- Matrix SETU VP236 supports few keys with single or dual color LED's for indication of a feature set e.g. Auto Call Back, DND, Hotline, etc.
- Keys 06 to 09 : The Dual color LEDs (Green and Red),
- Keys 01, 02 and 10 to 13 : The single color LEDs (Red)
- Ringer LED (Red) : Provided on right-top side of the phone to indicate the IC call.

The reference number for the keys are shown below:

**Keys with LED:**

Refer chapter '[Know your SETU VP236](#)'.

**Power ON sequence for the SETU VP236:**

- LED sequence during power ON is as shown below:

Single/Dual Color LED	Cadence (sec.)	
	ON	OFF
RED (Continuous)	15	0

**Indications on Call Appearance-LED for call under progress:**

- The status of the call is displayed on the LEDs embedded in the respective key assigned to the CA feature.

Call Event	Color	Cadence			
		On	Off	On	Off
Idle	Off	--	--	--	--
Dialing	Red (Cont.)	--	--	--	--
Proceeding	Red	500ms	1sec	--	--
Ringing (Far end)	Red → Green	500ms	--	500ms	--
Busy	Orange	500ms	1sec	--	--
Error	Orange (Cont.)	--	--	--	--
Connected	Green (Cont.)	--	--	--	--
Hold	Green	500ms	1sec	--	--
Hold Alert (for 5 secs.)	Green → Red	500ms	--	500ms	--
Remote Hold	Orange → Red	500ms	1sec	--	--
Incoming	Orange → Green	500ms	--	500ms	--
Far end rejection	Orange	500ms	500ms	--	--

### **Instruction for connecting the phone:**

This section describes the instructions on, how to connect the Matrix SETU VP236 to internet network.

Unpack the box. Get satisfied with the contents and the condition of the parts. Refer to '[Packing List](#)'. If parts are OK, proceed with connections as shown below:

- Mount your SETU VP236 in a safe and convenient location where cables for your network and phone system are accessible. For this, mechanical drawing can be used as a reference, which is provided at end of the chapter.
- Insert one end of the straight cable into the LAN port of SETU VP236 and connect the other end of the cable to an uplink port (a router or a DSL modem), which is used to access internet.
- Insert one end of the cross cable into the PC port of VP236 and connect other end of the cable to the PC.
- Connect the Audio Pin of the Headphone to the "Headset" port if you want to use the SETU VP236 in Headset-mode.
- Check the voltage from the power point from where the supply is to be accessed. It should be between 90-265VAC, 47-63Hz. Insert the DC output terminal of power adapter into the 'Power' jack of the SETU VP236 and connect the AC Mains pins of the power adapter to a wall outlet for AC Mains. You can also get power using PoE if it is supported.
- When you power ON the SETU VP236, booting sequence will start and observe normal display on the LCD.

### **Basic steps for configuration**

#### **Network Port:**

- Default Network IP Address for SETU VP236 is 192.168.001.161

#### **ABSOLUTELY NEEDED !**

Use WAN IP Information from your LAN Admin. or ISP

- IP Address (must)
- Subnet Mask
- Gateway
- DNS IP Address
- DNS Domain Name

- 
- Note that subnet of WAN port, Internet (Router) and PC are same.
  - If not, you will be required to correct the IP address of the PC or of the SETU VP236.

### To correct IP address of the PC:

Read the IP address of PC, by clicking on Network Neighborhood → Properties → TCP/IP Connection → Properties for Windows98.

- Choose the unique IP address of the PC such that its subnet is same as subnet of IP address of SETU VP236, provided by ISP.

#### PLEASE CHECK !

The subnet of your PC IP Address must be same as of VP236.

- For example, if IP address of VP236 is 192.168.5.10, then PC IP Address port should be programmed as 192.168.5.X. where 'X' can be any number from 1 to 254, except 10. By doing this, subnet remains same.
- Now if you require to correct the default Static IP Address of the Phone, proceed as mentioned below:
  - Your phone may be with 'key pad locked'. Then while pressing the key, the Phone will ask "User Password". Enter the default password '**1111**'. If password is correct, the keypad is 'unlocked' and you can further operate the Phone.

Remember that this 'keypad unlock password is Not **same** as the one for entering to Web jeeves from the 'Welcome' page.

- Press the 'Menu' key (if assigned) of the Phone. If key is not assigned, select 'Menu' from the LCD display and press 'Enter' key. Now you are in Menu mode. Navigate the parameters with up/Down key. Get the Parameter "Configuring the Phone". After selecting, when you press the 'Enter' key, the Phone will ask the Password.

Remember that this Config Phone Password is same as password for entering Web-Jeeves.

- You can navigate the parameters by up/down key after entering correct Password (Default is 1234). Select 'Config the Phone' Menu

and navigate to 'SIP Para'. Press 'Enter' to get different programmable parameters for SIP1 account. Select "IP-Address" if you want to modify it. Press 'enter'.

- Similarly you can enter to 'Network Parameters' and select the WAN IP Address. For example, if you want to enter IP Address 192.168.1.120. enter digits as: 192111681100111120 ('11' is interpreted as '.' by the Phone).

This is known as Static IP-Address which is enabled by default.

- Now at Powering ON, if Connection type programmed in the Phone is 'DHCP' (DHCP enabled) then System will request following information from DHCP Server:
  - IP Address
  - Subnet Mask
  - Gateway Address
  - DNS Address (If DNS Server is set for DHCP)
- If DHCP Server provides IP Address successfully, the Phone will use this address instead of Static IP Address.
- Once the SETU VP236 has its own IP address, it shall further try for SIP registration or auto configuration if programmed. These parameters can not be edited by the user.
- Similarly if PPPoE is enabled, while powering ON the Phone, then System will request following Authentication information from PPPoE Server:
  - User ID
  - User Password
- Once the SETU VP236 has its own IP address, it shall further try for SIP registration or auto configuration if programmed. These parameters can not be edited by the user.
- If authentication is successful, Phone will use the IP address assigned by PPPoE server.
- After setting or getting correct IP Address, you can enter Web-Jeeves.
  - Start Internet Explorer (IE6 with SP2, Service Pack) of PC connected to the Ethernet Switch. Type in the IP Address programmed above in the Address field. On the login page, enter default password '**1234**'.
- To configure the SIP Parameters, proceed as mentioned below:
  - Click on "SIP Parameters" from the menu provided on left side of

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the web page.

**Get Information from your ITSP for SIP Account**

- SIP ID
- Registrar Server Address
- Registrar Server Port
- Authentication User ID
- Authentication Password
- Outbound proxy Server address
- Outbound proxy Server Port

- Enter above information in corresponding fields. If these information is not available, keep the values as default. For e.g. Registrar Server Port = 5060.
- Select 'enable' for Status-field for SIP1 and also enable Outbound Proxy, if information is programmed.
- If you have a second internet phone Service Account (SIP2), then use above fields for SIP2 port and enable.
- Submit page.

**Note:**

The PC need not be dedicated to the SETU VP236 after installation is complete.

- You can have a look at all above parameters and software version information by clicking on 'Status' from the menu. This page will show following parameters which the SETU VP236 is currently using:
  - IP Address
  - Subnet Mask
  - Gateway Address
  - DNS Address
  - MAC Address
- SETU VP236 is ready for further IP calling service!
- User can also use Auto Configuration feature if he wants to avail facility of configuration of SETU VP236 by ITSP from central location.
- Default the system and start again, if you cannot establish connectivity from your SETU VP236.

### **Making Phone Calls**

- After SIP1/SIP2 indication on the LCD, user can Go OFF-Hook, get dial tone and press digits, wait for 'Inter Digit Wait Timer" and the number will be dialed out, if keypad is not locked. If it is locked, the Phone will ask 'User Password". On entering correct password, phone is unlocked.
- User can initiate a call by three ways as mentioned below:
  - Using Handset
  - Using Speaker
  - Using Headset
  - Hot Pad dialing
- User can use any method of dialing but after making a call, he can switch between handset, headset and speaker mode.
- User can also make a call by using following features:
  - Using the Phone Book
  - Using the Speed Dial
  - Using Call Logs
  - Using a selective SIP trunk
- For Example, you can press 'SIP2' key and dial a SIP ID number like [ajay246@MatrixTeleSol.com](mailto:ajay246@MatrixTeleSol.com). For this follow the logic for entering alphanumeric characters.
- You can also dial SIP ID number like '4567' for making proxy call.

### **Receiving a Call**

- When you get Ringing indication on Phone, go OFF-Hook and talk by any of the ways mentioned below:
  - Using the Handset-Pick up the Handset → Start talking.
  - Using the Headset-Press the Headset key → Start talking.
  - If Auto-Answer enabled-Without pressing headset key, the call will be answered after some time (programmable) and will get connected to the remote user, after going to Headset Mode.
  - Using the Speaker-Press the Speaker key → Start talking.
  - Pressing the Call Appearance key.

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## **Section 2: Features and Facilities**



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## 3-Party Conference

### Introduction

- Matrix supports the advanced feature of the phone, '3-Party conference'. With this feature you can:
  - Conduct a 3-Party Conference.
  - Terminate a 3-Party Conference.
  - Toggle between two remote users.
  - Disconnect one of the parties and come to 2-Party Speech.
  - Disconnect your self and connect remote user1 and remote user2.
  - Call Hold is not supported during active conference.
- Conference begins when you press the 'Conference' key. The LED of the key will glow.

Conference is possible:

- From Menu
- Using a key, assigned

### How to use it?

#### Conducting a 3-Party Conference using 'Conference' Key:

- To initiate a conference, press the Conference key while in a conversation.
- After the user presses the Conference key, remote user1, will be placed on hold and user will get a new dial tone.
- The user can now dial the 2nd party's telephone number.
- When 2nd party answers the call, the conference will be established.
- When the user hangs-up, all parties will be disconnected.

#### Terminating a 3-Party Conference:

To terminate Conference, the user will disconnect by going ON-Hook. All other parties are released.

#### • **Toggle between two remote users:**

- The user will press the CA key for the remote user who has to be held. Pressing the CA key holds the respective remote user whereas the other remote user remains in speech with the remote user. Thus remote users can be put in speech alternately.

- **Disconnect one of the parties and come to 2-Party speech:**
  - Pressing Conference key during active conference prompts/offers the user a Menu. The user can select disconnection of remote user1 or remote user2 (This feature can be used for multiple conference).
- **Disconnect himself and connect remote user1 and remote user2:**
  - While in Conference, the user can press 'Transfer' key. This will put remote users in speech and user himself will be disconnected.

### Configuring:

Conference is possible:

- From Menu. Refer the topic '[Menu](#)'.
- Using a key, assigned. For this click on Web Page, "Key Programming".
- Assign a key with **Single Colour LED** for this feature.

### Important Points:

- Each conference call is logged as an independent call.
- If the user performs Call Mute, both the remote users will not be able to listen user's voice.
- Waiting call can be attended in the same way as in case of a normal call. Refer chapter '[Call Waiting](#)'.
- Conference can be performed even for Peer-to-Peer Configurations.

### Relevant Topics:

1. [Call Hold](#) 56
2. [Call Waiting](#) 72
3. [Call Appearances](#) 49

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## Anonymous Call Rejection

### Introduction

- Some Incoming calls are from some unknown party whose Name/ Number is not displayed on the Phone. This type of calls without CLI numbers or Name (Identity) is called Anonymous calls. Matrix SETU VP236 supports a feature to reject such calls.
- Using this feature user can allow such calls as well.
- Configurable using 'Web Page' and through 'Config Phone' Menu.

### How it works?

- This feature works for IC calls only.
- 'Anonymous Calls Rejection' can be set/canceled through Web Jeeves or SETU VP236 (also through 'Config Phone').
- If set, the anonymous calls will be rejected.
- The rejected calls due to this feature are stored in the 'Rejected Calls' List, along with the calls rejected due to other reasons like manual call rejection or due to DND, etc.

### Configuring:

Anonymous calls rejection-enable/disable can be configured through Web or Phone.

- To configure 'Anonymous calls rejection' through Web-Refer the topic '[Web Jeeves](#)' for more details. Click on Web Page "**User Settings**".
- To configure from phone, access Config Phone-Menu and select set/cancel.

### Important Point:

- Anonymous call rejection is given priority over Call Forward. If feature is set the anonymous call gets rejected before checking for any Call Forward status.

### Relevant Topics:

1. [User Settings](#) 152
2. [Know Your SETU VP236](#) 17

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## Auto Answer

### Introduction

- When the user is busy with important work then for any IC calls, the speech of calling party can be routed to the Headset Port by the phone. This type of feature is called 'Auto Answer'.
- Incoming calls will be connected, after the time period of 'Auto Answer Timer'. For example, in 'Call Centers' persons receive many calls. It is not practically possible to press a button or lift the handset for each incoming call. In such cases, this feature is useful using Headset.
- Auto Answer can be set/cancel:
  - From Web Jeeves
  - From Menu
  - Using a key assign
- Key assigned should be with single color LED for this feature.

### How it works?

- This feature works only when Phone is in Idle state.
- Auto-Answer status and timer are stored even when phone is powered OFF.
- Caller will be connected to the user after time of approx. "Auto-Answer Timer", which is programmable.
- The user will get a beep of 500ms On, 500ms Off and then both the parties are connected.
- The key to set/cancel the feature works in toggle mode.
- Pressing the key will set/cancel Auto Answer. When set, the LED is glowing on the key. When cancelled, the LED is turned OFF.
- If key for Auto-Answer is not assigned, the status of Auto-Answer whether Set/Cancel can be known from the Menu. When the Auto-Answer option is selected, the display (while scrolling menu), will show the status of Auto-Answer whether set or cancel.

### Configuring:

- Auto-Answer can be set/cancel:
  - From Web Jeeves
  - From Menu
  - Using a key

- 
- Auto Answer Timer is programmable. Range of Auto-Answer Timer is from 00 to 99 seconds. Programming the Auto-Answer Timer can be done through Web Jeeves and Phone Configuration. Program '00' for immediate answer.
  - Refer the topic 'Web Jeeves'. Click on Web Page 'System Parameters'.
  - To program Auto-Answer Timer from Phone access Menu to set/cancel the feature.
  - A key can be assigned for Auto-Answer set/cancel feature. Refer Web page, "Keys Programming".

**Important Point:**

- You should wear Headphone if auto answer feature is set.

**Relevant Topics:**

1. Receiving a Call 119
2. Menu 100
3. Key Assignment 87
4. System Parameters 144

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## Auto Call Back

### Introduction

- The user would not like to dial the busy station repeatedly. He wants that if the called user is busy then his call should be queued and should be made when the called user becomes free after receiving notification within specified period (Notified Timer). This feature is called 'Auto Call Back' (ACB).
- The feature can be configured by using the 'Key' assigned and notification timer is programmable by Web Page.
- The key assigned is only with single colour LED (Do not assign key number 06 to 09).

### How to use it?

#### **To Set/Cancel ACB:**

- If the called party is busy, press the key assigned for ACB or select an option in the Menu. The assigned key works in toggle mode.

### How it works?

- When user is getting busy tone /busy message, he can set ACB by pressing the ACB key or selecting an option in the Menu. When ACB is set, message will be displayed on LCD for 2 seconds and the LED on the key will be turned 'ON'.
- When the called party is free, it will send 'ready-for-call completion' notification (ACB Notification) and the user gets the 'Ring' indication for this. Now he can make a call by pressing the ACB key. LED on the key will be tuned OFF and ACB character on LCD will be cleared when the call is made. This is like a normal OG call made using SIP1, SIP2 etc.
- If another ACB is set during busy state before the first one getting served, the earlier ACB request will be canceled and new ACB will be applicable.
- User can cancel ACB by selecting an option in the Menu or pressing the ACB key before receipt of the notification (ring).When ACB is cancelled, the message will be displayed for 2 seconds and LED will be turned Off.

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**ACB Notification Timer:**

- If ACB notification is **NOT received** within this time:
  - ACB request for the particular number will be cleared
  - LED will be turned off.
  - ACB character will not be displayed on the LCD.
- If ACB notification is **received** within this time:
  - User will get Ring if he is On-Hook (or when he goes ON-hook) and will get dial tone if he goes OFF-Hook during ringing.
  - ACB LED will flicker 500msec ON 500msec OFF.
  - ACB character will flicker 500ms ON 500ms OFF on the LCD.

**Configuring:**

- Using phone, the feature can be set/cancel from 'Menu'. For key assignment, refer topic, [Key Assignment](#).
- "ACB Notification Timer" can be programmed by clicking on Web page "**System Parameters**".

**Important Points:****ACB Status:**

- When the ACB option is selected from 'Menu', the display will show the status of ACB whether set or cancel with a cursor on the relevant row.

**ACB in Power down mode:**

- ACB status and ACB timer will be stored while Power down. The timer will continue from where it had stopped.
- ACB request will not be stored while Phone is powered down. Hence the user should again 'set' the required feature of 'ACB' when the phone is powered up.

**ACB subscription from the remote peer (calling party):**

- The called party will get ACB subscription request from the remote peer only when no CA is free.
- If at least one CA is idle on receipt of ACB subscription, the ACB notification will be sent to the calling party.
- Maximum four callers can subscribe for ACB.

**Relevant Topics:**

1. [Key Assignment](#) 87
2. [System Parameters](#) 144

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## Auto Configuration

### Introduction

- SETU VP236 configuration includes some basic settings like Registrar Server Address, Authentication user ID, and User Password which are provided by the Service Provider (ITSP). Also some special services like Call Forward are provided by the ITSP. Hence, it is desirable that the SETU VP236 is configurable by the ITSP from the central location. Matrix SETU VP236 supports this by providing Auto Configuration feature.
- The ITSP will keep the folders of configuration files in Central Server for different SETU VP236 units as per MAC Address number of the SETU VP236. When the customer connects the SETU VP236 to the ITSP's network, the SETU VP236 gets automatically configured and the customer starts using the services.

### How it Works?

- SETU VP236 makes use of TFTP (Trivial file Transfer Protocol) server to download the configuration files.
- User has to just enter the server address provided by ITSP and specify the config folder path provided by ITSP, in the Web page for 'Auto Configuration'. Then after selecting option for using the feature at power ON, Click on "submit" the page.
- Three options for using the feature are provided:
  - User can select that the SETU VP236 is not to be configured.
  - User can select that the SETU VP236 is to be configured, whenever it is switched ON.
  - User can select that the SETU VP236 is to be configured, when switched ON next time.
- SETU VP236 can use the address of TFTP server provided by the DHCP or static IP Address as programmed by ITSP.
- If the FTP server address is not found, the auto-configuration process is aborted and the current files are used by the SETU VP236.
- If all the required files are not available in the folder, then there will be incomplete configuration download to the SETU VP236. Also only those downloaded files whose version will match to the

- 
- version of SETU VP236 will be updated.
  - For example, 7 configuration files are necessary and if only 5 files are present, then these 5 files only will be downloaded by the IP Phone and will be updated in SETU VP236 which may not meet the required configuration of the SETU VP236.
  - When Auto Configuration is complete, the '**Status**' page displays following information:
    - In 'Status' field suitable message is displayed such as:
      - “**No Tftp Server Found**”
      - “**Auto Configuration Partial**”
      - “**Auto Configuration Success**”
    - In 'Statistics' field, suitable message is provided regarding how many files are down loaded successfully and total number of configuration files updated by the system. For example, if X=Number of files down loaded and Y=Total number of files updated. Then the message displayed will be as:
      - “**Files Downloaded/Configured = X/Y**”

### **How to program?**

Click on 'Auto Configuration' from the menu on the 'Welcome Page'.

### ***Auto Configuration:***

## Matrix

Parameter	Value/Default	Description
At Next Power ON	Default = Selected.	Select this option if you want to Auto Configure your IP Phone when you power it ON next time.
At Power ON	Not Selected	Select this option if you want to Auto Configure your IP Phone whenever you power it ON.
Never	Not Selected	Select this option if you do not want to Auto Configure your IP Phone.
TFTP Server	DHCP Static Default = DHCP	Select whether you will use TFTP Server address provided by DHCP or Static provided by ITSP.
TFTP Sever Address	N.N.N.N, 40 characters Max. N = 001 To 255 Or Domain Name Default = Blank .	Enter here the server address provided by ITSP. Do not enter if it is to be allocated by DHCP server, on request by IP Phone.
Config Folder Path	40 Characters max. Default = Blank.	Provided by ITSP. Enter here the path of the folder in which the configuration files are stored. For e.g. Matrix/SETU VP236.

### Important Point:

- SETU VP236 will not restart, if user clicks on ‘Submit’ on this page.

### Relevant Topics:

1. [Status](#) 138
2. [Software Upgrade](#) 134

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## Black Listed Caller's List

### Introduction

- Using this feature it will be possible to reject incoming calls from specific addresses or contacts.
- List is configurable using 'Web Page' and using phone.

### How to use it?

- The caller's number which is to be rejected is programmed in black listed caller's list.

### How it works?

- This feature is applicable for IC calls only.
- System supports a list with maximum 10 addresses (10 Entries).
- The number of calling party will be matched upto the last character with the entries in the list.
- The rejected calls due to this feature is stored in the 'Rejected Calls' list along with the calls rejected due to other reasons like manual call rejection or due to DND, etc.
- If DND is set for "DND List" and calling party address is present in both "DND List" and "Black Caller List", the system will not obey the DND List and will reject the call.

### Configuring:

- Numbers can be entered in the list by following ways:
  - Refer Web Jeeves and click on "**User Settings**". For more details refer chapter '[User Settings](#)'. **By default, the list is Blank.**
- The list can be configured from the Phone:
  - Navigating Menu > Config Phone > Black Listed Caller List.

### Important Points:

- Anonymous call gets rejected before checking for black callers list.
- Black listed caller is given priority over Call Forward. The call gets rejected before checking for Call Forward status.

### Relevant Topics:

1. [User Settings](#) 152

## Matrix

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2. Configuring the Phone-Menu 75
3. Anonymous Call Rejection 39

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## Call Appearances

### Introduction

- Matrix Phone supports four calls which are separate or joined. That means if during an ongoing call a second incoming call comes there will be an indication about that call and user can put the first call on Hold. Thus, there are two active calls which user can select. These two calls can be indicated on the LCD display. Now if third and fourth calls will coming then last two calls will also be indicated by the phone as 'waiting calls', Thus, we can have indications for 'four' calls maximum. This is known as Call Appearance (CA) for the SETU VP236.
- Matrix SETU VP236 supports maximum of '4' Call Appearances by default. This is programmable.
- Similarly for OG calls also CA is applicable.
- The state of the call will be indicated on the corresponding key for the CA.

### How it works?

#### **Selecting CA:**

- The user can attend any call by switching between these four calls.

#### **Assigning a Key:**

- Indication on the Key-LED is updated after answering the CA.
- Keys can be assigned for each Call Appearance. If keys are not assigned for Call Appearance then features like answering a call selectively, answering a waiting call, toggling between two calls, rejecting a call is not possible.
- ***By default, Keys 06-09 are used for Call Appearances.*** These are dual color keys.

Key Number	Feature
09	Call Appearance 1
08	Call Appearance 2
07	Call Appearance 3
06	Call Appearance 4

Refer the topic '[Key Assignment](#)' for more details.

### **Programming Number of CA:**

- Number of Call Appearances can be programmed. For example, if the user programs say, 2 then normal display will show the information of only first two CA.
- If user has configured only two CA and gets a third IC call then in such a case, it will be automatically rejected by the SETU VP236 so that user is not prompted to select that call.
- One Call Appearance is occupied for each IC call received/OG call made. This will happen in increasing order viz. CA1, CA2, CA3 followed by CA4.

### **For Example:**

Four CA can be used for four simultaneous calls.

- Consider a case for detailed explanation:
  - Suppose that user is busy with two calls: First one is an OG call occupying CA1 and second one is an IC call occupying CA2. The user is talking over CA1, whereas CA2 is held.
  - Now, if the user tries to make third call, then current call will go on hold and user will get dial tone.
  - If the user gets another IC call, it is treated as waiting call and is indicated by CA4.

### **Call Toggle:**

- Suppose there are two calls, one active and the other held. The user can toggle between these two callers just by pressing a key. Refer the chapter '[Call Toggle](#)'.
- Since we support four CA's, four calls can be handled simultaneously.

### **LED Indication:**

- The status of the call (Call state) is indicated on the LEDs embedded in the keys assigned to the CA feature. Refer '[Getting Started](#)' for more details.

### **How to use it?**

- Click on Web Page “Key Programming”. Refer chapter ‘Key Assignment’ for more details.
- Program the keys for CA1, CA2, CA3 and CA4. Default is 06 to 09 for CA4 to CA1 respectively.

**Important Points:**

- It is preferable to program the key for “Rejecting a Call” as this will be useful to reject the CA if required.
- Do not press the CA-Key and another key simultaneously. The SETU VP236 can read only a single key being pressed at a time.

**Related Topics:**

1. Call Waiting 72
2. Making a Call 94
3. Receiving a Call 119
4. Key Assignment 87
5. Menu 100

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## Call Forward

### Introduction

- Matrix SETU VP236 supports many supplementary services features like a normal Phone. For example: Call Hold, Call Forward (CF), Call Transfer, etc.
- The phone supports Call Forward-Busy, Call Forward-No Reply and Call Forward-All.
- **Call Forward-Busy:** Incoming calls on a user can be forwarded to another designated party if the user is busy.
- **Call Forward-No Reply:** Incoming calls on a user can be forwarded to another designated party when there is no response from the user.
- User can forward his calls to one destination number-when busy and to another destination number-when no reply.
- Separate key can be provided for Forward-Busy and Forward-No Reply, feature for the phone.
- User can get the 'status' of Call Forward whether set or cancel.

- Call Forward can be set/cancel:
  - From Menu
  - From Web
  - Using a key.
- Keys with dual color LED should NOT be assigned for Call Forward. Key with single colour should be assigned.

### How to use it?

#### To set Call Forward:

- The user can press the pre-assigned Call Forward key (Busy or No Reply).
- Navigate the menu using Up/Down keys. Select the option 'Set'. Enter the destination number to which the calls are to be forwarded. Press enter key after entering the number.
- LED on the key will be ON when Call Forward is set.

#### To cancel Call Forward:

- The user can press the preassigned Call Forward key (Busy or No Reply).

- 
- Navigate the menu using Up/Down keys. Select option 'Cancel' press enter key.
  - LED on the key will be turned 'OFF' when feature is 'Cancel'.

**Call Forward Timer:**

- This timer is applicable for Call Forward-No Reply only.
- All the calls are forwarded to programmed destination number after time period of 'Call Forward Timer'.
- 'Call Forward Timer' can be programmed from configuring phone menu and Web page, also.
- You can set Call Forward-All, by selecting the Call Forward Timer to '0'.

**Fox Example:**

- If destination number programmed for Call Forward-No Reply is 7662 (SIP ID) and timer is 30 seconds, then system will wait upto 30 seconds. If the phone doesn't reply, then the IC call will be forwarded to 7662.

**Configuring:**

- Call Forward can be configured:
  - From Menu (refer chapter-'[Menu](#)')
  - From Web page
  - Using a key (refer chapter [Key Assignment](#))
  - After pressing the key, you will get a Menu for set/cancel. Select 'set' to configure the feature.
- Call Forward Timer can be programmed by:
  - Configuring the Phone Menu (refer chapter)
  - Web Page
- Click on Web page "**User Setting**" for programming through web page.

Parameter	Value/Default	Description
<b>Call Forward-when Busy</b>	Select/Blank	Select if you want to use the feature.
	Default = Blank or ASCII characters max. 40	Enter here the destination number to which you want to forward the IC call. If your phone is busy.
<b>Call Forward-when No Reply</b>	Select/Blank	Select if you want to use the feature.
	Default = Blank or ASCII characters max. 40	Enter here the destination number to which you want to forward the IC call. If you cannot go OFF-Hook.

- Click on Web page, “System Parameters”. Refer chapter ‘[System Parameters](#)’.

Parameter	Value/Default	Description
<b>Call Forward Timer</b>	00-99 seconds, Default = 45 seconds	Enter the timer value after which you want the Phone to forward he number. If programmed call forward timer is 00 sec and call forward-no reply is set, and then all the incoming calls will be forwarded to desired destination number immediately.

#### Important Points:

- Call Forward set/cancel status and the destination addresses are retained during power down, of the phone.
- Call Forward-No Reply is applicable for waiting call also.
- Black Caller list is given priority over Call Forward. The call gets rejected before checking for Call Forward, if the number is in the list.
- Call Forward is given priority over DND.
- Each forwarded call is logged, in the system.

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**Relevant Topics:**

1. Do Not Disturb (DND) 83
2. System Parameters 144

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## Call Hold

### Introduction

- The user can support multiple calls by using the 'Hold' feature. This feature allows you to keep a call on hold and retrieve the call when required. It also allows making second call/answer a waiting call while holding a call.
- This feature is useful for various requirements for using the phone with multiple calls, as explained below:
  - User can 'Hold' Maximum of four calls and can toggle between calls as per requirement. If number of CAs are 4, four calls can be made/received simultaneously.
  - User can retrieve the held call.
  - User can make second call after holding a call.
  - User can receive a waiting call after holding the current call.
  - User can disconnect a held call.
  - User can be reminded of held call.
  - The held call will be disconnected, if not retrieved before programmed time.
  - 'Hold' can be accessed by navigating the features of 'Menu' or by assigning the feature to a key using Web page 'Key Programming'.
  - **Select the key with No LED.**

### How to use it?

- To hold a remote user during speech, press 'Hold' key or press the CA key of the active call.
- To retrieve the held remote user, press respective CA key.
- Call Hold and Retrieval can be done from 'Menu' also.

### How it works?

#### *Holding a Call*

- It is possible to hold and retrieve a call when the phone is in speech (connected)
- In all other states, call hold request is ignored.
- Held call with Headset or Speaker:
  - Phone will go to idle state.

- 
- The Speaker/Headset will go silent
  - The CA key blinks as per the cadence mentioned in the topic '[Call Appearances](#)'.
  - The LCD display will display the caller ID and the text 'Hold'.
  - Held call with Handset:
    - User will get silence
    - The user will go ON-Hook
    - Phone will be Idle
  - Indication of Held Call:
    - The user gets silence.
    - He is ready to do any other activity.
    - If the call is through handset, the user may go ON-Hook.
    - Held call is displayed on the LCD.

### ***Retrieving Held Call***

- User gets information about call retrieval by following ways:
  - User gets connected to the held party.
  - Held call will be changed to active call and hence the LCD display will display the call as a normal call.
  - If the call is retrieved while in speech with other remote user, the current remote user will be held.

### ***Alert for 'Hold' and Timer:***

- 'Call Hold Alert Timer' is not separate for each call held.
- User will not get any hold alert if the phone is off-hook.
- If user goes on-hook with at least one held call, 'Call Hold Alert Timer' will be started. Before the expiry of 'Call Hold Alert Timer', if user goes off-hook, he will get dial tone of next free CA. But if no CA is idle, he will get error tone.
- If user doesn't go off-hook till the expiry of 'Call Hold Alert Timer', he will get a ring to up to the time period of Ring Timer and all the held calls are disconnected.
- User will be connected with the held call on lowest CA, if there are multiple held calls and user goes off-hook before the expiry of ring timer.
- Matrix SETU VP236 has a facility to Alert the user by following ways:
  - The Caller ID will flash for 500ms ON, 500ms OFF.
  - LED of the respective CA will Glow/Blink. Refer the topic '[Call](#)

Appearances' for LED blinking cadence and color.

**Rejecting Held Call and Timer:**

- Held call rejection is indicated to the user in following ways:
  - The row which occupied the held call is cleaned and the contents are shifted to any of next rows upwards.
  - Turning OFF the LED of the respective CA. Refer the topic 'Call Appearances' for LED blinking cadence and color.

**Configuring:**

- 'Call Hold Alert Timer' is programmable only through Web Jeeves. Click on Web Page 'Timers' for this. Refer chapter 'Timers'.
- Range of 'Call Hold Alert Timer' = 000 to 999 seconds.
- Assigning 000 second to Call Hold Alert timer will mean that these timers are not applicable. Refer Web pages for more details.

**Remote Held:**

When remote user puts you on 'Hold' it is called 'Remote Held call'. During this you can not 'Hold' that remote user. If you go ON-Hook, when on same CA of remote held call, the call will be disconnected.

- But during Remote held call you can:
  - Make a new call by-Pressing respective CA key.
  - Receive IC Call by-Pressing respective CA key.
  - Disconnect the call if you go ON-Hook with one Remote Held call.

**Indication of Remote Held Call:**

- You can get indication of remote held call by:
  - Getting beeps.
  - Getting Display on LCD.
  - Indication of blinking of Respective CA LED.

**Note:**

- If there are multiple remote held calls and user goes OFF-Hook before the expiry of ring timer, the user will be connected with the remote held call on lowest CA.
- Remote held Call will be Rejected if user presses 'Reject' key.

**Resuming Remote Held Call:**

- In case the remote user resumes the 'Held' call, you can get

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indication when you are in ON-Hook condition by:

- Getting Continuous Ring
- Display of flashing 'Rhold' of respective CA.
- Indication of blinking respective CA LED.
- In case the remote user resumes the 'Held' call, you can get indication when you are in OFF-Hook condition by:
  - Getting beeps. like tone. This is called 'Remote Held Resuming Tone'. It is same as CCWT.
  - Display of flashing 'Rhold' of respective CA.
  - Indication of blinking respective CA LED.

### **Retrieving Held Call:**

- In case the remote user retrieves the 'Held' call, you can get indication by:
  - Getting connected to the remote party and the Remote held call will be changed to 'Normal' call and you will get 'Display' on LCD for this.

### **Important Points:**

- The user can answer the held call by pressing the respective CA key. This might lead to Call Toggle. Please refer '[Call Toggle](#)' for more details.
- This feature is related with other features as mentioned below:
  - Call Transfer: A held call can be transferred.
  - Peer-to-Peer Calling : P2P can also be held.
  - Rejecting a Call : It is possible to reject a held call.

### **Relevant Topics:**

1. [Call Toggle](#) 68
2. [Call Appearances](#) 49
3. [Call Waiting](#) 72
4. [Receiving a Call](#) 119
5. [Making a Call](#) 94

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## Call Logs

### Introduction

- This feature is provided to know the history of calls. With this facility you can view:
  - Last 20 dialed calls.
  - Last 20 answered calls.
  - Last 20 unanswered calls.
  - Last 20 rejected calls.
  - Last 20 forwarded calls.
- It is also useful to dial the number after viewing such numbers from the Call Logs menu. The log will be as shown at end of chapter.
- This feature can be assigned a key with No LED. Feature can be viewed on the Web Page.

### How to use it?

- To view the log, press the key for 'Call Logs' or select 'Call Logs' option from the menu to view all the dialed numbers, select the option 'Dialed'. To view all the calls which are answered, select the option 'Answered' etc.
- Using the phone you can scroll the 'Menu', and select 'Call Log' after selecting the type press 'Enter' key to see call details. Use < key to go to previous menu. Finally to dial the selected number, go OFF-Hook or press 'Speaker' key or press the 'Headset' key.

### How it works?

- The call details can be viewed in brief or in details from the 'Menu' or using a key.
- Feature works during all call states of the phone while receiving, or dialing etc.
- System will retain the Call Log even during power failure also.
- Call log contains following parameters for a call:
  - Number
  - Name of the called party
  - Details of the call about Date and Time
  - Duration of the call
- On pressing 'Call Logs' key or selecting the Call Logs option from the 'Menu', a list of options for the call type is available:

- Dialed
- Answered
- Unanswered
- Rejected
- Forwarded
- For any option, the details of Name, Number and Date and Time whichever is available, is displayed on the LCD.
- If Name or Number (in case of calls other than dialed ones) is not available for a call, it will be displayed as 'ANONYMOUS'.

**Dialed Number:**

- All the dialed numbers, for making calls are stored.

**Answered Number**

- The Number is considered as an 'Answered' number when the user answers the number of the received call. The Number is stored at the end of the call.

**Unanswered Number**

- The Number is considered as an 'Unanswered' number when the user does not answer it till ring time is expired. The number is stored at the end of Ring Time Out Timer.

**Rejected Number**

- The number is considered as a 'Rejected' number when it is rejected by the user manually. The numbers rejected due to lack of CA or DND or Black Listed Caller's logic are also stored as 'Rejected' number.

**Forwarded Number**

- The number is considered as a 'Forwarded' number when it is forwarded by the phone as per the configuration setting of the Phone. This number is stored after receiving the call forward notification.

**Dialing a Number**

- Go OFF-Hook/press 'Speaker' key/press 'Headset' key from dialed numbers' list (when displayed in brief or in detail) to make an OG call or second call.

### Configuring:

- User can assign a key for this feature. Refer web page 'Keys Programming'.
- 'Call Log' feature can also be selected from 'Menu'.

### How to use Web Page?

Click on Web Page, "**Call Logs**" to view the log.

- The parameters of the Log for each call are as shown below:
  - **Name:** Name of the calling party. If name is not available, 'Anonymous' is seen in place of name. If call is not anonymous, it is 'blank'.
  - **Number:** Here calling party's number is displayed. If call is anonymous, it is blank.
  - **Date:** Date of the call is displayed in DD-MM-YYYY format.
  - **Time:** Time of call is displayed as HH-MM, in 24 hour format. For example, call at afternoon 3.00 hours will be displayed as 15:00.
  - **Duration:** It is a total duration of the call, starting from the time of call maturity, in the format: HH:MM:SS.
- You can select the type of call for which log is required. For example, if you click on "**dialed**", the phone will display the log of all 'dialed numbers'. Similarly you can click on: **Answered, Unanswered, Forwarded and Rejected** calls.
- **Clear all:** Click on this to delete all the entries in call logs of selected type. If selected log is for 'dialed', you will find the alert message "This will delete all dialed entries". Similarly, alert message is found for other types.

### Important Point:

- All the calls of the 'Conference Call' are stored as independent calls.

### Relevant Topics:

1. [Menu](#) 100
2. [Know Your SETU VP236](#) 17

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## Call Mute

### Introduction

- While you are in speech, during active call, you can hear the second party but second party can not hear you. This feature is called “Call Mute”.
- Call Mute can be set/cancel:
  - From Menu
  - Using a key
- The key to be assigned is with single color LED.

### How to use it?

- To Mute the Call:
  - Press the Call Mute key if assigned or use ‘Call Mute’ option from the Menu.
  - Press the Mute key/select ‘Set Call Mute’ option from Menu
  - The call will mute and LCD will display ‘Call Mute Set’.
  - The remote user will not be able to listen you.
- Press the Mute key again/selecting ‘Cancel Call Mute’ option from Menu, call will unmute and LCD will display ‘Call Mute canceled’. Now the remote user will be able to listen you.

### How it works?

- This feature is applicable only for an active current call.
- When a call ends, Call Mute is canceled and not applicable for next call.
- When Call Mute is set:
  - OG speech is disconnected and the remote user will hear silence (and not noise).
  - LCD will display a text message for 2 seconds and come back to the previous state.
- When Call Mute is canceled:
  - OG speech is connected back and the remote user can hear the user.
  - LCD should display a text message for 2 seconds and come back to the previous state.

**Configuring:**

- For key assignment, refer web page 'Keys Programming'.

**Relevant Point:**

1. Key Assignment 87

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## Call Progress Tone Generation (CPTG)

### Introduction

- Different types of tones are played during specific event of establishing the call, speech duration and termination of the call. This is called Call Progress Tone Generation (CPTG). As per ITU standard these tones are of different specification for different country. For example, Dial Tone for India consists of 400Hz modulated by 25Hz, but for USA/Canada, it is 350+440Hz, without modulation. Based on specific frequency, modulating frequency and cadence, tones are categorized as shown below:
  - Dial Tone
  - Ring Back Tone
  - Busy Tone
  - Error Tone
  - CO Call Waiting Tone (CCWT)
- Matrix SETU VP236 supports CPTG for many countries.
- SETU VP236 plays the Dial Tone when it is ready to accept the first digit of a remote address to make an outgoing call. Normally played when FXS goes OFF-Hook.
- SETU VP236 plays the Ring Back Tone when the called device is alerting the callee.
- SETU VP236 plays the Busy Tone when the callee is busy.
- SETU VP236 plays 'Error Tone' when, there is no registration with proxy server or user has performed some wrong operation.
- SETU VP236 plays the 'CCWT' only for first incoming call (Waiting Call) which arrives while the user is connected to another party.

### How it works?

- SETU VP236 supports call progress tones for various countries.
- Frequency and Cadence of "Call Progress Tone" for a country is fixed (not programmable). Refer the table at the end of the chapter for details of the frequency and cadence for different CPTG for countries.

### How to program?

- Click on Web Page, "System Parameters" Refer [Web Jeeves](#).
- Refer chapter "[System Parameters](#)" for CPTG programming.

## Matrix

- Select Country: Click on combo box and select the 'Country Name' from the list of following countries:

Country	Dial Tone		Ring Back Tone		Busy Tone		Error Tone		CCWT	
	Freq.	Cadence	Freq.	Cadence	Freq.	Cadence	Freq.	Cadence	Freq.	Cadence
	Hz	second	Hz	second	Hz	second	Hz	second	Hz	second
Australia	425*25	cont.	400*25	0.4on 0.2off 0.4on 2.0off	425	0.375on 0.375off	425	0.375on 0.375off	400	0.3on 6.0off
Argentina	425	cont.	425	1.0on 4.0 off	425	0.3on 0.2off	425	0.3on 0.4off	400	0.3on 6.0off
Brazil	425	cont.	425	1.0on 4.0 off	425	0.25on 0.25off	425	0.25on 0.25 off	400	0.3on 6.0off
China	450	cont	450	1.0on 4.0off	450	0.35 on 0.36off	450	0.7on 0.7off	400	0.3on 6.0off
Egypt	425*50	cont	425*50	2.0on 1.0off	425*50	1.0on 4.0off	450	0.5on 0.5off	400	0.3on 6.0off
France	440	cont	440	1.5on 3.5off	440	0.5on 0.5off	440	0.25on 0.25off	400	0.3on 6.0off
Germany	425	cont	425	1.0on 4.0off	425	0.48on 0.48off	425	0.24on 0.24off	400	0.3on 6.0off
Greece	425	0.2on 0.3off 0.7on 0.8off	425	1.0on 4.0off	425	0.3on 0.3off	425	0.15on 0.15off	400	0.3on 6.0off
India	400*25	cont	400*25	0.4on 0.2off 0.4on 2.0off	400	0.75on 0.75off	400	0.25on 0.25off	400	0.3on 6.0off
Indonesia	425	cont.	425	1.0on 4.0off	425	0.5on 0.5off	425	0.25on 0.25off	400	0.3on 6.0off
Iran	425	cont.	425	1.0on 4.0off	425	0.5on 0.5off	425	0.25on 0.25off	400	0.3on 6.0off
Israel	400	cont	400	1.0on 3.0off	400	0.5on 0.5off	400	0.25on 0.25off	400	0.3on 6.0off
Italy	425	cont.	425	1.0on 4.0off	425	0.5on 0.5off	425	0.2on 0.2off	400	0.3on 6.0off
Japan	400	cont	400*20	1.0on 2.0off	400	0.5on 0.5off	400	0.25on 0.25off	400	0.3on 6.0off
Kenya	425	cont	425	0.67on 3.0off 1.5on 5.0off	425	0.2on 0.6off 0.2on 0.6off	425	0.2on 0.6off	400	0.3on 6.0off
Korea	350+440	cont	440+480	1.0on 2.0off	480+620	0.5on 0.5off	480+620	0.3on 0.2off	400	0.3on 6.0off
Malaysia	425	cont	425	0.4on 0.2off 0.4on 2.0off	425	0.5on 0.5off	425	0.5on 0.25off	400	0.3on 6.0off
Mexico	425	cont.	425	1.0on 4.0off	425	0.25on 0.25off	425	0.25on 0.25off	400	0.3on 6.0off
New Zealand	400	cont.	400+450	0.4on 0.2off 0.4on 2.0off	400	0.5on 0.5off	400	0.25on 0.25off	400	0.3on 6.0off
Philippines	425	cont.	425+480	1.0on 4.0off	480+620	0.5on 0.5off	480+620	0.25on 0.25off	400	0.3on 6.0off
Poland	425	cont.	425	1.0on 4.0off	425	0.5on 0.5off	425	0.5on 0.5off	400	0.3on 6.0off
Portugal	425	cont.	425	1.0on 5.0off	425	0.5on 0.5off	450	0.33on 1.0off	400	0.3on 6.0off
Russia	425	cont.	425	0.8on 3.2off	425	0.4on 0.4off	425	0.25on 0.25off	400	0.3on 6.0off
Saudi Arabia	425	cont.	425	1.2on 4.6off	425	0.5on 0.5off	425	0.25on 0.25off	400	0.3on 6.0off
Singapore	425	cont.	425*24	0.4on 0.2off 0.4on 2.0off	425	0.75on 0.75off	425	0.25on 0.25off	400	0.3on 6.0off
South Africa	400*33	cont.	400*33	0.4on 0.2off 0.4on 2.0off	400	0.5on 0.5off	400	0.25on 0.25off	400	0.3on 6.0off
Spain	425	cont.	425	1.5on 3.0off	425	0.2on 0.2off	425	0.25on 0.25off	400	0.3on 6.0off
Thailand	400*50	cont.	400	1.0on 4.0off	400	0.5on 0.5off	400	0.3on 0.3off	400	0.3on 6.0off
Turkey	450	cont.	450	2.0on 4.0off	450	0.5on 0.5off	450	0.2on 0.2off 0.6on 0.2off	400	0.3on 6.0off
UAE	350+440	cont.	400+450	0.4on 0.2off 0.4on 2.0off	400	0.375on 0.375off	400	0.4on 0.35off 0.225on 0.525off	400	0.3on 6.0off
UK	350+440	cont.	400+450	0.4on 0.2off 0.4on 2.0off	400	0.375on 0.375off	400	0.4on 0.35off 0.225on 0.525off	400	0.3on 6.0off
USA/Canada	350+440	cont.	440+480	2.0on 4.0off	480+620	0.5on 0.5off	480+620	0.25on 0.25off	400	0.3on 6.0off

**By default, Country is India.**

### Remarks:

Frequency notations are as:

- f1\*f2: f1 is modulated by f2.
- f1+f2: The juxtaposition of two frequencies f1 and f2 without modulation.

**Important Point:**

- Rx volume of Rx Speech tones can be increased/decreased by using left and right arrow key on the phone.

**Relevant Topic:**

1. Volume Settings 160

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## Call Toggle

### Introduction

- It is often required for the user to toggle between an active on going call and a held call.
- For this Matrix SETU VP236 provides the feature called 'Call Toggle'.
- There is no extra key assignment for this feature.

### How it works?

- This feature is a just an extension of multiple CA.
- While answering/retrieving a call, the active call will get held and the user will get connected to the remote party.

### For Example:

- Suppose CA1 has an active call whereas, CA2 has a held call. Now if you wish to talk to held call on CA2, just press the key for CA2.
  - This will act as retrieving a call since CA2 has a held call.
  - The current call i.e. on CA1 will get held and CA2 will be active.
- User can toggle between two calls during call conditions: Ringing or Matured.
  - User will not be able to toggle between calls in any other states.
  - Whenever a user is held while toggling, all the actions applicable for a held call like Call Hold Alert Timer, etc. are applicable. For this refer chapter 'Call Hold'.

### Relevant Topics:

1. Call Appearances 49
2. Making a Call 94
3. Making Second Call 99
4. Receiving a Call 119
5. Call Hold 56

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## Call Transfer

### Introduction

- As three parties are involved for this feature, following terms are used to explain the feature:
  - **Transfer Target:** The remote user to whom the call is to be transferred. (1<sup>st</sup> party).
  - **Transferor:** The user who transfers the call.
  - **Transferee:** The user who is to be transferred (2nd party)
- Call Transfer is possible:
  - From Menu
  - Using a key (without LED)
- “Transfer Notification Timer” to provide information about ‘Transfer’ to the user is programmable from Web page “System Parameters”.
- Matrix SETU VP236 supports two types of Call Transfer:
  - Unattended (Blind) Transfer
  - Attended Transfer

### Unattended or Blind Transfer

- In this type the user can transfer the active call to the transfer target irrespective of his state of calling. User need not first talk to him before transferring.
- The user will get notification for the status of transfer.

### Attended Transfer

- For this type of transfer, user will first talk to the transfer target party before transferring the call without the transferee party being able to listen.
- While talking to transfer target, if the user presses ‘Transfer’ key, call will be transferred, but CA2 will not be released. The port will be idle.
- The CA1 and CA2 will display the transferee and transfer target number till the expiry of transfer notification timer OR 5 seconds after successful notification ‘Conc’ received from transferee.
- ‘Xfer1’ will be displayed on CA1 and ‘Xfer2’ will be displayed on CA2 for the cadence: 500ms ON 500ms OFF.
- The system will wait for the notification from the transferee. If notification is not received within transfer notification time, both the CAs will be released.

## How to use it?

### Unattended or Blind Transfer

- To initiate an unattended or blind transfer:
  - Press the 'Transfer' key while in a conversation.
  - The 1st call will be placed on hold and you will get a new dial tone.
  - Dial the 2nd party's telephone number.
  - The call will be transferred after Inter Digit Wait Timer time.
  - You will be notified by Remote user1 about the transfer action i.e. failure/successful provided remote user1 supports notification.

### Attended Transfer:

- To initiate attended transfer:
  - Press the 'Hold' key while in a conversation.
  - The 1st call will be placed on hold and the user will get a new dial tone, when selecting second CA after pressing CA key.
  - You can now dial the 2nd party's telephone number. When the 2nd party answers, you can have a private conversation with the 2nd party without the 1st party being able to listen.
  - If the 2nd party wants to speak with the 1st party. Press Transfer key to complete the call transfer.
  - You will drop out of the conversation with both parties, but the call will not be disconnected, and you will get suitable notification.

### Transfer Notification:

After transferring the call, the user will get information or notification about the status of transfer. This will be displayed for a period of some time known as "Transfer Notification Timer".

- User will get display of the transfer target's number on the CA on which transfer activity is done.
- The display will be for the time of Transfer Notification Timer.
- If notification is not received within transfer notification time, CA is released.
- **Call transfer notification (Display) (Blind Call Transfer)**
  - Transfer target busy: 'Busy'.
  - Transfer target ringing: 'Free'.
  - Transfer target ringing over; 'Disc.'
  - Transfer target answered: 'Conc.'
  - Transfer target error: 'Error'. Now if the user goes ON-Hook in

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this state, the user will get 'Ring' indication. When the user goes OFF-Hook, remote user1 is retrieved and connected to the user.

- **Call Transfer Notification (Attended Call Transfer)**
- Call transfer notification is provided for following cases of transfer target :
  - Transfer target busy
  - Display will be 'Busy'. The CA2 (transfer target) will be released immediately after notification is received and CA1 will be released after the expiry of transfer notification time.
  - Transfer target answered.
  - Display will be 'Conc'. The CA1 will be released after 5 seconds and CA2 will be released immediately after 'Conc' notification is received.
  - Transfer target error.
  - Display will be 'Error' till transfer notification timer.
  - Now if the user goes ON-Hook in this state, user will get Ring. When the user goes OFF-Hook, the transferee will be retrieved and will be connected to the user (transferor). The CA1 will be released after the expiry of notification time and the CA2 will be released immediately after 'Error' notification is received.

### **Configuring:**

This is a feature that does not need any configuring except assigning a key for it.

- Call Transfer can be done:
  - From Menu
  - Using a key
- If key for Call Transfer is not assigned, the user can select 'Call Transfer' from Menu.
- Key with no LED can be assigned for this feature. For key assignment, refer topic, '[Key Assignment](#)'.
- "Transfer Notification Timer" is programmable from Web Page "System Parameters".

### **Relevant Topics:**

1. [Call Hold](#) 56
2. [Key Assignment](#) 87

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## Call Waiting

### Introduction

- Normally, the user will come across many calls. Some calls may be coming to him at the same time when he is busy with an active call. The Phone supports to handle these calls by putting them in the 'Waiting' state.
- Matrix SETU VP236 supports the IC call as Waiting Call with following conditions. For all the condition, if user accepts the waiting call, the active call will be held.
  - The user has one active call.
  - The user has one held and one active call.
  - The user has one held, one active and one waiting call.
- The new IC call received with one held, one active and two waiting call is not considered as waiting call and will be rejected.

### How it works?

- A waiting call can be answered by pressing a respective CA keys
- After answering a waiting call the active call will be held.
- Number of waiting calls depends on the number of CA programmed.
- The user can answer the waiting call by pressing the respective CA key. This will be like Call Toggle. Please refer 'Call Toggle' for more details.

### Relevant Topics:

1. Call Hold 56
2. Call Appearances 49
3. Call Toggle 68
4. Call Progress Tone Generation 65

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## Change Password

### Introduction

- The SETU VP236 configuration is user specific. Hence user should be able to select his personnel password to replace the default password.
- User can change both types of password from its default value:
  - **Configuring Password:** Used to enter the 'Configuring the Phone' menu from the main Menu, and also to enter 'Web Jeeves'.  
**Default = 1234.**
  - **User Password:** Used to unlock keypad. **Default = 1111.**

### Configuring:

- Click on Web '**Change Password**'.

### How to use Web Page?

Parameters	Value/Default	Description
<b>Configuring Password</b>		
Enter New Password	4-digits maximum 0-9	Enter the new password. If you want to change the Password. Remember it.
Retype Confirm Password	-do-	Enter same password as above
<b>User Password</b>		
Enter New Password	4-digits maximum 0-9	Enter the new password. Remember it.
Retype to Confirm Password	-do-	Enter same password as above

### Relevant Topics:

1. [Menu](#) 100
2. [Keypad Lock](#) 88
3. [Web Jeeves](#) 168
4. [Default SETU VP236](#) 79

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## CLIR

### Introduction

- CLIR is “Calling Line Identity Restriction”. This feature is used by the user when he doesn’t want to disclose his identity to the called party.
- Assign the key for CLIR with single color LED.

### How to use it?

- Set/cancel the feature from Web page ‘User Setting’.
- Press the key assigned to set the CLIR. LED will glow.
- By using phone, you can navigate Menu to select CLIR and press ‘Enter’ key.

### How it works?

- The key assigned for CLIR set/cancel feature will work in toggle mode.
- When ‘set’, LED of the key will glow. So that user is notified that CLIR is set. When ‘cancel’, the LED of the key is turned OFF.
- If key for CLIR is not assigned, and when the CLIR option is selected from the Menu, the display will show the status of CLIR.
- The phone will retain the CLIR status while power down.

### Configuring:

- For key assignment, Click on ‘Keys Programming’.
- Refer the topic ‘[Menu](#)’ to know how to set/cancel CLIR from Phone (Menu).
- Click on ‘**User Setting**’ to set/cancel from Web. Refer chapter ‘[User Settings](#)’.

### Relevant Topics:

1. [Menu](#) 100
2. [Know Your SETU VP236](#) 17

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## Configuring the Phone-Menu

For more details, refer chapter "[Know Your SETU VP236](#)".

### How to use it?

- Select the parameter, 'Config Phone' from scrolling the 'Menu' Parameters.
- Press "Enter" key. Enter User Password.
- Navigate 'Config Phone Menu' by up/down key.

For entering "Config Phone Menu", follow the steps mentioned below:

- Press 'Menu' key, if assigned or select 'Menu' from LCD display and press 'Enter' key.
- Navigate the list with Up/Down key and select "Configuring the Phone" option.
- Press 'Enter' key. System will ask 'Password'.
- On entering correct password (default 1234), you can navigate the menu using up/down key.
- Select the feature by scrolling the list and press 'Enter' key.

### For example:

- User can select 'DND List' to edit the number in the List, displayed on the LCD Display.
- All the five editable keys are used for editing.
- Here, < key will not work as 'back key, rather it will shift the cursor on the LCD display to the left by one character.
- After editing when the user presses 'Enter' key, the number/ address is saved whereas the cursor comes to the initial position.
- The user may use Up/Down key to change the rows.
- To exit the mode, user should press < key (back key). Whatever changes have been made will be saved.

### Related Topics:

1. [Know Your SETU VP236](#) 17
2. [Menu](#) 100
3. [Do Not Disturb](#) 83

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## Daylight Saving Time

### Introduction

- It is the new local time assigned to a region for a portion of the year, usually an hour forward from its standard official time.
- The clocks are advanced at the beginning of the daylight saving time and delayed at the end of the daylight saving time.
- DST can be set according to day-month wise or date-month wise.
- DST can be disabled by the SE if not applicable in the country.
- DST is changed on a particular day and month every year (say Type1) for some countries. The DST is changed on exact date and month (say Type 2) every year for some other countries and remaining countries don't follow DST.
- If DST type 1 is selected, the DST is effective on the programmed day and the month whereas, if DST type 2 is selected, the DST is effective on the programmed date and the month of the year.

### Configuring:

- Click on Web page “**Daylight Saving Time Adjustments**”.  
Program the parameters as per the DST rules of the Country where the IP Phone is installed.

### **Forward Time Adjustments:**

**Type:** Select the type from

- None
- Day-Month wise: Program this option only for a country where DST is observed as per Day-Month wise every year.
- Date-Month wise: Program this option only for a country where DST is observed as per Date-Month wise every year.

**Default = None.**

### **Day-Month Wise:**

- **On:** Program 'day' and 'month' on which DST will be applied every year. For this, program Ordinal, Day and Month. For this option, 'Type' should be selected as 'Day-Month wise'. **Default = 1<sup>st</sup>. Sunday January.**

**Ordinal:** Program 1<sup>st</sup>, 2<sup>nd</sup>, 3<sup>rd</sup>, 4<sup>th</sup>, or 5<sup>th</sup>. For example if DST starts on

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first Sunday of January every year, then program '1<sup>st</sup>' here.

**Note:**

- For last Sunday (or other day) of the month, always set Ordinal Number = 5th. For e.g. If the month has 4 Sundays in a particular Calendar year then the last Sunday will be automatically the fourth one and if the month has 5 Sundays in a particular month of the Calendar year then the last Sunday will be automatically the fifth one, and since while programming it will not be known that in a given year, the last Sunday will be fourth or fifth one, user should program 5th for last day of month.

**Day:**

- Program the day on which DST starts. Select from Sunday to Monday of the week. In above example, select 'Sunday'.

**Month:**

- Program the month on which DST starts every year. Select the month from January to December of the year. In above example, select 'January'.

**Change Time From:**

- It is the current time settings at which DST will start to change. The time is in HH:MM (Hours-Minutes) format, where HH=00 to 23 and MM=00 to 59. **Default = 00:00.**

**To:**

- Enter forwarded time here at which DST will be forwarded. Enter the time in HH:MM (Hours-Minutes). **Default = 00:00.**

**Date-Month Wise:**

- **On:** Program 'date' and 'month' on which DST will be applied every year. For this, program Date and Month. For this option for 'Type' should be selected as 'Date-Month wise'. **Default = 01 January.**
- **Date:** Program the date on which DST starts. Select from '01' to '31' of the month. For example if first January is the DST date, then select '01'
- **Month:** Program the month on which DST starts every year. Select the month from January to December. In above example, select 'January'.
- **Change Time From:** It is the current time settings at which DST will start to change. The time is in HH:MM (Hours-Minutes) format.

**Default = 00:00.**

- **To:** Enter forward time here at which DST will be forwarded. Enter the time in HH:MM (Hours-Minutes). **Default = 00:00.**

**Backward Time Adjustments:**

- Program all the parameters as explained for 'Forward Time Adjustment' except the 'To' time. Default values are same.
- **To:** Enter the backward (delayed) time here at which DST will be backwarded (delayed).

**Note:**

- If for some country, the Backward Time Adjustments is at 00:00 hours, use previous date with "from" time = 23:59 and "to" time as required.

**Example 1:**

- In New Zealand, the DST starts on Last Sunday of October. The clock changes from 02:00 to 03:00. Hence, parameter 'On' will be as 5th Sunday, October as per Day-Month wise option, of 'Forward Time Adjustment'.

**Example 2:**

- In New Zealand, the DST ends on Third Sunday of March. The clock changes from 03:00 to 02:00. Hence, parameter 'On' to affect DST in New Zealand, will be programmed as 3rd Sunday, March as per Day-Month wise option of 'Backward Time Adjustment'.

**Example 3:**

- In Cuba, the DST starts on 1st April of every year. The clock changes from 01:00 to 02:00. Hence, parameter 'On' will be programmed as '01 April' as per Date-Month wise of 'Forward Time Adjustment'.

**Example 4:**

- In Cuba, the DST ends on last Sunday of October every year. The clock changes from 23:59 (in fact 00:00 midnight) to 23:00. Hence, parameter 'On' will be programmed as 5th Sunday, October as per option 'Day Month wise' of 'Backward Time Adjustment'.

**Relevant Topic:**

1. Time Settings 148

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## Default SETU VP236

### Introduction

- Some times while user is configuring and programming the Phone, he would like to start from beginning and wants to reset the Phone to its factory set values.
- This is done by the feature “Default the SETU VP236” Matrix phone can be defaulted only from the Web Page. But the passwords can be default using phone as explained below.

### Default Password using keypad:

- The SETU VP236 supports two passwords: Config Password and User Password.
- If you default the system, these passwords will also get default. But if you forget the config password, you can not enter ‘Jeeves’ to change the password. To avoid such problems, the SETU VP236 supports following procedure:
  - Go OFF-Hook. You will get ‘dialtone’. Navigate ‘Menu’ to select ‘Alphanumeric Dialing’.
  - Press characters **‘reset’** the string will be dialed out.
  - You will get dial tone after resetting of both passwords and a message will be displayed for resetting of password.

### How to use it?

- Click on web page ‘Default IP Phone’.
- Click on ‘Yes’ when the phone shows Alert message.
- Using phone, navigate Menu > Alphanumeric Dialing > Press charaters ‘reset’.
- You will get dial tone after reset.

### For Example:

- ‘This shall assign default values to all the parameters of SETU VP236, do you want to continue?’
- Click on ‘OK’ if you want the SETU VP236 to get settings as per set by factory.
- Click on ‘Cancel’ if you want to continue with current status of the Phone.

**Relevant Topic:**

1. Change Password 73

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## Dial Plan

### Introduction

- When user has two SIP accounts, he can use SIP1 for making calls to US and SIP2 for making calls to Gulf. He may like to have a facility to select the SIP Line by the system automatically for a specific Number to be dialed out. Matrix provides such facility by providing a feature called “Dial Plan”.
- This feature is programmable by Web Page “OG Call Route Selection”.

### How to use it?

- Program the number to be dialed in the dial plan table.
- Dial that number.
- Call will be made using ‘SIP Account’ as per the table.

### How it works?

- The system will make an OG call based on the Number Dialed.
- The Dial Plan table has maximum 10 entries.
- Each entry has two parameters viz. Address and SIP line.
- The Number to be entered is of maximum 40 characters.
- SIP line can be SIP1 or SIP2 account as only two lines are supported by the Phone.
- Now Whenever an OG call is made, the number to be dialed out on SIP is compared with the entries in this table. If a match is found the number is dialed using the respective SIP account, else the number is always dialed using OG call Route selection logic.

### Example 1:

If the user wishes to make all calls starting with 33 through SIP1 but calls made to numbers starting with 331 through SIP2, he should enter 33 in first index with SIP1 and 331 in the second index with SIP2.

### Example 2:

Suppose that the dial plan table is programmed as shown below:

## Matrix

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Index	Address	SIP Account
01	5678	SIP1
02	56	SIP2
:	:	:
10		

If user dials say 567894, the call is made using SIP1 as this number best fits with the number in index 01.

But if the user dials 561234, the call is made using SIP2 as this number best fits with the number in index 02.

### How to program?

Click on “**OG Call Route Selection**”.

Refer corresponding chapter for more details.

### Related Points:

1. [OG Call Route Selection](#) 110
2. [Selective SIP Line](#) 127

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## Do Not Disturb (DND)

### Introduction

- Many times, user needs privacy for some duration and do not want to receive any calls without logging off from the server or switch OFF the phone. But at the same time, he would like to receive calls from specific callers. For such requirements, Matrix SETU VP236 supports a feature “Do Not Disturb” (DND).
- User can set/cancel DND.
- User can set DND for all callers or for few callers and allow other callers to make calls on his address.
- User can use ‘Menu’, key or Web page for configuring this feature. The key assigned is with single color LED.

### How to use it?

- Set DND, by pressing the DND key assigned
- Navigate the menu using Up/Down keys and select the desirable option and press ‘Enter’ key.

### How it works?

- DND is applicable for IC calls only. OG calls can be made as usual.
- User can allow maximum 10 callers to call him even if he has set DND. The numbers of such callers can be programmed in a list called ‘DND List’.
- DND list can be programmed from Web Jeeves as well as from Phone (Menu → Configuring the Phone).
- DND can be set/canceled:
  - From Menu
  - Using a key
  - From Web
- When DND key is pressed user will get following three options:
  - DND All
  - DND List
  - DND Cancel

### DND All:

- If the user selects DND All, no incoming call will be accepted. The call will be rejected by the SETU VP236 with cause ‘Busy’. LED

assigned, will glow.

**DND List:**

- If the user selects DND List, the caller ID of the IC call is fully matched with numbers in the DND List and if the number matches the call will be accepted else it will be rejected with cause 'busy'. Refer chapter '[Configuring the Phone-Menu](#)' for programming the list.

**DND Cancel:**

- If user selects 'DND Cancel' then 'DND' feature is not applicable and the SETU VP236 will work normal as per other settings.

**Configuring:**

DND List can be configured through Web or Phone.

- DND through Web-Refer the topic '[Web Jeeves](#)' for more details. Click on the Web page 'User Setting'.
- DND through Phone-Navigate Menu > DND.

**Important Points:**

- Black Caller list is given priority over DND. The call gets rejected before checking for DND.
- Each rejected call due to DND will be logged as rejected call. Rejected calls log may also contain the calls rejected manually by the user or calls rejected due to Black Caller List.

**Relevant Topics:**

1. [Call Forward](#) 52
2. [Rejecting a Call](#) 122
3. [Configuring the Phone-Menu](#) 75

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## Hotline

### Introduction

- In order to avoid entering and dialing of long digits every time, user will prefer that these numbers are dialed out by the system, automatically.
- Matrix SETU VP236 provides this feature by allowing preprogrammed number to be dialed by the SETU VP236 after a short delay. This feature is called “Hot Line”. The delay is programmable, by parameter ‘Hotline Timer’.
- It can be programmed by Web page, Menu parameters and using a specific key.
- Assign the key with single color LED indication, only.

### How to use it?

- |  |
|--|
| <ul style="list-style-type: none"><li>• Set Hotline by pressing a key or using Menu options.</li><li>• Go OFF-Hook for the time of “Hot Line Timer”.</li><li>• The programmed number will be dialed out.</li></ul> |
|--|

### How it works?

- The status, Hotline address and timer will be retained when Phone is powered OFF.
- If the user dials a number before expiry of the Hotline Timer, the number dialed by the user is dialed.
- If Hotline is set but the hotline address is not programmed then user will get error tone.

### How to program?

- Hotline can be set/cancel:
  - From Web
  - From Menu
  - Using a key (refer, '[Key Assignment](#)')
- Click on Web page “System Parameters” for configuring ‘Hotline Timer’. Refer chapter '[System Parameters](#)'.
- Click on Web page, “User Setting” for programming ‘Hotline Address’. Refer chapter '[User Settings](#)'. Configure Hotline feature as shown below:

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Parameter	Value/Default	Description
Hotline	Set or Cancel	Select 'Set' if you want to use the feature.
Hotline Address	Default = Blank or ASCII characters max. 40	Enter here the Address you want to be dialed by the Phone when using the Hotline feature.

**Important Points:**

- It is advisable for the user to set hotline timer less than First Digit Wait Timer.
- If hotline timer set is more than First Digit Wait Timer, user will get error tone after the expiry of First Digit Wait Timer and the programmed hotline address will not be dialed.

**Relevant Topics:**

1. Menu 100
2. Key Assignment 87
3. System Parameters 144
4. User Settings 152

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## Key Assignment

### Introduction

- This chapter explains, how you can assign a specific key to a specific feature.
- After assigning the key press that key to use the feature.

### How to use Web Page for key assignment?

- Click on the Web Page “Keys Programming” for assigning the specific number of key.

### For Example:

Suppose you want to assign a key number 01 to a feature.

- Click on box for key number 1. The list appears. Select a feature for which key number ‘01’ is to be assigned.
- User should assign the feature a single or dual color LED as per requirement only.

**BE CAREFUL TO SELECT:**

A KEY WITH SINGLE COLOR LED or A KEY WITH DUAL COLOR LED or A KEY WITH NO LED, FOR THE FEATURE WHICH IS TO BE ASSIGNED. REFER RELATED CHAPTER FOR THE REQUIRED KEY.

- Assign required feature to key number from **01-18**. You can operate the feature by pressing the ‘Key number’ assigned here. Refer chapter ‘[Know Your SETU VP236](#)’ for list of feature.
- If you have selected “Phone Book” from the feature list, then select the index number at which required contact phone number is to be assigned as per the phone book table.

### Relevant Topics:

1. [Know Your SETU VP236](#) 17
2. [Phone Book](#) 117

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## Keypad Lock

### Introduction

When the keypad is locked it is not possible to make OG calls.

Using this feature it is possible:

- To lock the keypad by pressing a key assigned or using the option 'Lock Keypad' from Menu.
- To lock the keypad automatically after a programmed time.
- To receive IC calls in locked condition by using CA keys, Headset key, speaker key and Hook switch.
- To unlock the keypad the password is required.
- Assign the key with single color LED.

### How to use it?

- **To lock the keypad:**
  - Press the Keypad Lock key which is pre-assigned.
  - Select the option 'Lock Keypad' from the Menu.
- **To unlock the keypad:**
  - Press the keypad lock key again or press any other key, which is assigned
  - Enter the password (default 1111).
- If Auto keypad lock is enabled, the keypad will get locked after time = Keypad Lock Timer.

### How it works?

- This feature is applicable during any state of the Phone like receiving, calling, etc.
- Auto keypad lock can be selected from Web as well as from Phone (Config Phone Menu).
- This feature works with three parameters:
  - Keypad Lock Methods: It has two modes viz. Manual and Auto.
  - Keypad Lock Timer : It can be set from Web as well as from Phone (Config Phone Menu).
  - Keypad lock/unlock (manual) : It can done using a key/Menu.
    - Keypad lock (manual) : It can done using an option 'Keypad Lock' from user Menu.
    - Keypad unlock (manual) : It can done by pressing any key and entering password.

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**Keypad Lock : Manual**

- You can assign a key for Keypad lock feature.
- On pressing the 'Keypad Lock' key when the keypad is open:
  - The keypad is locked.
  - LED of the key is ON.
  - You will get display of a text message, 'Keypad Locked' for 2 Sec.
  - Phone will go to idle state/standby state or previous state.
- Pressing the 'Keypad Lock' key or any other key when the keypad is locked:
  - System will ask 'User Password'.
    - If you enter correct user password: (Default user p/w is 1111).
      - You will get display 'Keypad Unlocked' for about 2 seconds.
      - LED is turned OFF.
      - Phone goes to idle state/previous state.
    - If you enter incorrect user password:
      - You will get the message on LCD: 'Invalid Password, try again' for about 2 seconds.
      - Go to idle/standby state or previous state.
    - If No password is entered:
      - You will get the message on LCD: 'User Password not entered'.
      - Phone will go to idle/standby state or previous state.
- Keypad lock status is stored while power down.

**Keypad Lock : Auto**

- When you make any activity/event, keypad lock timer is started.
- If no activity/event is done till expiry of keypad lock timer:
  - Keypad is Locked.
  - LED of the key is turned OFF.
  - You will get text message as "Keypad Locked" for about 2 seconds.
  - Phone goes to Idle/Previous state.
- Keypad status is retained even during power down.

**Configuring:**

- Auto Keypad Lock and 'Keypad Lock Timer' are configurable from Web Jeeves and Phone (Config Phone Menu).

- Click on Web Page, “**System Parameters**”. Refer chapter ‘System Parameters’ to set/cancel ‘Auto Keypad Lock’ and ‘Keypad Lock Timer’.

**Important Points:**

***In Keypad Lock-Manual***

- If the user forgets the password, he can change the user password from Web Jeeves and open the keypad. Alternatively, the user may default the SETU VP236.
- If key for Keypad lock is not assigned, the keypad can be locked from the Menu.

***In Keypad Lock-Auto***

- If the call is being made and the keypad lock time gets expired then the keypad will get locked. However, the call will not get disconnected.

**Relevant Topic:**

1. System Parameters 144

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## Last Number Redial

### Introduction

- It is desirable that user should just press one key if he wants to dial same number frequently. For such case a feature is required by which a user is able to redial the last number by pressing a key or selecting option from the Menu. This is called 'Last Number Redial'.
- The key to be assigned will be without LED.

### How to use it?

- Press the Last Number Redial key, which is preassigned.
- Last dialed number will get dialed out.

### How it works?

- By pressing the assigned key for Redial, user can redial the last number on which OG call was made or number dialed.
- To dial the last dialed number again, the last number can be obtained from the last dialed numbers' Call Log.
- If key for Redial is not assigned, the feature can be used from the Menu.
- Calls with redialed numbers are stored in the Call Logs.

### Configuring:

- Refer Web Jeeves and click on 'Keys Programming' for assigning key number.
- To redial, using phone, select option 'Last Number Redial' by navigating 'Menu'.

### Relevant Topics:

1. [Know Your SETU VP236](#) 17
2. [Key Assignment](#) 87

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## LCD Settings

### Introduction

- Matrix SETU VP236 supports the adjustment of the LCD contrast and brightness to improve readability in various conditions. The parameters like Backlight, brightness and Contrast can be adjusted.
- The Feature is configurable by Web page and “Config Phone” Menu.

### How it works?

#### **Backlight:**

- User can turn ON or OFF the Backlight of the LCD display on the Phone.
- The Backlight will be turned ‘ON’ in any of the following conditions:
  - User receives an incoming call.
  - User goes OFF-Hook
  - User dials any digit in any state
  - User pressed any key in any state
  - User goes ON-Hook
  - When Remote end disconnects/ rejects/ held the call
  - User receives any notification (ACB, Message Waiting Notification, Transfer)
- User can also program the ‘time’ after which the backlight will be turned OFF if no activity is performed on the Phone. For example, if 10 sec. is programmed, the Backlight will be turned OFF after 10 sec if no activity is done during this time.
- Thus following options can be programmed:
  - Always ON
  - Always OFF
  - For 10sec
  - For 30sec
  - For 60sec
  - For 120sec
  - For 240sec
  - **By default it is ‘For 10sec.’**

#### **Contrast:**

- LCD Contrast Level 1 to Level 4 can be set. Level1 will provide

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lowest and Level 4 will provide highest contrast.

**Brightness:**

- LCD Brightness Level 1 to Level 4 can be set Level1 will provide lowest and Level 4 will provide highest brightness.

**Configuration:**

- Click on Web page “System Parameters”. Refer chapter ‘[System Parameters](#)’.
- Refer topic ‘Config Phone Menu’ for more details, refer ‘[Know Your SETU VP236](#)’.

**Related Topic:**

1. [System Parameters](#) 144

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## Making a Call

### Introduction

- The Matrix IP telephone provides advanced calling functionality. For example, it can support multiple calls on a single line. It can also support multiple lines.
- Maximum number of simultaneous outgoing calls is equal to the number of CAs programmed. Means if number of CAs programmed are four, maximum four calls can be made simultaneously.
- For Example, if number of CAs programmed are 3, first OG call will be made from CA1 (if does not select explicitly other CA).
- If user selects CA2, first call on CA1 will go 'held' and user will get dial tone on CA2 to make a new call.
- Similarly, user can make further calls (maximum four calls when CAs programmed are 4) by holding the current call.

User can make a call by following methods:

- Using Handset
- Using Speaker
- Using Headset
- Hot Pad Dialing

These are explained below:

- Using Handset:
  - Lift the handset, wait for the dial tone and then dial the desired number.
- Using Speaker:
  - Press the speaker key, Wait for the dial tone and then dial the desired number.
- Using Headset:
  - Press the "Headset" key, wait for the dial tone and then dial the desired number.
- Hot Pad Dialing:
  - This is a type of ON-Hook Dialing.
  - Press the number directly being ON-Hook
  - The speaker gets turned ON
  - Dial the digits
  - You will get call progress tones.

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- You can use any method of dialing but after making a call, you can switch between handset, headset and speaker mode.
  - User can also make a call by using following features:
    - Using the Phone Book
    - Using the Speed Dial
    - Using Call Logs
    - Using a selective SIP trunk
  - Refer corresponding chapter to use these features.

### How it works?

The system supports the feature “End of Dialing” so that system starts dialing out the digits as per the way in which the digits are entered.

This is explained first followed by the methods for dialing the number:

- **End of Dialing:** SIP Number Dialing and IP Number Dialing.
- **ON-Hook Dialing:** Enter the number first and then dial the number, also called Pre-dialing.
- **OFF-Hook Dialing:** Pick up the handset and then dial the number.

### End of Dialing:

The Phone supports the feature by following options:

- You can press ‘Enter’ key, after dialing digits string.
- You can Program the Inter Digit Wait Timer after expiry of which the system dials out the number.
- You can program the “Length’ of the “Fixed Number of Digits’. **By default, it is disabled.**
- Program these in the Web page for “**System Parameters**”. You can enable any of above option. If all are enabled then system will dial out as per the first action detected. For example, ‘Enter’ flag is enabled, number length is set to 5 and Inter Digit Timer is set to 6 seconds, then end of dialing is considered using ‘Enter’, if “” is received after 3 digits. Likewise, end of dialing is interpreted if the Inter Digit Wait Timer expires after few digits.
- **By default, ‘Enter’ key is enabled, Inter Digit Wait Timer = 6 secs. and Fixed Number of digits Length = 40 characters.** (For more details refer chapter ‘Timers’).

### Caution!

***User should never program all options of “End of Dialing” as “Disable”. Other wise system cannot dial any number.***

### **How SIP Number (SIP ID) can be dialed?**

- Press a key which is assigned as 'Alphanumeric Dialing Key'.
- This will indicate that SIP address is to be dialed.
- Dial the SIP ID. You can dial SIP ID which considers numbers first and then alphabets. For example, if you dial 2, '2' will be interpreted first followed by a, b, c.

### **For Example:**

You can dial a SIP ID number like [ajay246@MatrixTeleSol.com](mailto:ajay246@MatrixTeleSol.com) by first pressing key which is assigned as 'Alphanumeric Dialing Key' and then dialing the ID with all alphabets and numerical.

### **How IP number can be dialed?**

- You can dial IP address by replacing 'dot' with 'asterisk'
- You can dial using the same way as used for dialing a SIP ID. Both the options can be used.
- If you dial the number using first option, the system will consider the number to be an IP number when it receives 'Asterisk' as the fourth character of the number string.

### **ON-Hook Dialing**

- Phone is in Idle state.
- Dial the digits till End of Dialing.
- Number is dialed out as per Peer-to-Peer Table if best match is there with number string in the Peer-to-Peer Table.
- If match not found, it will be dialed out using SIP account as per OG call route selection logic.
- The speech is established through speaker. One Call Appearance (CA) is occupied and the LED glows for the respective CA key. The number dialed is displayed on the LCD in the CA selected. The trunk through which the call is to be made depends on the OG route selection method. If the call is made through SIP1 trunk, SIP1 is displayed in the call status characters. Likewise, 'SIP2' is displayed, if the call is made through SIP2 trunk.
- If you dial end of dialing digit as the first digit, Error message is displayed on the LCD.

### **OFF-Hook Dialing**

- You can go OFF-Hook, pressing speaker key, headset key or lifting

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- handset.
- Dial tone and state of First Digit Wait Timer.
  - On expiry of 'Timer', you will get Error Tone for Error Tone Timer and then phone will go in 'Standby' and 'Silent' state.
  - You will get 'error message' on the LCD.
  - Go ON-Hook. Phone will go in 'Idle' CA key is released, LED of speaker key will be OFF.
  - You will get 'Idle' message on the LCD.
  - But if you dial any digit, during FDWT, and if you don't dial any digit till IDWT then dialed number string is dialed out call will be processed further.

### **Examples for dialing different types of IP address/numbers:**

- **12345/ vivek/vivek12345/\*12345**
  - When you dial the SIP Address which contains only numerals/ alphabets/ combination of numerals & alphabets/ special characters as prefix ("\*" etc), then system will apply the OG call routing logic and number will be sent to the SIP server found from the OG routing logic.
- **12345@192.168.1.37**
  - If you dial the SIP Address which contains the IP address, the Phone will compare the IP address dialed with the registrar server address (any one among the two, as IP phone support two SIP lines) programmed in SIP parameters configuration.
    - If the IP address dialed match perfectly with the registrar server address programmed, then it will send the dialed SIP address, to the same SIP server to which match found, using the same SIP line.
    - If the IP address dialed doesn't match with the registrar server address programmed, then it will consider the call as peer to peer call and the logic for peer to peer calls is used for dialing.
- **12345@xyz.com**
  - If you dial the SIP Address which contains the domain name, the Phone will compare the domain dialed by the user with the registrar server address (anyone among the two as IP phone support two sip lines) programmed in SIP parameters configuration.
    - If the domain dialed match perfectly with the registrar server

address programmed, then it will send the dialed SIP address to the same SIP server to which the match found, using the same SIP line.

- If the domain dialed doesn't match with the registrar server address programmed, then it will consider the call as peer to peer call and the logic for peer to peer calls is used for dialing.

### **Important Points:**

- If the user has two calls held by remote users, it will not be possible to make new call since no resources are available to entertain the new call. Refer chapter 'Call Hold' for 'Remote Held'.
- However, user will get dial tone when selects the free CA to make a new call. Since, no resources are available, call will not be completed. But to make a call, user will have to release one of the calls held by remote user.
- User can make a new call when user has Active Conference. Maximum two simultaneous OG calls can be made when user has active conference.
- When user has active conference which will occupy two CAs and user selects the free CA to make new call, two calls involved in conference will go 'held' and the user will get the dial tone.

### **Relevant Topics:**

1. Switching between Handset, Headset and Speaker 142
2. Peer-to-Peer Calling 113
3. Call Hold 56

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## Making Second Call

### Introduction

- This feature allows you to make another call while holding current call. It also allows disconnecting the second call and retrieving back the first call.
- This feature is useful for using the phone with multiple calls, as explained below:
  - User can hold one call (IC or OG call) and make another call.
  - User can toggle between the two calls. Refer topic '[Call Toggle](#)' for more details.
  - User can make 3-Party conference after establishing second call.
  - User can transfer the held call to the second call.

### How to use it?

You are busy with one active call, first call.

- You are in speaker mode. Press 'Hold' key.
- Dial the number string to make second call.
- Now CA2 is occupied.

The same logic can be extended for subsequent calls. For example, maximum four simultaneous calls can be made if the number of Call Appearances are programmed as 4.

### Note:

- You can also select 'Hold' from the features by navigating the 'Menu', to make a second call.

### Related Points:

1. [Call Toggle](#) 68
2. [Call Appearances](#) 49
3. [Call Waiting](#) 72
4. [Call Transfer](#) 69
5. [Call Hold](#) 56
6. [3-Party Conference](#) 37

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## Menu

### Introduction

- Some features can be accessed from the Phone just by navigating the 'Menu'. It is a kind of 'short cut' method instead of entering Web Jeeves, to set the feature.
- 'Menu' can be accessed by following ways:
  - Using Normal Display on the LCD: Select the line displaying 'Menu' using up/down key and press 'Enter' key.
  - Using the specific key assigned for 'Menu'. The key assignment can be done only from the Web Page, 'Keys Programming'.

### How it works?

- Menu key can be pressed and features can be navigated by up/down key provided on the phone.
- For example, to configure parameters of SETU VP236 from the phone itself, user will have to navigate the options of the 'Menu' and select the option "Configuring the Phone". Note that SETU VP236 will ask 'Password', if you want to enter the 'Config the Phone'. Similarly user can select "Hotline" to set hot line feature.

### ***Following cases explain the use of Menu Mode:***

#### **If no activity during Menu mode:**

- If the user does not perform any activity, he will go to Normal Display after about 15 seconds or by pressing the ← key (Back key).

#### **OG/IC Call when not connected:**

- You are in Menu state, then to make an OG call or receive an IC call, you will have to come out of the Menu.
- Also In case of receipt of IC call, the phone will ring and the CA on which the call has arrived will display normal IC call activities.

#### **OG/IC Call when connected:**

- When you are in connected state (during speech) and receive IC call the call will be displayed on the LED of the CA, and you will get indication by beeps for call waiting.
- In such case you can press the CA on which the IC call has

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arrived, to get connected to the arrived IC call.

**Network Status:**

- Following network parameter will be displayed on IP Phone if user selects 'Network Status' from the menu:
  - IP Address
  - Subnet Mask
  - Gateway Address
  - DNS Address
  - Domain Name

**OFF-Hook during Search contact option:**

- If you are in 'Search Contact' option of the Menu select the number and go OFF-Hook, to make an OG call.

**Accessing SIP Line:**

- During 'Menu' mode if you use a feature like 'Accessing SIP2' while talking to remote user1 then the current active call gets held and you will get dial tone to dial out another number of remote user2.

Which features can be accessed?

Refer chapter '[Know Your SETU VP236](#)'.

**Important Points:**

- 'Call Forward' can be set by entering Destination Number from keypad.
- When selecting 'Config Phone' Menu during speech, using keypad digits will not affect current call or speech.

**Related Topics:**

1. [Getting Started](#) 27
2. [Know Your SETU VP236](#) 17

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## Network Port

### Introduction

- For considering variations of network conditions and usage by various customers, Network Port for SETU VP236 are programmable.
- For this Matrix SETU VP236 provides a feature called “Network Port Parameters”.
- It can be programmed from Web page, but some parameters can be programmed from the phone also.
- You can also get display of few network parameters by accessing ‘Network Status’ in ‘Menu’. Refer chapter ‘[Menu](#)’.

### How it works?

Matrix SETU VP236 supports the network parameters:

- Configurable by Web
- Configurable by Phone

### Configurable by Web

- Connection Type
- PPPoE User ID
- PPPoE Password
- Static IP Address
- Subnet Masks
- Gateway Address
- DNS Server
- DNS IP Address
- DNS Domain Name
- STUN Server Address
- STUN Server Port
- NAT Keep Alive Status-Enable/Disable
- NAT Keep Alive Interval
- SIP QoS Type
- SIP QoS Level
- RTP QoS Type
- RTP QoS Level
- Use VLAN
- VLAN Id

- 
- SIP Listening Port
  - RTP Listening Port

Refer the chapters “[Web Jeeves](#)” and “[VoIP Basics](#)” for detailed explanation of IP Address, PPPoE, DHCP, STUN, NAT, etc.

### **Configurable by Phone, after selecting ‘Network Parameters’ from Config Phone Menu.**

- Connection Type-DHCP, PPPoE and Static
- IP Address
- Subnet Mask
- Gateway Address
- DNS Server-Automatic/Static
- DNS Server Address
- DNS Domain Name
- PPPoE User ID
- PPPoE User Password
- VLAN Setting

### **How to use?**

- For programming by the phone, navigate Menu > Config Phone > Network Parameter.
- Click on Web page “**Network Parameters**”. Refer [Web Jeeves](#) for details of programming the parameters.

### **Connection Type:**

Select IP Address allocation from three options:

- DHCP : If clicked on DHCP, IP address, Subnet Mask, Gateway Address will be replaced by the dynamic values sent by the DHCP server.
- PPPoE : If clicked on PPPoE, IP Address will be allocated by PPPoE Server.
- Static IP : If clicked on the option ‘Static IP’, you can use IP Address by your ITSP. Default = Static IP.

***By default, Connection type is ‘Static IP’.***

Enter PPPoE parameters as shown:

- **PPPoE User ID:**

It is a string of maximum 16 characters. It can be extended ASCII

character set.

**By default, PPPoE User ID is Blank.**

- **PPPoE Password:**

Enter a string of maximum 16 characters. It can be extended ASCII character set.

**By default, PPPoE Password is Blank.**

Enter Static IP parameters as shown:

- **IP Address:**

It is relevant only if Connection Type = Static IP.

It is a string of maximum 15 characters.

(Enter the dot '.' using the digit '1' as used in case of alphanumeric dialing).

**Default IP Address is 192.168.001.161.**

- **Subnet Mask:**

It is a string of maximum 15 characters.

**By default, Subnet Mask is 255.255.255.0.**

- **Gateway Address:**

Gateway Address is a string of maximum 15 characters.

**By default, Gateway Address is Blank.**

Make DNS (Domain Name Server) settings as:

- **DNS Server:**

- Obtain Automatically

- Static

If 'Obtain Automatically' is selected and connection type is DHCP,

then DNS address provided by the DHCP server, will be used. If

'Obtain automatically' is selected and connection type is PPPoE,

then DNS address provided by PPPoE server will be used. If static

is enabled then phone will use the DNS address programmed

manually. You cannot select 'Obtain Automatically' when connection

type is set 'Static'. **By default, DNS Server is static.**

- **DNS Address:**

DNS IP Address is a string of maximum 15 characters using 0-9

digits. **By default, DNS Address is Blank.**

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- **DNS Domain Name:**

It is a string of maximum 24 characters. It can be extended ASCII character set.

***By default, DNS Domain Name is Blank.***

Enter NAT Parameters as:

- **NAT Keep Alive Status:**

Select from Enable/Disable.

If enabled, the SIP messages will use NAT router's public IP address and SIP port number. The NAT will also be configured with this port number to forward the traffic with this port number. NAT eliminates the need for STUN.

***By default, NAT Keep Alive Status is Disable.***

- **NAT Keep Alive Interval:**

This timer specifies the time after which the SETU VP236 will send specific SIP messages to the SIP server. Digits can be used from 001-200 Seconds.

***By default, NAT Keep Alive Interval is 120 Seconds.***

Enter STUN parameters as:

If NAT is configured do not program STUN:

- **STUN Server Address:**

STUN server address is the address of the STUN server. It can be IP address also. Maximum 40 characters can be used. It can also be extended ASCII characters.

***By default, STUN Server Address is blank.***

- **STUN Server Port:**

STUN Server Port specifies the STUN Server's listening port for SIP. User may change this field if the ITSP provides STUN server port number other than the default.

Range is from 1024 to 65535.

***By default, STUN Server Port is 3478.***

Enter QoS (Layer3) parameters as:

- **SIP QoS Type:**

This field specifies the Quality of Serve (QOS) type. The SETU VP236 sends all the SIP signaling messages with this setting.

Select from two options:

- Precedence (also called TOS/Priority)
- DiffServ is used for voice traffic.

**By default, DiffServ is selected.**

- **Enter the SIP QoS Level** in the field provided.  
'Precedence' uses 3 bits and hence has valid range from 0 to 7.

**By default, Precedence is 5.**

DiffServ uses 6-bits and hence has valid range from 00 to 63.

**By default, DiffServ is 26.**

- **RTP QoS Type:**

This field specifies the QoS type.

Precedence (Also called TOS/Priority).

DiffServ is used for voice traffic. The SETU VP236 sends all the RTP media with this setting.

**By default, DiffServ is selected.**

- **Enter the RTP QoS Level** in the field provided.  
Precedence uses 3 bits and hence has valid range from 0 to 7.

**By default, Precedence is 5.**

DiffServ uses 6-bits and hence has valid range from 00 to 63.

**By default, DiffServ is 46.**

### **VLAN Setting:**

#### **Use VLAN?**

- Select the option 'Yes' from Yes/No if 'VLAN Tag' is required.
- If selected 'Yes', all the packets leaving the network port will include the VLAN Id programmed otherwise the packets will be sent untagged. **Default, use VLAN option is No.**

#### **VLAN Id**

- Program the Id also called 'VLAN identifier', if in above parameter 'Use VLAN?', 'Yes' is selected.
- Id can be programmed using Max. 4-digits from digits 0-9.
- Range = 0000-4095
- **By default, VLAN Id is '0'.**

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### SIP Listening Port

- During SIP signaling, source and destination ports are specified for the listening and transmission of SIP messages. This is specified by 'SIP Listening Port'. User can configure the SIP listening port as per requirement.
- System will use common SIP listening port for all the IP calls.  
Range = 1024 - 65535  
**By Default = SIP Listening Port is 5060.**

### RTP Listening Port

- RTP listening port is required to listen for RTP packets transmitted by remote peer after RTP session being established. User can configure the RTP listening port as per requirement.
- System will use unique RTP listening port for each RTP session (for call on each CA).
- User can program only **even** port number (as per RFC standard) for RTP listening port  
Range = 1024-65526  
**By Default = RTP Listening Port is 8000.**

### Relevant Topics:

1. [Configuring SETU VP236](#) 16
2. [Web Jeeves](#) 168
3. [Know Your SETU VP236](#) 17
4. [VoIP Basics](#) 162
5. [Peer-to-Peer Calling](#) 113
6. [VLAN Tagging](#) 154

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## Notifications

### Introduction

- Matrix SETU VP236 has the facility to provide the indication to the user that the Phone has responded the operation carried out by him. This is done by providing suitable 'Notification', which will be displayed on LCD for time of 'Notification Display Timer'.
- Following features are supported for providing notifications:
  - Call Transfer
  - Voice Mail
  - Call Forward
  - ACB Notifications
  - Remote Hold

### For Example:

- **Call Transfer Notification Timer:** This timer signifies the time for which the SETU VP236 should wait for the notification of transfer function from the transferee. This timer is programmable.
- **ACB Notification Timer:** This timer signifies the timer for which the SETU VP236 should wait for ACB 'ready for call completion' notification.
- **ACB SET:** If user has successfully set the Auto Call Back feature, he will get this notification.
- For other features refer related chapters and web pages.

### Notification Display Timer:

- This timer is the time for which the IP phone will display the notification '**Conc**' received from the transferee.
- Notification display timer is fixed of 5 seconds. The Call appearance will be released after expiry of display timer.
- Notification display timer is applicable for both blind and attended transfer.
- It is not used for other transfer notifications like 'Ring', 'Busy', 'Error' etc.

### For Example:

- If transfer notification timer programmed is 30 seconds and transferor receives the 'Conc' notification after 9 seconds after the

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transfer activity being performed, then the notification is displayed for 5 seconds on the LCD and CA is released after 5 seconds. Similarly for above case, if transferor receives notification 'Conc' after 27 seconds after transfer activity being performed, message will be displayed on the LCD for 3 seconds only.

**Important Point:**

- Transfer Notification Timer will always have priority over notification display timer.

**Relevant Topics:**

1. Call Transfer 69
2. Auto Call Back 42

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## OG Call Route Selection

### Introduction

- Matrix VP236 supports two SIP lines and different line can be selected by phone while making an OG Call. This is called “OG Call Route Selection”.
- This feature allows you to make a call with either of the SIP line registered with ITSP.
- One important application of this feature is like a case in Least Cost Routing (LCR). If you want to automatically select a low cost SIP Trunk ‘SIP2’ whenever you make an OG call, select the SIP2 trunk on the Web page.
- SIP1 or SIP2 can be selected alternately if ‘Rotation’ option is selected.
- This feature is programmable only through Web Jeeves.

### How it works?

- “OG Call Route Selection” supports the following options:
  - SIP1
  - SIP2
  - Rotation
- All the calls are made using SIP1 if SIP1 is programmed.
- All the calls are made using SIP2 if SIP2 is programmed.
- If ‘Rotation’ is selected, call can be made using SIP1 and SIP2 alternately.
- The Phone can select the SIP line automatically on the basis of number dialed (Refer chapter “[Dial Plan](#)”).

### Note:

- For ‘Rotation’ if last time calling had used SIP2-Line, then for current dialing, phone will select the SIP1-Line.

### How to use Web page?

Click on “OG Call Route Selection” from the menu.

Parameter	Value/Default	Description
Call Route Selection	SIP1, SIP2, Rotation. Default = SIP 1	Select the option for the ITSP Registered SIP Account line on which you want to initiate a call.
<b>Dial Plan</b>		
Number	Max. 16 digit, 1-9, 0, #, * Default = Blank	Enter the Number which should be dialed using specific SIP-Line. Maximum 10-numbers can be programmed with SIP Line selected
SIP Line	SIP1, SIP2	Select the Registered SIP Trunk line which will be selected by the Phone while making OG Call on the Number specified

**Relevant Topics:**

1. [Dial Plan](#) 81
2. [SIP Parameters](#) 128
3. [Selective SIP Line](#) 127

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## PC Port

### Introduction

- At this port you can connect the PC for configuring the VP236.
- This is one of the ports of the “Two Port Switch” which is inbuilt to the VP236. The other port is Network Port, for connecting the cable Modem or ADSL Modem. Hence, all the parameters of PC Port are same as Network Port.
- Refer the chapter [Network Port](#) for detailed information.

### Relevant Topic:

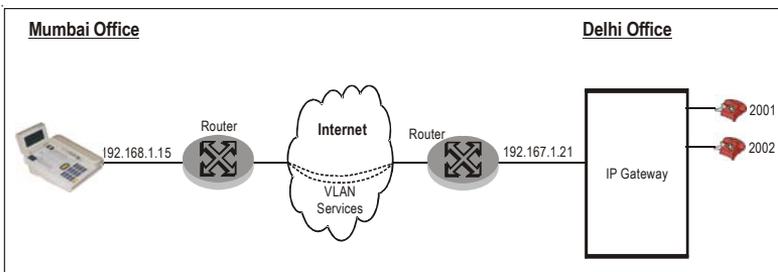
1. [Network Port](#) 102

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## Peer-to-Peer Calling

### Introduction

- Calling over IP network without using SIP server is called Peer-to-Peer Calling (P2P Calling).
- You can call another SETU VP236 connected by the VLAN Services provided by the ITSP, by the destination number which can be a number with IP Address or Domain Name.
- This feature works by programming:
  - Peer-to-Peer Config Table
  - Peer-to-Peer Table



### Peer-to-Peer Table:

- Suppose SETU VP236S are connected in Delhi office and Mumbai office and these two sites in different cities are connected to each other by the VLAN services.
- Then a call can be made from Delhi office to Mumbai office and vice versa, without using SIP proxy server.
- Now there are two ways to reach each other: One way is to dial the IP address. But many times, the offices use Gateways at both the ends. In such case, another way is to reach a particular station by dialing his station number followed by the IP address or Domain Name, which is programmed as destination address.

### Peer-to-Peer Configuration Table:

- User can use different codec's or DTMF schemes to take care of different Network conditions while making peer to peer call. Hence, different configuration of codec's and 'dtmf' option may be required for peer-to-peer calls and proxy calls. These can be programmed in

peer-to-peer configuration table.

- Program 'Yes' for the parameter, 'Use STUN'. Use 'Network Parameters' web page for programming STUN Server Address and Port Number.

### How to use it?

- Before entering Numbers and Destination Address for the peer-to-peer call at different entries, configure the peer-to-peer configuration table for these parameters;
  - Local User's Name
  - Vocoder Preferences
  - DTMF option
  - 'Use STUN' option
- Prepare a table of your contact Numbers with their Destination Address. Then program the information in the Web Page for 'Peer-to-Peer Table'.

### How it works?

- "Peer-to-Peer Table" supports 500 entries.
- Each entry consists of the Number and corresponding Destination address.
- The best fit logic is used to match the dialed string with programmed destination address in "Peer-to-Peer Table".
- If best match found, the call is considered as Peer-to-Peer call and <dialed number>@Destination Address will be dialed.
- If best match is not found, the call is considered as Proxy call. Suppose 'Peer-to-Peer Table' is programmed as follows:

Index	Number	Dest. Address
001	12345	abc.com
002	123	192.168.1.37
:	:	:
500		

- Now, if user dials say 123, the number dialed will be 123@192.168.1.37, as it best match with the entry in Index 02.
- But if the user dials 1234567, the number dialed will be 1234567@abc.com as this entry best fits with the number in index 01.

- For making an OG Call system will use the configuration for sending Name, selecting Vocoder, DTMF option and STUN which are programmed in 'Peer-to-Peer Configuration Table'.
- User can use global (Unique) IP Address in case of WAN and private IP Address in case of LAN to make P2P call.

### How to use Web Page?

- Click on Web page, "**Peer-to-Peer Configuration**". Refer Web Jeeves for programming details.
- Click on Web '**Peer-to-Peer Table**'.
- Program both as explained below:

#### Peer-to-Peer Configuration Table:

Parameter	Value/Default	Description
Display Name	Max. 24-characers with ASCII Default = Blank	Enter here Local user's Name which you want to be displayed at remote end.
DTMF option	Default = SIP (INFO)	Refer chapter SIP Parameters. Phone will use the option programmed here.
Vocoder Preference	10-preferences. Refer chapter SIP Parameters for default.	Select As per SIP Parameters. Refer chapter SIP Parameters. Phone will use the preference programmed here.
Use STUN	Yes/No. Default = No.	It specifies whether STUN should be used for P2P call or not. If 'Yes', STUN will be used to get the IP Address. If No, the STUN need not be used.

#### Peer-to-Peer Table:

## Matrix

Parameter	Value/Default	Description
Index	001 to 500	Number at which entry is made. 001 to 100 can be programmed. Click on links 101-200 or 201 to 300 or 301 to 400 and 401 to 500 to program other entries.
Number	Maximum 8 digits, 0-9, *. Default =Blank	Enter the extension number of the user agent to be called
Destination Address	Max. 40 chars. Including ASCII characters. Default=Blank.	Enter IP Address or 'domain name' of the user agent who will be called by P2P Calling.

### Important Point:

#### Programming of Authentication ID/Password for SIP Account '1':

- The Phone will use the Authentication ID and Authentication Password of SIP Account1 when Peer-to-Peer call is made and called party IP Phone (remote client) asks for Authentication. Hence the call will be terminated if remote client challenged for authentication but SIP Account1 is disabled or Authentication ID/ Password is not programmed.

### Relevant Topics:

1. [SIP Parameters](#) 128
2. [Making a Call](#) 94
3. [Network Port](#) 102
4. [Configuring the Phone-Menu](#) 75
5. [VLAN Tagging](#) 154

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## Phone Book

### Introduction

- Storing of the Name and Number in the memory of the phone, is very much useful to the user for using the SETU VP236. This is called “Phone Book” feature. You can store maximum 100 contacts.
- You can program the number/name of a contact in the Phone Book either from Web or through Phone (Using Configuring the Phone Menu).

### How to use it?

- Press the Search Contact key or select the Search Contact option from the ‘Menu’.
- All the contacts, programmed in Phone Book is displayed in alphabetical order.
- Navigate the Phone Book using up/down navigation keys.
- Press enter key on the name selected, to see the details (i.e. number) of the contact.
- Press ‘←’ navigation key to go back one step each time it is pressed.
- Finally to dial the number, go OFF-Hook or press speaker key or press headset key or press ‘Enter’ key.

### How it works?

- In the Phone Book, 100 entries (contacts) can be entered with a name and a number for each entry. **By default, Blank.**
- The numbers are stored index wise (as these are programmed index wise) and not in alphabetical order in the Phone Book. However, when the user uses Search Contact feature, he will get the entries in alphabetical order.
- A contact can be searched from the Phone Book using the feature ‘Search Contact’.

### Configuring:

- **Configuring the Phone Book through Phone.**
  - Phone Book is one of the options in the feature ‘Configuring the Phone Menu’.
  - When the user selects the option Phone Book, a menu with following options are available:

- Add new
  - Edit
  - Delete Specific Contact
  - Delete All Contacts
  - The user may select any of the menu option using up/down navigation key.
  - The user may press ← navigation key to go to previous step.
  - If the user selects an option 'Add New', phone will prompt the user to enter Name followed by Number.
- **How to use Web page?**
    - Please refer topic, '[Web Jeeves](#)' for more details. Click on "Phone Book".

Parameter	Value/Default	Description
Index	01 to 100	Number at which entry is made. Non programmable.
Name	24 digits max. Default = Blank. It can be ASCII Characters.	Enter here the Name of the Contact number whose SIP number is to be stored in the Phone Book.
Number	40 digits max. Default = Blank. It can be ASCII Characters.	Enter the Number of the contact party for whom the name is entered above.

### Important Point:

- While entering an IP address, '.' i.e. dot is entered (dialing \* is interpreted as dot only while dialing and not while editing or entering the number or address).

### Relevant Topics:

1. [Speed Dial](#) 136
2. [Know Your SETU VP236](#) 17
3. [Key Assignment](#) 87
4. [Search Contact](#) 126

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## Receiving a Call

### Introduction

- Normally Incoming Calls to the phone will be on the first available line of SIP1 or SIP2.
- The associated LED of the line-key (if assigned) will glow and the phone will ring.
- If there are multiple lines ringing, you can select the line, which you would like to answer.
- At the end of the call, hang up the handset to disconnect the call.
- Maximum calls the IP phone user can receive simultaneously is equal to the number of CAs programmed. This means, if number of CAs programmed are '4', user can receive maximum of four calls simultaneously.
- For Example, if number of CAs programmed are 4, first incoming call will occupy the CA1.
- Second incoming (waiting) call will occupy the CA2. If user answers the call on CA2, the current call will be put on hold and user will come in speech on CA2.
- Third incoming (waiting) call will occupy the CA3. If user answers the call on CA3, the current call (on CA2) will be put on hold and user will come in speech on CA3.
- Fourth incoming (waiting) call will occupy the CA4. If user answers the call on CA4, the current call will be put on hold and user will come in speech on CA4. Now, user has four held calls and user can toggle between the calls as per the requirement. All further IC calls will be rejected.

### Ways of answering a Call:

- Using the Handset
- Using the Headset
- Using the Speaker, If Auto Answer enabled
- Pressing the Call Appearance Key

### Using the Handset:

- This is a most common method like analog telephones. You can just pick up the handset before expiry of rings and start talking to the caller.

### **Using the Headset:**

- In this method, you can press the Headset key and start talking.

### **If Auto-Answer enabled:**

- This is a special feature, enabled on the Web page programming or by the programming of key. After some time, call will be answered automatically by the Phone and you will get audio of calling party through head phone.

### **Using the Speaker:**

- Like normal analog phone, you can press the speaker key and start talking.

### **Pressing the Call Appearance key:**

- When Incoming call comes and you are busy with an outgoing call, the corresponding Call Appearance Key LED is glowing.
- Press the key and start talking.
- Refer chapter '[Call Appearances](#)' for more information.

### **How it works?**

#### **How user gets information of the IC call ?**

- When Phone is in Idle condition, ringing indication is available on the Phone.
- When LED of the Call Appearance (CA) key on which the call has arrived, is glowing/blinking. Refer topic '[Call Appearances](#)' for more details.
- Getting the display of the Calling Party's Number/Name (CLI) on the LCD.

### **When user can answer the incoming call?**

- When phone is Idle.
- While user is talking to another user.
- When one user is held during a speech to the called party.
- When more than one user is held.

### **Important Points:**

- If the user is in speech with a remote user and if another call arrives which can be treated as waiting call, Auto Answer will not be effective. The user will get Call Waiting indication.

- 
- Auto Answer is effective only if there is no held call and the user is idle.
  - If the user is trying to configure some feature. That means, the phone is under 'Configuration', then the Incoming call cannot be answered. But if still the user wants to answer the call, he will have to stop configuring the Phone, exit the SE Mode and should go OFF-Hook.
  - While using headset, if user unplug the headset, call is not disconnected. User can again plug the headset and start conversation.
  - Since, the Phone has two DSP channels and remote held call uses the DSP channel, maximum of two calls can be received if the same calls are held by remote user after answering the call. Refer chapter 'Call Hold' for 'Remote Held'.
  - However, if there is an incoming/ waiting call when user has two calls held by remote users, user will still get incoming/waiting call notification on relevant CA. But, it will not be possible to retrieve the waiting call since no resources are available to entertain the waiting call. But to receive the waiting call, user will have to release one of the calls held by remote user.
  - It is possible to answer the waiting call when user has Active Conference. Maximum two calls can be answered when the user has active conference.
  - When user has active conference which will occupy two CAs and there is a waiting/ incoming call, user can answer the waiting call and the two calls involved in conference will be put on 'hold'. User will come in speech with the waiting caller.

**Relevant Topics:**

1. [Switching Between Handset, Headset and Speaker](#) 142
2. [Making Second Call](#) 99
3. [Call Appearances](#) 49
4. [Getting Started](#) 27
5. [Call Hold](#) 56

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## Rejecting a Call

### Introduction

- Some times when user is busy with an important meeting and he gets an Incoming call on his SETU VP236 then he would like to reject that call.
- Matrix SETU VP236 supports this feature by providing a programmable key on the Phone, to reject the call.
- It can be programmed from Web page 'Keys Programming'. Call can be rejected by menu option, also.
- Assign the key with No LED.

### How to Reject a call?

- First answer the required IC/Waiting call.
- Press the 'Reject' key to reject all other IC call/Waiting call.
- You can also select 'Rejecting the Call' from the 'Menu' options to reject the call.

### How it works?

- If there is one IC call then on pressing the 'Reject' key, the call can be rejected.
- If there are more than one IC calls then on pressing the 'Reject' key all the calls will get rejected.
- Hence you have to first answer the waiting call on any CA and then reject the other calls. For example, If user is in speech on CA1 and has waiting calls on remaining CA2 to CA4 and user wants to answer the call on CA2 and reject calls on CA3 to CA4, then he should first answer the waiting call on CA2 and then use 'Reject' feature to reject calls on CA3 and CA4.
- When the phone is ON-Hook and has IC call, remote held call and local held call, then priority will be given to Remote Held Call, to answer the call first when user goes OFF-Hook as shown below:
  - Priority 1 : Remote Held Call
  - Priority 2: IC Call
  - Priority 3: Held Call
- A call on CA which is busy can also be rejected.
- After the 'Reject' key is pressed or desired option is selected from the menu, the call is rejected and phone will display normal condition.

---

**How to program?**

- Click on “**Keys Programming**” on the Web page assign a Key Number to ‘reject’ feature. Use this key for rejecting the call.
- Navigate the Menu key and select “Rejecting a Call’ and then press “Enter” to reject a call. Refer the topic ‘Menu’ to know how to reject a call from ‘Menu’.

**Response during rejecting a call:**

The IP Phone will generate and send to remote party, following responses during rejecting a call:

- For Manual Rejection : Request Terminated.
- For Black Listed caller’s list : Forbidden.
- For Do Not Disturb : Temporarily not available.
- For Anonymous Call Rejection : Forbidden.
- For Call Reject Rejection due to any hardware limitation : Busy here.

**Important Points:**

- User must assign a key which does not have LED.
- Do not assign key numbers 06 to 09 for this feature, which have dual color LED.
- Function of rejecting a call is same while using key or ‘Menu’.

**Related Points:**

1. Call Appearances 49
2. Menu 100
3. Key Assignment 87

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## Rings

### Introduction

- When Incoming call comes for the SETU VP236, the user will get fixed ring cadences as per the country.
- Matrix SETU VP236 supports specific ring cadences for different countries. Specific ringer tones can also be programmed in the system.

- Select the Name of country and Ringer Tone Type from the Web page.

### How it works?

- The ring is played over the Phone, as per the ring cadence specified by the country selected. Refer the table at the end of the chapter, where ON1 and OFF1 is ON/OFF time for the first cadence and ON2, OFF2 is the same for the second cadence.
- The tone of the ring will be as programmed.

### How to program?

- Click on Web Page “**System Parameters**”. Refer chapter ‘System Parameters’ for more details.

Code	Country	CADENCE (In Seconds)			
		ON1	OFF1	ON2	OFF2
01	Australia	0.4	0.2	0.4	2
02	Brazil	1	4		
03	China	1	3		
04	Egypt	2	4		
05	France	1.5	3.5		
06	Germany	3.5	5.5	0.79	1.1
07	Greece	1	4		
08	India	0.4	0.2	0.4	2
09	Israel	2	3		
10	Italy	1	4		
11	Japan	1	2		
12	Korea	1	3		
13	Malaysia	2	3		
14	New Zealand	2	3		
15	Poland	2	3		
16	Portugal	1	5		
17	Russia	1	3		
18	Singapore	0.4	0.2	0.4	2
19	South Africa	0.4	0.2	0.4	2
20	Spain	1.5	3		
21	Thailand	2	3		
22	UAE	2	3		
23	UK	0.4	0.2	0.4	2
24	USA/Canada	2	4		

**Remarks:**

- Waveform is Sine Wave.

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## Search Contact

### Introduction

- While using SETU VP236, the user can make a call, after searching the contact. He can dial out the number by pressing a key. This advanced feature of the Phone is called “Search Contact”.
- Assign the key with No LED.
- Press ‘Search Contact’ key to display the contacts stored in the Phone Book as per their names in alphabetical order.

### How it works?

- When the user presses any of the dial pad key, the names matching with the characters of that key is displayed.
- For example, after pressing the ‘Search Contact’ key, the user presses ‘6’, the contact starting with ‘M’ are displayed. However, now if the user presses ‘6’ one more time the user will get contact starting with ‘M’ only and not with N. Refer Table of characters in chapter ‘[Know Your SETU VP236](#)’.
- If the user presses ‘5’ and the contact starting with ‘J’ is not available in the phone book, the user will get the contacts starting with ‘K’ and so on.
- When user presses enter key on any of the contacts, the number stored against the name of the contact is displayed.
- When the user presses ‘Up/Down’ key, other contacts are displayed.
- Finally user can make a call by going OFF-Hook or pressing the speaker key/headset key.

### Configuring:

- Click on Web page “Keys Programming” to assign a key for this feature.
- You can also select ‘Search Contact’ by navigating options in ‘Menu’ on Phone.

### Related Topics:

1. [Phone Book](#) 117
2. [Know Your SETU VP236](#) 17

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## Selective SIP Line

### Introduction

- You can select a specific line and then make an OG Call. This is called “Selective SIP Line” feature.
- Matrix SETU VP236 supports this feature by providing programmable keys on the Phone for both the SIP Lines: SIP1 and SIP2.
- The speech and display on the LCD are just like a normal OG call which was initiated without selecting SIP1/SIP2.
- Assign a key with single color LED for this feature.
- To use this feature:
  - Press the SIP1 or SIP2 key and get the dial tone. Then dial digits and press “Enter” for OG Call.
  - You can also, select the SIP1 or SIP2 option from the Menu, by navigating the parameters with up/down key and get the dial tone to dial a number.

### How it works?

- When the user presses SIP1/SIP2 key or make a selection using the options in the menu:
  - User gets dial tone.
  - Display on LCD for respective CA-row will be as: Dial....
  - The LED of the SIP1/SIP2 key glows.
  - Display on LCD in the call status characters of the respective CA-row will be as: SIP1/SIP2.

### How to program?

- Refer chapter [Key Assignment](#), to assign key for SIP1 or SIP2.
- Select ‘Access SIP1 Trunk’ or ‘Access SIP2 Trunk’ by navigating ‘Menu’ on phone.

### Related Topics:

1. [Key Assignment](#) 87
2. [Menu](#) 100
3. [OG Call Route Selection](#) 110

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## SIP Parameters

### Introduction

- Matrix SETU VP236 supports a feature ‘SIP Parameters’, by which some parameters of SIP trunks like registrars, Port addresses can be programmed.
- It can be programmed by Web Page. But some parameters like SIP ID, Registrar Server Address, etc. are also programmable from Phone so that the user can start using the Phone without the need of a computer to configure it.
- Refer [Glossary](#) and [VoIP Basics](#) for explanation of Voice Compression (codecs), SIP, RTP, STUN, OUTBOUND Proxy, etc.

### How it works?

Matrix SETU VP236 supports the SIP parameters:

- Configurable by Web
- Configurable by Phone

Parameters configurable by Web using SIP1 and SIP2:

- SIP Account-Enable/Disable
  - Display Name
  - SIP ID
  - Registrar Server Address
  - Registrar Server Port
  - Re-Registration Timer
  - Registration Retry Timer
  - Authentication User ID
  - Authentication Password
  - Vocoder Preference
  - DTMF Option
  - Outbound Proxy-Enable/Disable
  - Outbound Proxy Server Address
  - Outbound Proxy Server Port
  - Use STUN?-Yes/No
- 
- SIP parameters programmable from phone from ‘**SIP1 Parameters**’ or ‘**SIP2 Parameters**’ after entering ‘Config Phone Menu’.

- 
- SIP Status
  - SIP ID
  - Registrar Server Address
  - Registrar Server Port
  - Authentication User ID
  - Authentication User Password

### How to use?

- For programming by the Phone, Navigate Menu → Config Phone → SIP1 Parameters.
- Refer Web Jeeves. Click on Web page “SIP Parameters”.

### **Registrar Settings:**

Program following parameters for both SIP1 and SIP2.

- **Status:**

It shows which SIP line is enabled/disabled:

- Select ‘enabled’ if you want to make OG calls/receive IC calls from this SIP line. If it is enabled, corresponding characters ‘SIP1’ or ‘SIP2’ will be flashing on the LCD display, with a cadence of ON/OFF.
- Select ‘disabled’ if you do not want to allow SIP line to make OG calls even if it gets selected by the OG call routing logic. Also IC calls will not be received.
- **By default, Status is Disabled.**

- **Display Name:**

Enter here the name which you want the called party to receive on his Phone.

Maximum 24 characters can be entered. ASCII characters can be used. **By default, Display Name is Blank.**

- **SIP ID:**

SIP ID is a full SIP URI. This can be a number or text. Maximum of 24 characters can be entered with all ASCII characters. A SIP URI provided by the ITSP can be 12345@abc.com or anil@abc.com . When the user part is a text, the SIP ID of a user is distinguished from his mail ID or any other ID by a SIP Tag. i.e. SIP: anil@abc.com. **By default, SIP ID is Blank.**

- **Registrar Server Address:**

SIP Registrar Servers' Address is the address of the SIP Registrar. It can be IP address also. This may same as the SIP Server address since the SIP Server might also be acting as Register Server.

40 characters maximum with the extended ASCII characters.

***By default, Registrar Server Address is Blank.***

- **Registrar Server Port:**

SIP Registrar Servers' Port is the Registrar Server's listening port for SIP. User may change this field if the ITSP provides a SIP port number other then the default.

Valid range = 1024-65535. This may be same as SIP Server's Port address.

***By default, Registrar Server Port is 5060.***

- **Re-Registration Timer:**

- Normally, the Register server deletes an entry of its client from its database on expiry of a fixed timer. This timer is set by the Register Server. Hence, in order to be registered always, the IP Phone should send a Registration request before this timer expires.

- This timer signifies the time after which the SETU VP236 should send Registration request again to be registered. To be on safe side as a rule, the SE should program this time = half of the Registration time set by the Register Server. Suppose, the Register Server deletes the entry after 30 minutes of registering a client. In such case, the SE should program this time=15minutes. Re-Registration Timer range is from 00001 to 65535.

***By default, Re-Registration Timer is 3600 seconds.***

- **Registration Retry Timer:**

On expiry of Re-Registration Timer, send the registration request. If the registration attempt fails, send the registration request on expiry of this timer. Phone will send the registration request till IP Phone gets registered and once it gets registered again Re-Registration Timer will be started.

Registration Retry Timer range is from 00001 to 65535.

***By default, Registration Retry Timer is 10 seconds.***

## Authentication

- **Authentication User ID:**

This is the user name for registering the SIP account with the SIP register server.

Enter maximum of 16 characters with all ASCII characters.

**By default, Authentication User ID is Blank.**

- **Authentication Password:**

This is the password associated with the user name above.

Maximum of 16 characters can be entered with all ASCII characters.

**By default, Authentication Password is Blank.**

- **Vocoder Preference:**

- To compress bandwidth of data in the RTP Packets, various codecs are used.

- As per optimization criteria of bandwidth and speech quality, the coders are supported by the phone like G.722, G.723, etc.

- While call is established, the vocoder is selected as per autonegotiation with remote SETU VP236. This parameter signifies the Vocoders in preferential order which can be requested to the remote user while making OG call.

- In case of IC calls, the preference requested by the remote end will be considered. If none of the Vocoders requested by the remote end is supported by user, the call is dropped and logged as rejected call in Call Logs.

Select the vocoder from the options shown below:

### **For SIP1 and SIP2 Default Vocoder Preferences**

1 <sup>st</sup> Pref.	2 <sup>nd</sup> Pref.	3 <sup>rd</sup> Pref.	4 <sup>th</sup> Pref.	5 <sup>th</sup> Pref.	6 <sup>th</sup> Pref.	7 <sup>th</sup> Pref.	8 <sup>th</sup> Pref.	9 <sup>th</sup> Pref.	10 <sup>th</sup> Pref.
<b>G.729</b>	G.729								
G.723-5.3	<b>G.723-5.3</b>	G.723-5.3							
G.723-6.3	G.723-6.3	<b>G.723-6.3</b>	G.723-6.3						
G.722	G.722	G.722	<b>G.722</b>	G.722	G.722	G.722	G.722	G.722	G.722
G.726-16	G.726-16	G.726-16	G.726-16	<b>G.726-16</b>	G.726-16	G.726-16	G.726-16	G.726-16	G.726-16
G.726-24	G.726-24	G.726-24	G.726-24	G.726-24	<b>G.726-24</b>	G.726-24	G.726-24	G.726-24	G.726-24
G.726-32	G.726-32	G.726-32	G.726-32	G.726-32	G.726-32	<b>G.726-32</b>	G.726-32	G.726-32	G.726-32
G.726-40	<b>G.726-40</b>	G.726-40	G.726-40						
PCMA	<b>PCMA</b>	PCMA							
PCMU	<b>PCMU</b>								

The user may select all the ten preferences to be same.

Default Vocoder is shown as bold.

**For Example:**

If the user selects G.729 in all the preferences then if the remote user supports this Vocoder, then only it will accept the request and the call will be through. But if the remote user does not support the request, the call shall be rejected.

• **DTMF Option:**

This parameter will decide how the DTMF digits will be sent over SIP.

This option is used while making OG calls. In case of IC calls, the option requested by the remote end will be considered.

Three options are provided.

- **RTP (RFC 2833)**
- **SIP (INFO)**
- **In-band**

***By default, DTMF option is SIP (INFO).***

**Outbound Proxy:**

***Status:***

Select from Yes/No if you want to enable Outbound Proxy and if your ITSP has this server to handle voice calls.

***By default, Outbound Proxy is No.***

• **Outbound Proxy Server Address:**

This specifies the Outbound Proxy Servers' Address. It can be IP address also. This may be same as the SIP Server address since the SIP server might also be acting as Enter here the address of Outbound Proxy of 40 characters maximum It can be with Extended ASCII characters.

***By default, Outbound Proxy Server Address is Blank.***

• **Outbound Proxy Server Port:**

This specifies the Outbound Proxy Server's listening port for SIP. User may change this field if the ITSP provides an Outbound Proxy Server port number other than the default. This may be same as SIP Server's Port address.

Enter from the Range = 1024-65535.

***By default, Outbound Proxy Server Port is 5060.***

---

## **STUN**

### ***Use STUN:***

This parameter specifies whether STUN will be used for this SIP trunk.

Select “Yes” if STUN is to be used to get the IP address.

Select “No” if STUN is not to be used.

***By default, STUN is No.***

### **Important Point:**

- Refer “[Network Port](#)” for other parameters like, IP address and the port number for the STUN server.

### **Relevant Topics:**

1. [Configuring SETU VP236](#) 16
2. [Web Jeeves](#) 168
3. [Know Your SETU VP236](#) 17
4. [Network Port](#) 102

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## Software Upgrade

### Introduction:

- Matrix SETU VP236 supports to upgrade the IP phone for desired config files, drivers, firmware, call logs, html files using FTP server.
- File Transfer Protocol (FTP) is used for exchanging files over LAN/WAN.
- It is the most frequently used protocol to download file from a FTP server or to upload file to the FTP server.
- FTP server is also required to make desired configuration files which can be downloaded using Auto Configuration feature which is used by ITSPs.
- For Example, if user purchases the IP phone from ITSP, ITSP will be administrator of IP phone and customer will be user of IP phone. It is assumed that ITSP will not reveal config password to their customer to restrict them to use some feature/settings. It is also assumed that ITSP will retain the customer wise record of config password of IP phone units for future reference. In case, ITSP engineer needs to configure/repair IP phone at customer site and forgot config password, then he can make the password configuration file (using feature 'Software Upgrade') using another IP phone and copy/paste that file into the user folder stored in TFTP server. Now, after restarting the IP phone, IP phone will download the configuration files from TFTP server using 'Auto Configuration' and update the config password to the desired one as changed by ITSP.

### How to use it?

- SE/Admin can view and modify the relevant files of Firmware, Drivers, Web Pages, Configuration and Call Logs.
- Click on '**Software Upgrade**'.
- You will get the FTP window which displays the following folders:
  - Appl
    - Clicking '**Appl**' folder will display following files:
      - Mxipphone-Main application firmware
      - Mxhttpd-Web Server
      - Boardcnfg-Configuration of Ethernet interface

- 
- Driver
    - Clicking '**Driver**' folder will display the following files:
      - Hapi.o-DSP driver
      - Mxlcd-LCD driver
      - Mxlcd.o-LCD driver
  - Webpage
    - Clicking '**Webpage**' folder will display all the html files of Web Jeeves.
  - Config
    - Clicking '**Config**' folder will display all the configuration files which IP phone is currently using.
  - Logs
    - Clicking '**Logs**' folder will display all the call logs files.
    - These files are required in typical conditions when user wants to store the few call logs for future reference.
  - Clicking any folder will display the relevant files which IP phone is currently using.
  - SE/Admin can cut, copy and paste desired files here.
  - Finally new configuration/ driver/ firmware will take effect only during next power ON.

**Relevant Topic:**

1. Auto Configuration 44

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## Speed Dial

### Introduction

- Using this feature it is possible for the user to dial a number simply by pressing a key instead of pressing long digits of the number string of the Phone Book.
- Assign a key with No LED from web page.

### How to use it?

- Press the key which is assigned Index Number of 'Phone Book'. The number string programmed in the phone book will be dialed out by the phone.

### How it works?

- SETU VP236 has a database of contacts viz. Phone book. This database is used to implement this feature.
- While configuring the keys, the user may select Phone Book as a feature for the key and the respective Index in the address book.
- Since SETU VP236 supports a maximum of 18 programmable keys, maximum of 18 phone numbers/addresses among the address book entries can be programmed as speed dial numbers.
- When the user presses the speed dial key, the number is displayed on the LCD and the call is made.
- If the user programs an index of the address book which does not have a number, then he will get display 'No Number Programmed' and will get error tone if that number is dialed.

### How to program?

- Ensure that 'Phone Book' is programmed as required.
- Refer the topic '[Web Jeeves](#)' for more details. Click on '**Key Programming**'.
- Select 'Key Number' which has no LED for which 'Speed Dial' feature is required.
- **Feature:** Select 'Phone Book' in this field.
- **Phone Book Index Number:** Select Index Number of the number to be dialed, as per Phone Book.

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**Relevant Topics:**

1. Key Assignment 87
2. Phone Book 117

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## Status

### Introduction

- Using 'Status Page' you can have quick glance for some Network Parameters and SIP Parameters. You can change it from respective page, if required.
- It also provides 'status' and 'statistics' parameters of Auto Configuration.
- You can also get the status on the display of phone, by accessing 'Network Status' from the 'Menu'.
- Click on 'Status' page. Following basic parameters are displayed:

Parameters	Display	Description
<i>System Name</i>	Matrix SETU VP236	This displays the name of system
<i>Version/Revision</i>	VxRy	This displays the current version 'x' and Revision 'y' of software
<b>Network Parameters</b>		
<i>IP Address</i>	192.168.001.161	This displays the static address or address located by DHCP or PPPoE
<i>Subnet Mask</i>	255.255.255.0	This display the programmed subnet mask
<i>Gateway Address</i>		This displays the programmed gateway address
<i>DNS Address</i>		This displays the programmed Domain Name Server Address
<i>DNS Domain Name</i>		This displays the 'Name' of the Domain Name Server.
<i>NTP Address</i>		This displays the programmed SNTP server Address used by phone

<i>MAC Address</i>	Unique Address for the IP Phone	This displays the MAC Address for the Network Port programmed at factory
<i>NAT Type</i>	Blank display or Unknown or full cone or Restricted or Port Restricted or Symmetric or Symmetric Behind Firewall or Blocked	NAT is configured for option as shown. (1) If STUN is not enabled, it is Blank (No Display) (2) if none of the type of NAT is detected by VP236, then 'Unknown' is displayed.
<i>Public IP</i>	N.N.N.N. N = 001 to 255	When STUN is enabled 'Public IP' is displayed here.
<b>SIP Account 1:</b>		
<i>Registration Status</i>	Not Registered/ Registered	When 'Register' button is pressed the IP Phone sends registration request to the server and the 'Status' is displayed, here as 'Registered' or 'Not Registered'.
<i>Registration Time</i>		This time is provided by server during registration and displayed here
<i>Registration Retry Count</i>	0,1,2,.....	This shows how many time after failure, IP Phone sent registration request. As retry count increase this number displayed as increasing 0,1,2,.... After the page submit this will get reset to '0'.

Matrix

<i>Registration Last Failed Reason</i>		This displayed the reason of last time failure. For example, Message sent failure. Response timeout. Failed response 486, etc.
<b>SIP Account 2:</b>		
<i>Registration Status</i>	Not Registered/ Registered	When 'Register' button is pressed the IP Phone sends registration request to the server and the 'Status' is displayed, here as 'Registered' or 'Not Registered'.
<i>Registration Time</i>		This time is provided by server during registration and displayed here
<i>Registration Retry Count</i>	0,1,2.....	This shows how many time after failure, IP Phone sent registration request. As retry count increase this number displayed as increasing 0,1,2,.... After the page submit this will get reset to '0'.
<i>Registration Last Failed Reason</i>		This displayed the reason of last time failure. For example, Message sent failure. Response timeout. Failed response 486, etc.

<b>Auto Configuration</b>		
<i>Status</i>	Blank	Message displays after Auto Configuration is applied during Power ON: for e.g. 'No tftp Server Specified', 'Auto Configuration Fail', 'Auto Configuration Success'.
<i>Statistics</i>	Blank	Message displays after Auto Configuration is applied: for e.g. 'Files Downloaded/Configured =X/Y'.

**Relevant Topics:**

1. [Network Port](#) 102
2. [SIP Parameters](#) 128
3. [Auto Configuration](#) 44
4. [Menu](#) 100

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## Switching between the Handset, Headset and Speaker

### Introduction

- Matrix SETU VP236 supports talking and listening by Handset, Headset and Speaker/MIC. This feature allows you to switch between handset, headset and speaker during the call.
- Speaker mode or Headset mode can be used by pressing respective key.
- 'Spk' key is with LED indication.
- The number for Headset feature is configurable by Web page "Key Assignment".

### How it works?

- When using handset: (To switch to Headset and Speaker)
  - Press "Headset" key to switch to headset Mode Speech audio is connected to the Headset port. User can go ON-Hook irrespective of headset is connected or not.
  - Press the "Speaker" key to switch to Speaker Mode. The Speaker is turned ON. Replace the handset on the cradle.
- When using headset: (To switch to Handset and Speaker)
  - Lift the handset to use handset. While using headset, if user goes OFF-Hook, the voice will be available on the handset port.
  - Press the "Speaker" key to use speaker. The headset port is disabled and the speaker is turned ON.
- When using speaker: (To switch to Handset and Headset)
  - Lift the handset to use handset. While using speaker if user goes OFF-Hook, the speaker is turned OFF and the voice will be available on handset port.
  - Press "Headset" key to use headset. While user presses "Headset" key, the voice will be available on the headset port and the speaker is turned OFF irrespective of headset is connected or not.

### How to Configure?

- Click on Web page "Keys Programming".
- Assign a key to 'Headset' function.
- Refer chapter '[Key Assignment](#)' for more details.

**Important Points:**

- In headset mode, if user unplugs the headset, call will not be disconnected. User can again plug the headset and start the conversation on headset.
- **Using Headset:**  
To configure the phone to operate in '**Headset Mode**', the user can select the feature 'Headset ON-Hook/OFF-Hook' from the menu.
- **Using the Speakerphone:**  
To configure the phone to operate in '**Speakerphone Mode**' the user can press 'spk' key.

**Related Points:**

1. [Making a Call](#) 94
2. [Receiving a Call](#) 119
3. [Key Assignment](#) 87
4. [Menu](#) 100

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## System Parameters

### Introduction

This chapter describes important system parameters used by the SETU VP236 for its satisfactory operation. Refer corresponding chapter for more details.

### How to use Web-page?

Click on Web Page “System Parameters” and program the parameters as shown below:

Parameter	Value/Default	Description
<b>System Name</b>	Max. 24 char. Default Matrix SETU VP236	Enter here the name of phone you are using.
<b>CPTG Type</b>		
Selected country	List of countries Default = India	Select the ‘country name’ from the list of countries for country-specific CPTG.
<b>End of dialing</b>		
Enable End of dialing digit	Selected/Blank Default=Selected (Enable)	Do not select if you want to disable end of dialing feature.
Enable Inter digit wait timer for End of Dialing	Selected/Blank Default=Selected (Enable)	Do not select if you want to disable ‘Inter digit Wait Timer’ feature.
Fixed Number of digits for End of dialing	Selected/Blank Default=Selected (Disable)	Select if you want that system should dial out the number after entering programmed fixed, number of digits.
---do----	1-40 Default = ‘40’	This shows that system considers end of digit after receiving ‘40’ digits of number string.

Parameter	Value/Default	Description
<b>Keypad Lock</b>		
Auto keypad Lock	Set / cancel Default = Cancel	Select 'Set' if you want that your Phone keypad can not be operated after a time of 'Keypad lock timer' when Phone is in Idle state. You need not press key for 'keypad lock'. Else select 'Cancel'.
<b>LCD Settings</b>		
Backlight	Always ON, Always OFF, For 10 sec. (Default), 20 sec., 30 sec., 60 sec., 120 sec., 240 sec.	Select the option as per your requirement for LCD setting.
Brightness	1- 4, default = 2	Select level from 1 to 4. With '1' it is lowest brightness for suitable viewing
Contrast	1- 4, default = 2	Select level from 1 to 4. With '1' it is lowest contrast for suitable viewing
<b>Number of Call Appearances</b>		
No. of CA's	1- 4, default = 4	Select number as per your requirement of indication for waiting calls on the LCD. For indication of more IC calls, reject one of previous calls.
<b>Ring Type</b>		
Ringer Tone Type	0-9, Default = Ttype 6.	Select the tone type of ring for IC call
Select country	List of countries Default = India	Select the country name from list of country.

Parameter	Value/Default	Description
<b>Volume Setting</b>		
Handset Tx	0-7, Default = 4	Select volume of transmit speech as per your desired level.
Handset Rx	0-7, Default = 4	Select volume of received speech as per your desired level
Headset Tx	0-7, Default = 4	Select volume of transmit speech as per your desired level
Headset Rx	0-7, Default = 4	Select volume of received speech as per your desired level
Microphone Tx	0-7, Default = 4	Select volume of transmit speech as per your desired level
Speaker Rx	0-7, Default = 4	Select volume of receive speech as per your desired level
Ringer	0-7, Default = 4	Select volume of incoming ring as per your desired level
<b>Timers</b>		
ACB notification Timer	01-99, Default = 30 Min.	Refer chapter 'Timers' for definition
Auto Answer Timer	01-99, Default = 02 sec.	Enter time for which Phone will wait before auto answering the IC Call
Call Forward Timer	00-99, Default = 45 sec.	Refer chapter 'Timers' for definition. Refer chapter 'Call Forward'
Call Hold Alert Timer	000-999, Default = 180 sec.	Refer chapter 'Timers' for definition
First Digit Wait Timer	01-99, Default = 15 sec.	Refer chapter 'Timers' for definition

Parameter	Value/Default	Description
Hotline Timer	0-9, Default = 0	Enter here time delay after which you want your phone to dial the hotline address when OFF-Hook.
Keypad Lock Timer	01-99, Default = 15 sec.	Enter time after which keypad will be Automatically locked. Key can be unlocked by pressing assigned key.
Inter Digit Wait Timer	01-99, Default = 06 sec.	Refer chapter 'Timers' for definition
Ring Timer	01-99, Default = 45 sec.	Enter time up to which 'Ring' will be played after IC call comes for your Phone.
Transfer Notification Timer	01-99, Default = 60 sec.	Refer chapter 'Timers' for definition

**Relevant Topics:**

1. [Timers](#) 150
2. [Auto Answer](#) 40
3. [Hotline](#) 85
4. [Call Hold](#) 56
5. [LCD Settings](#) 92

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## Time Settings

### Introduction

- For the Matrix SETU VP236, the LCD display consists of Date and Time. Also some features like Call Logs use the actual time of call etc. But the Phone does not have Hardware design for the RTC (Real Time Clock). Hence the SETU VP236 needs to be up dated automatically for correct time at fixed time intervals.
- The SETU VP236 provides feature for time setting by using Simple Network Time Protocol (SNTP) to get time from the Time Server.
- Refer NTP in 'Glossary'.

### How it works?

- User has to program the time server in the Web page.
- Phone will get the time from the NTP server every 10 seconds (approx).
- The time server provides GMT (Greenwich Mean Time).
- The SETU VP236 is located at the country which is situated in the specific time zone.
- Based on the time zone settings, Matrix SETU VP236 shows correct time for that country. For example, if time shown by ntp1.cs.wisc.edu server for India (Calcutta, Chennai, Mumbai, New Delhi) is 10:44:50AM, then current time shown for Kabul is 09:44:50AM. Because for India GMT is 'GMT +5.30' but for Kabul it is 'GMT +4.30' (1 hour behind India). The date is displayed such as, 17May2006.
- **By default following three Time Servers are programmed:**
  - Ntp1.cs.wisc.edu
  - Time.windows.com
  - Time.nist.gov
- You can select any one option among three to obtain the date and time or may enter NTP Server address of your own choice.
- The countries for "Time Zone" supported are same as supported by Windows. **By default selection is Calcutta, Mumbai, Chennai and New Delhi.**

### How to use Web Page?

Click on Web page “**Time Setting**” and program Time Server and Time Zone. Refer [Web Jeeves](#).

Parameter	Value/Default	Description
Current Date	DD-Month name-YYYY. Default = 01 January 2007	Phone will display here the current date available from Server
Current Time	HH:MM:SS. Default = 10:00:00	Phone will display here the current time available from Server
NTP Address	Ntp1.cs.wisc.edu Time.windows.com Time.nist.gov default = Ntp1.cs.wisc.edu	Select suitable Time server or Enter the desired NTP server address for 'time' display on the Phone
Time Zone	List of places as per MS Windows. Default = Calcutta.	Select the place name as per the country where Phone is working.

### Relevant Topic:

1. [Daylight Saving Time \(DST\)](#) 76

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## Timers

### Introduction

- SETU VP236 supports various kinds of timers. For example, if user waits for some period after entering digits then instead of expecting required number of digits, system will automatically dial out the number string after a fixed time interval. This time is called “Inter Digit Wait Timer”.
- Matrix SETU VP236 is designed with following types of timers:
  - ACB Notification Timer
  - Transfer Notification Timer
  - Auto Answer Timer
  - Ring Timer
  - First Digit Wait Timer
  - Inter Digit Wait Timer
  - Call Hold Alert Timer
  - Call Forward Timer
  - Hotline Timer
  - Key Pad Lock Timer

### Explanation:

- **Notification Timers:** These timers are related to response from the phone while setting a feature.
  - **Transfer Notification Timer:** This timer signifies the time for which the SETU VP236 should wait for the notification of transfer function from the transferee. This timer is programmable.
  - **ACB Notification Timer:** This timer signifies the time for which the SETU VP236 should wait for notification for ACB Subscribe ACB ‘ready for call completion’ notification. This timer is programmable.
- **Ring Timer:** This timer signifies the time for which the SETU VP236 will give ring indication, when incoming call comes. When ring will be stopped the Phone will go to idle state.
- **First Digit Wait Timer:** This timer signifies the time for which the

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SETU VP236 should wait for first digit. On expiry of this timer, phone will go to idle state or standby state. This timer is programmable.

- **Inter Digit Wait Timer:** This timer signifies the time for which the SETU VP236 should wait between two digits. On expiry of this timer, the number is considered to be complete and is taken up for further processing. This timer is programmable.
- **Call Hold Alert Timer:** This timer signifies the time after which the SETU VP236 will remind the user about held call. This timer is programmable.
- **Call Forward Timer:** This timer signifies the time after which SETU VP236 will forward the call to programmed destination number. This timer is programmable.

For other timers, refer chapter '[System Parameters](#)'.

### **Programming:**

All the timers are programmable from Web only. Click on the Web page '[System Parameters](#)'.

### **Relevant Topic:**

1. [System Parameters](#) 144

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## User Settings

### Introduction

- This chapter explains how you can set the features which are frequently used, like Auto Answer, CLIR, etc. using Web page. It is also possible to set some of the features using 'Menu' key available on the SETU VP236.
- Refer corresponding chapter for more details.
- Refer '[Web Jeeves](#)'. Click on '**User Setting**'.

### Which features can be configured?

<b>Feature</b>	<b>Description</b>
Anonymous Call Rejection	Select 'Set' or 'Cancel '. Default = Cancel
Auto Answer	Select 'Set' or 'Cancel '. Default = Cancel
Black Listed Caller List	Enter Number from which you do not want to receive a call. Max. 10 numbers can be entered. with 40 characters max. Default = blank
Call Forward	Refer chapter Call Forward to program the features: CF-Busy and CF-No Reply.
CLIR	Select 'Set' or 'Cancel '. Default = Cancel
Do Not Disturb	Select the required option : DND All, DND List, Cancel. Default = Cancel.
DND List	Enter a number which you want to allow as Incoming call on your Phone even if you have set DND feature. Upto 40 characters can be entered. May be with ASCII. Max.10 numbers can be entered. Default = Blank
Hotline	Refer chapter "Hotline"
Hotline Address	Refer chapter "Hotline"
<b><i>Voice Mail Account1</i></b>	
VM Subscription Address	Enter Max. 40 characters for VM subscription address-1 available from ITSP. It can use all ASCII characters. <b>Default = Blank.</b>

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VM Retrieval Address	Enter Max. 40 characters for VM retrieval address-1 available from ITSP. It can use all ASCII characters. <b>Default = Blank</b>
<b>Voice Mail Account2</b>	
VM Subscription Address	Enter Max. 40 characters for VM subscription address-2 available from ITSP. It can use all ASCII characters. <b>Default = Blank.</b>
VM Retrieval Address	Enter Max. 40 characters for VM retrieval address-2 available from ITSP. It can use all ASCII characters. <b>Default = Blank</b>

**Relevant Topics:**

1. Call Forward 52
2. Hotline 85
3. Voice Mail 157

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## VLAN Tagging

### Introduction

- LAN is defined as a single broadcast domain with Standard (**IEEE 802.1Q**). VLAN is Virtual LAN.
- If a user broadcasts information on his/ her LAN, the broadcast will be received by every other user on the LAN. VLANs allow to logically segmenting a LAN into different broadcast domains.
- In order to extend VLANs across different switches, a link called 'trunk' must interconnect the switches.
- Now, when you want traffic from multiple VLANs to be able to cross a link that interconnects two switches, there is a need to configure a VLAN tagging method on the ports that supply the link. A frame is tagged when it leaves the switch with information about the VLAN that the frame belongs to.

### How it works?

- IP phone has three port inbuilt switch. These ports are CPU port, Network port and PC port. Network port of IP phone will be connected to the external switch in which multiple VLANs will be configured.
- If you enable the VLAN, the IP phone will tag all the packets leaving the Network and PC port and also receives only packets on PC port and Network port with VLAN tag.
- Now, if you enable the VLAN by mistake and PC/ switch to which IP phone is connected does not support VLAN tag, you will not be able to access the Jeeves for further configuration. This is because the Phone will expect only tagged packets and ignore all the packets which are untagged.
- Hence the Phone supports to enable/disable the VLAN, from the Jeeves as well as 'Configuring the Phone-Menu', so that in above case if Jeeves can not be accessed, VLAN can be disabled from the 'Configuring the Phone-Menu' and then Jeeves can be used for further configuration.

### For Example:

- The example will be useful for programming of the feature.
- Suppose 4 VLANs are configured in external switch, VLAN1

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(Marketing), VLAN2 (Accounting), VLAN3 (R&D), and VLAN4 (HR). To limit the broadcasting only to marketing department, VLAN Id configured in IP phone will be '1'. Now, when the frames will leave from the network port of IP phone, IP phone will add the VLAN tag in frames with VLAN Id1. When these frames will reach the port of external switch/ router to which IP phone is connected, it will read the VLAN header and identify that frame belong to VLAN1 (marketing) and will forward the frame to VLAN1.

- In above example, if you want to shift the user who is using IP phone to the other department say HR, just change the VLAN Id to 4 and IP phone will tag the frames leaving network port with VLAN Id 4.
- Thus following parameters are required to be programmed for this feature:
  - Option for using (adding) VLAN Tag-Yes or No.
  - VLAN ID.

### How to program?

- Refer Web Jeeves.
- Click on 'Network Parameters'.
- The option to use VLAN tag or not (Yes/No) and VLAN ID, can be programmed from web page.
- Refer chapter '[Network Port](#)' for programming details.
- 'VLAN Setting' can be enabled/disabled by navigating:
- Menu > Config Phone Menu > Network Parameters > VLAN Setting. 'VLAN Setting', provides two parameters:
  - Status
  - VLAN IDSelect 'Status' to access enable/disable.  
Select 'Enable' to use VLAN feature.
- After selecting 'VLAN ID', program it from range 0 to 4095.  
**Default=0.**

### Important Point:

- IP phone will not tag the packets coming from PC port and leaving the network port of the 3-port switch built-in to the Phone.

**Relevant Topics:**

1. Network Port 102
2. Configuring the Phone-Menu 75

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## Voice Mail

### Introduction

- SETU VP236 supports to notify you about a new message in your mailbox. (This is possible only if it receives a notification from the Voice Mail Server from ITSP). This feature is called “Voice Mail”. (VM) Matrix supports two VM servers.
- You can also access the mailbox by selecting a user option (Menu).
- You can assign separate key with single color LED to access VM Account1 or VM Account2. Two Voice Mail Server Address are programmable from Web, also.

### How to use it?

To access the mailbox:

- Press a separate key assigned to access Voice Mail Account1 or Voice Mail Account2.
- You can also, select the option ‘Call VM Server1’ or ‘Call VM Server2’ from the Menu.

### How it works?

- This feature is implemented with two requirements viz. Voice Message Notification and Voice Message Retrieval. Voice message Notification is by the LED whereas, Voice Message Retrieval is through any of the two programmed keys.

### Voice Mail notification from VM Server

- You can get the voice mail notification message from VM servers in any of the port states of the Phone.
- On receipt of notification message from either or both VM servers:
  - LED of the key will glow Red.
  - The message on the LCD will be displayed.
  - Default VM key is for ‘VM Account1’.

### Voice Mail retrieval from VM Server:

- User can program the separate keys for VM Account1 and VM Account2 from web jeeves.
- For each voice mail account following two parameters can be programmed from Web page:

- VM Subscription Address
- VM Retrieval Address
- The system will send the VM subscription request to the address programmed for the 'VM Subscription Address' for the corresponding VM account.
- You can retrieve the voice mail/messages:
  - From Menu
  - Using a key
- On pressing any of the separate key for 'Call VM Server1' or 'Call VM Server 2':
  - If there is **no** notification from VM servers, the display will be as 'Call VM Server1' & 'Call VM Server2'. You can call the selected Server.
  - If there is notification from **one** of the VM servers say from VM Server1, on pressing 'Call VM Server1' key, the Phone will display 'New message in VMS1' & 'Call VM Server2'. If you select the first option, voice mail server1 will be called and the LED of the key will be turned OFF, the display will be changed to 'Call VM Server1'. Also, displayed notification message on LCD will be removed.
  - If there is notification of new mail/message from **both** the VM servers, the Phone will display 'New message in VMS1' & 'New message in VMS2'. If you select one of the option say VM server2, VM Server2 will be called. Now if user selects 'New message in VM Server1', VM Server1 will be called. LED will be turned off and display will be changed to 'Call VM Server1'. Also, displayed notification message on LCD will be removed.
- If a key is not assigned, then you can access voice mail box from the Menu by selecting the option 'Call VM Server1' or 'Call VM Server2'.

### Configuring:

Following two parameters can be configured to make this feature work:

- Programming of the Voice Mail Server Address. Refer topic '[Web Jeeves](#)', page "User Setting". Refer chapter '[User Settings](#)' for programming of parameters.
- For assigning a key. Refer topic '[Key Assignment](#)'. Assign separate key with single color LED to 'VM Account1' and 'VM Account2'

- feature.
- To access from Menu:
  - Select the option 'Call VM Server1' or 'Call VM Server2' and press 'Enter' key.
  - System will dial out voice mail server address.

**Important Points:**

- All the features like hold, transfer, conference, etc. applicable to a normal call are applicable in this case also as this call is also considered as a normal call (except that it is made to a Voice Mail Server and not to a human).
- Follow the prompts for the voice mail system to listen to and delete your voice mails.

**Relevant Topic:**

- User Settings 152

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## Volume Settings

### Introduction

- SETU VP236 supports to increase or decrease the speech audio level of the call.
- You can set:
  - **Handset Transmit Volume:** So that your voice is heard clearly by the remote user while using handset.
  - **Handset Receive Volume:** So that the remote user's voice is heard clearly by you while using the handset.
  - **Headset Transmit Volume:** So that your voice is heard clearly by the remote user while using headset.
  - **Headset Receive Volume:** So that the remote user's voice is heard clearly by you while using headset.
  - **Speaker Transmit Volume:** So that your voice is heard clearly by the remote user while using speaker.
  - **Speaker Receive Volume:** So that the remote user's voice is heard clearly by you while using speaker.
  - **Ring Volume:** So that you can hear clearly the ringing cadence played by the phone.
- Feature can be configured from Web and Menu options.

### How to use it?

- **To adjust handset transmit volume:**
  - Go to "Config Phone Menu", select the option 'Volume Settings'
  - Select option 'Handset Transmit Volume'
  - Enter any value between 0 to 7 to get desired level. (you can confirm the effect by asking the remote user).
- **To adjust handset receive volume:**
  - Go to "Config Phone Menu", select the option "Volume Settings".
  - Select option "Handset Receive Volume"
  - Enter any value between 0 to 7 to get desired volume level.
- Similarly use the feature for headset and speaker options.

### How it works?

- You can adjust volume settings while in speech or when idle. If in speech, the speech will get effect of increase/decrease. If set in Idle state of phone, you can feel the level by picking up the

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- handset or turning ON the speaker or headset.
  - The Phone supports levels of Volume from Level 0 to Level 7. (Level0 being lowest and Level7 being highest).
  - You can feel the setting of Ring Volume only when you receive the Incoming Call, when your phone is NOT in Idle condition.
  - But if your phone is in 'Idle' condition, you can feel the setting when RBT (Ring Back Tone) is played.
  - You can set volume during different condition of using the phone as mentioned below:
    - **When using Handset:**
      - Press '←/→' key on the Phone, to increase/decrease Rx gain of speech/call progress tones respectively.
    - **When using Headset:**
      - When you are in speech over headset, your Rx level of the headset can be changed in same way as explained above. This will be required if remote user has not set his Tx audio level properly.
    - **During Ringing:**
      - Follow same method while the ringer is ringing on receipt of Incoming call and listening it.

**Configuration:**

- To configure the feature, Click on Web Page "**System Parameters**".
- To configure from Phone: Navigate Menu > Config Phone Menu > Volume Setting.

**Relevant Topics:**

1. System Parameters 144
2. Menu 100
3. Call Progress Tone Generation 65

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## VoIP Basics

VoIP is the sending of voice signal over Internet Protocol. Calling on VoIP allows user to make phone calls and send faxes over the internet at a negligible cost compared to the normal circuit switched Telephone network. The main components of VoIP are:

- SIP
- Network Servers
- RTP
- NAT
- STUN
- Outbound Proxy
- SIP Call Flow
- Important Voice Parameters: Jitter Buffer, Voice Compression and QoS.

### **SIP:**

- The signaling protocol which is used to establish an IP call is called SIP Session Initiation Protocol. It is an application layer protocol that handles the setting up, altering and tearing down of voice and multimedia sessions over the internet. Refer RFC 3261 for more information.
- SIP handles telephone calls and can interfere with PSTN network.
- A SIP account uses complete identity called, URI (Uniform Resource Identifier) or SIP Address, like e-mail address and is given by: SIP-Number@SIP service Domain.
- SIP number can use letters or numbers like `ajay@ITSP Provider.com` or `73121@ITSP Provider.com`. A SIP service domain is the domain name in the SIP URL. For example, if SIP address is `73121@VoIP Provider.com`, then 'VoIP Provider.com' is the SIP service domain.
- In general, SIP follows the Client Server Architecture. To support this, there are two main entities user agent and network servers. The peers in a session are called User Agents (UAS). A user agent can function in one of the following roles:
  - User Agent Client (UAC)-A client application that initiates the SIP request.
  - User Agent Server (UAS)-A server application that contacts the

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user when a SIP request is received and that returns a response subject to user's input.

**Network Servers:**

There are three types of **network servers** in a SIP network: Proxy Server, Redirect Server and Registrar Server.

**Proxy Server:** A SIP proxy receives a request, makes a determination about the next server to send it to, and forwards the request, possibly after modifying some of the header fields. As such, SIP requests can traverse many servers on their way from UAC to UAS. Responses to a request always travel along the same set of servers the request followed, but in reverse order.

**Redirect Server:** The redirect server does not forward requests to the next server. Instead it sends a redirect response back to the client containing the address of the next server to contact.

**SIP Registrar:** SIP Registrar is an entity where SIP users can get themselves registered. Registrar imparts mobility to the SIP users. A SIP user can register himself with a registrar. If the user changes his location, he has to register again with the registrar stating his latest contact information. Whenever a call is to be delivered to that user, Registrar can provide the information about the location where the user was active recently.

**RTP:**

After a VoIP call is established using SIP, the RTP (Real Time Protocol) is used to handle voice data transfer in the packets form. Refer RFC 1889 for more details.

**NAT:**

NAT (Network Address Translation-RFC 1631), is the translation of IP address of a host in a packet. For example, the source address of an outgoing packet, used within one network is changed to a different IP address known within another network.

- One network is designated the inside network and the other is the outside. Typically, a company maps its local inside network addresses to one or more global outside IP addresses and unmaps

the global IP addresses on incoming packets back into local IP addresses. This helps ensure security since each outgoing or incoming request must go through a translation process that also offers the opportunity to qualify or authenticate the request or match it to a previous request.

- NAT also conserves on the number of global IP addresses that a company needs and it lets the company use a single IP address in its communication with the world.

### **STUN: (RFC 3489)**

#### **Note:**

STUN is used if there is a NAT router between SETU VP236 and SIP Server (from ITSP).

- STUN stands for Simple Traversal of UDP over NAT. It is a protocol, which enables an SETU VP236 to detect the presence and type of NAT behind which the phone is placed. An SETU VP236 that supports STUN can intelligently modify the private IP address and port in its SIP/SDP message by using the NAT mapped public IP address and port through a series of STUN queries against a STUN server located on the public Internet.
- This will allow SIP signaling and RTP media to successfully traverse a NAT without requiring any configuration changes on the NAT. STUN presents a working solution for most NATs that are not symmetric NAT, e.g., most of the SOHO routers have non-symmetric NAT and in this case, it is OK to use STUN. However, STUN does NOT work with symmetric NAT and if your routers have built-in symmetric NAT, do not use STUN.
- A STUN server can help facilitate traversing through most NATs, except for symmetric NATs.

### **Outbound Proxy**

VoIP service provider can host a SIP outbound proxy server to handle all of the SETU VP236's VoIP traffic. User has to enable this feature if VoIP service provider has a SIP outbound server to handle voice calls. This allows the SETU VP236 to work with any type of NAT router and eliminates the need of STUN.

### **SIP Call flow:**

A simple example of how SIP requests and responses are sent will

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help understanding SIP and RTP.

When A calls B, the steps for the set up and tear down of a SIP call is explained as below:

A		B
1. INVITE		
		2. Ringing
		3. OK
4. ACK		
	5. Voice Data-RTP	
6. BYE		
		7. OK

1. A sends SIP INVITE request to B. This message is an invitation to B to participate in a SIP telephone call.
2. B sends a response that it is ringing.
3. A answers the call. B responds it by 'OK'.
4. B also answers the call and A sends acknowledgment by sending 'ACK' message.
5. Now, voice session starts between A and B in form of RTP (Real Time Protocol).
6. After the talk, A hangs up and sends a BYE request.
7. Finally, B sends the reply by sending 'OK' response and the call is terminated.

### **Important Voice Parameters:**

#### **Jitter Buffer:**

- The clarity of voice heard from the called party is also distorted due to non-uniform delay introduced by the Telephony network known as 'Jitter'.
- In order to reduce the effect of 'jitter', some sort of 'filters' are used in the voice-channel called 'Jitter Buffer'. The SETU VP236 has built-in adaptive buffer that helps to smooth out variations in delay (jitter) for voice traffic. This will ensure good voice quality. It is not programmable.

### **Voice Compression:**

- Voice compression is a technique to reduce the band width of voice channel, using suitable codec or Coder-Decoder. The codec converts voice signals from analog form to digital signals and vice versa. It uses the compression alongwith, CELP.
- Compression-decompression is a voice compression-decompression algorithm that defines the rate of speech compression, quality of decompressed speech and processing power requirements. It is called companding.
- Codec of suitable bandwidth is selected during the speech reception or transmission, to minimize the digital voice channel bandwidth.

### **QoS:**

Quality of Service, QoS is the capability of a network to provide better service to selected network traffic over various technologies, including Ethernet and 802.1 networks, and IP-routed networks. SETU VP236 supports type of Service Tagging and Differentiated Services Tagging as mentioned below: (This allows SETU VP236 to tag voice frames so they can be prioritized over the network).

1. Type of Service (ToS)
2. DiffServe

### **ToS:**

- Network traffic can be classified by setting the ToS values at the data source (at the SETU VP236) so that a server can decide best method of delivery, such as the least cost, fastest route, etc.
- This parameter allows you to configure Type of Service (ToS) bits by specifying the **precedence** and **delay** of audio and signaling IP packets.

### **DiffServe:**

DiffServe is differentiated Services, which defines a Class of Service model that makes packets so that they receive per hope treatment at DiffServe-compliant network services along the route, based on the application types and traffic flow. Packets are marked with DiffServe Code Point (DSCP) indicating the level of service desired.

**Relevant Topic:**

1. Glossary 179

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## Web Jeeves

### Introduction

- As, the set up of SOHO and Enterprise is normally equipped with LAN and Router on the Internet line, it is very convenient if the device has the capability to be programmed via HTML-based interface. The Matrix SETU VP236 has this provision.
- Matrix SETU VP236 needs Web browser IE6 with SP2 (Service Pack) patch or Mozilla Fire Fox installed on the PC. Description of following features related to web will help in programming the system.

### IP Address

- An IP address (Internet Protocol address) is a unique number, similar in concept to a telephone number, used by network devices (computers, time-servers, some telephones).
- It is attached to a network to refer to each other when sending information through a LAN or a WAN or the Internet for example.
- An example of IP address is 192.168.001.151. Converting a number address to a more human-readable form called a domain address is done via the Domain Name System.
- The Internet Protocol (IP) knows each logical host interface by a number, the so-called IP address. On any given network, this number must be unique among all the host interfaces that communicate through this network.
- Users of the Internet are sometimes given a host name in addition to their numerical IP address by their Internet service provider.
- For all programs that utilize the TCP/IP protocol, the sender IP address and destination IP address are required in order to establish communication and send data.

### Sub Net

- It is a mechanism that is used to split a network into a number of smaller sub networks. It can be used to reduce traffic on each sub network by confining traffic to only the sub networks for which it is intended, thereby eliminating issues of associated congestion on other sub networks and reducing congestion in the network as a whole.

- Makes entire network more manageable.
- Each sub network functions as though it was an independent network, keeping local traffic local, and forwarding traffic to another sub network only if the address of the data is external to the sub network. Such decisions are made on the basis of routing-tables contained within the various routers, with each table comprising an IP address table.
- Subnet is a portion of the network, which may be a physically independent network, which shares a network address with other portions of the network and is distinguished by a subnet number.
- The first octet of the IP address shows the network address, the second one shows the subnet number and the last two shows the host part.

### **Default Gateway**

- A default gateway is a node on a computer network that serves as an access point to another network.
- In enterprises, the gateway is the computer that routes the traffic from a workstation to the outside network that is serving the Web pages.
- In homes, the gateway is the ITSP that connects the user to the internet.
- A default gateway is used by a host when an IP packet's destination address belongs to someplace outside the local subnet.
- The default gateway address is usually an interface belonging to the LAN's border router.

### **Domain Name System (DNS)**

- DNS is a system that stores information about **host names** and **domain names** in a type of distributed database on networks, such as the Internet.
- Of the many types of information that can be stored, most importantly it provides a physical location (IP address) for each domain name, and lists the mail exchange servers accepting e-mail for each domain.
- The DNS provides a vital service on the Internet as it allows the transmission of technical information in a user friendly way.
- While computers and network hardware work with IP addresses to perform tasks such as addressing and routing, humans generally

find it easier to work with hostnames and domain names (such as www.example.com) in URLs and e-mail addresses.

- The DNS therefore mediates between the needs and preferences of humans and of software.

### Dynamic Host Configuration Protocol (DHCP)

- The DHCP is a client-server networking protocol.
- A DHCP server provides configuration parameters specific to the DHCP client host requesting, generally, information required by the host to participate on the Internet network.
- DHCP also provides a mechanism for allocation of IP addresses to hosts.

### How to configure SETU VP236?

To start basic VoIP configuration first enter the default LAN IP address of the SETU VP236, obtained from SE in the 'Address' bar of the Web-browser screen as explained below:

- Launch the Web browser as explained below:
- Double Click on **Internet Explorer** icon.

**Or**

Click on **Start → Programs → Internet Explorer**

- Enter IP Address at Address Bar after confirming from SE. Let us say IP Address 192.168.1.200 is entered at Address Bar on opening page of **Microsoft Internet Explorer**.
- SETU VP236 displays Welcome page with a field for entering 'Password'.

Parameter	Value/Default	Description
Password	Up to '4' digits (0-9), 1234	Enter the default password '1234' to access the system

- Click **Login** after entering valid password.
- If the user enters a wrong password then system will pop up a window which will prompt him as, "Password is invalid, please relogin" If user clicks on "**log In**" enters correct password and press 'Enter', he will get 'Welcome' Page.
- To log out, click on "**Logout**". You will find a thanks page asking for Login Password.
- Enter the password if you want to work with SETU VP236.

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## MATRIX SETU VP236

### Welcome

User Address	Login expires if idle for
xxx.xxx.xxx.xxx	10 Minutes

**xxx.xxx.xxx.xxx** : Address of the computer from where the user has logged in.

- **Login Expiry Time:** This is fixed (maximum) time for which user can keep a page without programming any parameter in it. If this time is exceeded, he cannot submit current page. He has to login again.
- All the above fields are non-editable. The user can now go to any of the screens by clicking on the links provided at the left of the screen.
- Main Menu is displayed on the left side of Welcome Page (without Password field).
- The 'Main Menu List' shows all the important parameters that are required to be programmed, to make the SETU VP236 work as expected.
- Click .....from the 'Main Menu'.

### Common screen command buttons: (General Note for all the screens)

Note that at the end of the every screen, options are provided for really applying these changes to the SETU VP236. This is conveyed by 'Submit' and 'Default All':

Button	Meaning
Submit	Click, to save changes to VP236
Default All	Click, to set all parameter on the screen to factory default

Links for following features are visible on the Welcome Page:

- Auto Configuration
- Call Logs
- Change Password
- Daylight Saving Time Adjustments
- Default IP Phone

## Matrix

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- Keys Programming
- Network Parameters
- OG Call Route Selection
- Peer-to-Peer Configuration
- Peer-to-Peer Table
- Phone Book
- SIP Parameters
- Software Upgrade
- Status
- System Parameters
- Time Setting
- User Setting

### Screen Summary:

Link	Functions
Auto configuration	This is configure the Phone with specific MAC Address number by the ITSP
Call Logs	This shows record of various types of calls like, IC, OG, Conference, Rejected calls etc
Change Password	This screen is used to change Keypad unlock and Config Phone Passwords for security reasons
Daylight Saving Time Adjustment	This is to program new start and end times as per the country's DST rule.
Default IP Phone	This is to set again all parameters to factory set values.
Keys Programming	This page is used to assign important feature with a specific key with or without LED
Network Parameters	This is to configure Network port parameters
OG Call Route selection	This is to select the SIP1 or SIP2 Trunk for call routing
Peer to Peer Configuration	This is for programming the configuration parameters.
Peer to Peer Call Table	This is for programming Peer to Peer call Table with Number and Address
Phone Book	This is for programming phone-book entries

SIP Parameters	This screen is used for configuring SIP settings like IP Address, Server Address etc for SIP1, SIP2
Software Upgrade	This is to upgrade the software for any new or current features.
Status	This is a brief status of important parameters like Network port parameters etc
System Parameters	This for programming parameters like, CPTG, Ring, volume setting etc
Time Setting	Use this screen to configure system's time to synchronies with the time server
User Setting	This is to set some features like, Call Forward, Auto Answer etc.

### How to make changes in feature?

In order to make changes, click on any corresponding feature, from the list of the Web page and then refer related topic.

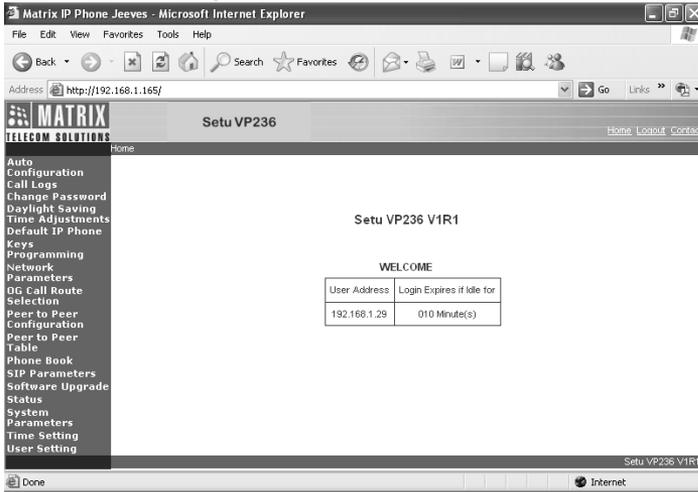
### Example:

To use Phone Book feature for speed dialing.

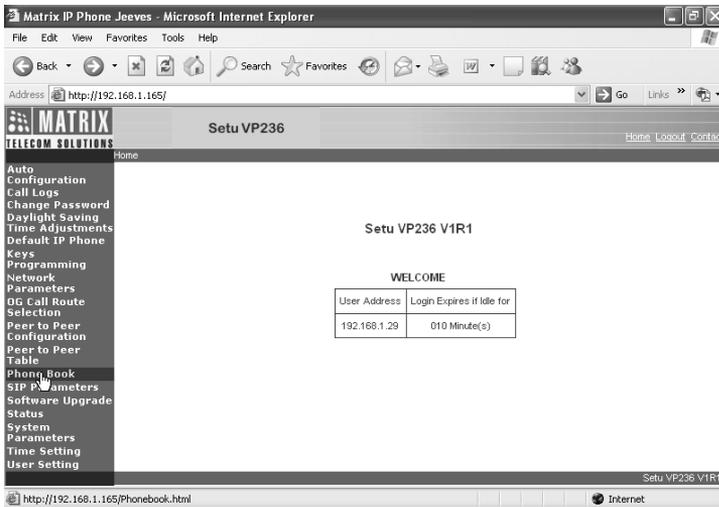
- After entering suitable IP Address in address field the Welcome Page will look as:



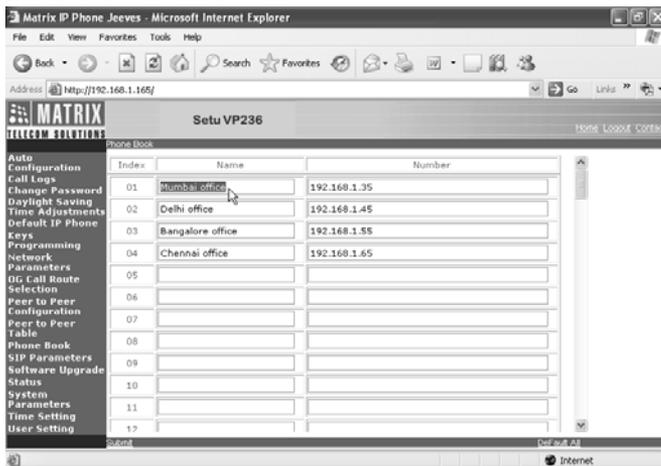
- After log-in, the following page “Login Status Page’ displays the time after which log-in will expire as shown below:



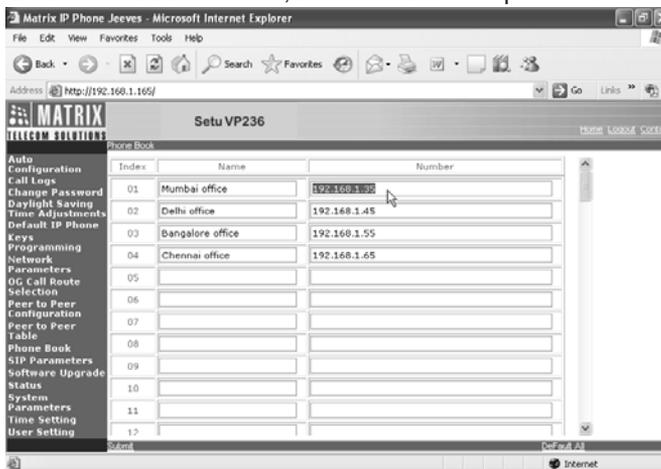
- Click on “Phone Book’ as shown:



- Enter in the first Index, the name for which you want to dial a number as shown below:



- Enter for the same index, the number of the person to be called:



- Similarly enter other Index and click on “Submit”.
- The phone book can be used now.
- Other features can be configured in the same way.

**Relevant Topics:**

1. Change Password 73
2. Know Your SETU VP236 17

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## **Section 3: Appendices**



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## Appendix A: Glossary

### **CELP**

It is the Code Excited Linear Prediction compression. It is the compression algorithm used in low bit-rate voice encoding. Used in ITU-T Recommendations G.728, G.729, G.723.

### **DHCP**

It is the Dynamic Host Configuration Protocol. It provides a mechanism for allocating IP addresses dynamically so that addresses can be reused when hosts no longer need them.

### **DNS**

It is the Domain Name System. System used on the Internet for translating names of network nodes into addresses.

### **Firewall**

Router or access server or several routers or access servers, designated as a buffer between any connected public networks and a private network. A firewall router uses access lists and other methods to ensure the security of the private network.

### **TFTP**

File Transfer Protocol - A protocol used to transfer files over a TCP/IP network.

### **G.723**

It is an ITU standard for speech codecs that uses the ADPCM method and provides toll quality audio of 20 and 40 Kbps.

### **G.729**

Describes CELP compression where voice is coded into 8-kbps streams. There are two variations of this standard (G.729 and G.729 Annex A) that differ mainly in computational complexity; both provide speech quality similar to 32-kbps ADPCM. It is described in the ITU-T standard in its G-series recommendations.

## **IP**

It is the 'Internet Protocol', Network layer protocol in the TCP/IP stack offering a connectionless internetwork-service. IP provides features for addressing, type-of-service specification, fragmentation and reassembly and security. It is defined in RFC 791.

## **MAC**

Media Access Control Address-The unique address that a manufacturer assigns to each networking device.

- MAC is Media Access Control address in the form of 48-bit number, which is unique to the LAN NIC (network interface card).
- It is programmed into the card at the time of manufacture. IEEE registration authority administers MAC address scheme for all LANs which conform to IEEE, 802stds. Including both Ethernet and token ring.
- Consists of two parts: 24-bit company id (manufacturer ID0 and 24-bit Extension id (board id)
- Destination and source MAC names are contained in the header of the LAN packet and are used by various devices like hubs, bridges.
- A VoIP Service Provider will typically have its subscribers register with the Service Provider's VoIP service server before starting a subscriber's service. Often, an ITSP will require the registration of the MAC addresses of any devices directly connected to their network.

## **SDP**

It is the Session Definition Protocol, an IETF protocol for the definition of Multimedia Services. SDP messages can be part of SGCP and MGCP messages.

## **TCP**

It is the Transmission Control Protocol, a connection-oriented transport layer protocol that provides reliable full-duplex data transmission. TCP is part of the TCP/IP protocol stack.

## **UAS**

It is the User agent server (or user agent), a server application that contacts the user when a SIP request is received and then returns a

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response on behalf of the user. The response accepts, rejects or redirects the request.

**UDP**

User Datagram Protocol-A network protocol for transmitting data that does not require acknowledgment from the recipient of the data that is sent.

**VoIP**

It is the Voice over IP, the capability to carry normal telephony-style voice over an IP-based Internet with POTS-like functionality, reliability, and voice quality. VoIP enables a router to carry voice traffic (for example, telephone calls and faxes) over an IP network. In VoIP, the DSP segments the voice signal into frames, which then are coupled in groups of two and stored in voice packets.

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## Appendix B: Product Specifications

### Configuration and Capacity:

- Headset Port : 1 (Audio Jack), 1 (MIC)
- Handset Port : 1 (RJ11)
- Network Port : 1 (RJ45) (PoE Supported)
- PC Port : 1 (RJ45)
- DC Power : 1 (Power Jack)

### VoIP and Networking Protocol:

- SIP (RFC 3261, Ver.2)
- IPv4-Internet Protocol Version 4 (RFC791)
- ARP-Address Resolution Protocol (RFC 0826)
- ICMP-Internet Control Message Protocol (RFC0792)
- TCP-Transmission Control Protocol (RFC793)
- UDP-User Datagram Protocol (RFC768)
- RTP-Real Time Protocol (RFC1889)
- RTCP-Real Time Control Protocol (RFC1889)
- SNTP-Simple Network Time Protocol
- PPP-Point-to-Point Protocol (RFC 1661)
- DNS Protocol (RFC 2136)
- Outbound Proxy
- STUN
- NATKeep Alive
- VLAN Tag

### Voice Functionalities:

- Maximum two SIP Accounts can be used simultaneously
- Maximum four Call Appearances are supported
- DTMF Options: Inband, RTP (RFC2833), SIP (INFO)
- Call Progress Tone Generation: Select as per the country
- QoS: RTP QoS and SIP QoS using DiffServ and ToS (Precedence)
- Full Duplex Audio
- Echo Cancellation
- Speech Volume Setting-Transmit and Receive

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**Voice Codecs:** PCM (a-Law and u-law), G.729, G.723-5.3, G.723-6.3, G.726-16, G.726-24, G.722, G.726-32, G-726-40.

**Telephony features:**

- Voice Calls using SIP proxy and Voice calls without using SIP proxy (Peer-to-Peer Calling)
- Call Waiting and Cancel Call Waiting
- Caller ID Blocking
- Call Forwarding
- 3-Party Conference
- Call Transfer
- Caller ID: Display
- Speed Dialing with Phone Book Feature
- Voice Mail Account: Two
- LCD Displays: Size : 2 line/6 line, with 24 characters and tilt facility. Brightness, Contrast : Programmable

**Time Settings:**

- Synchronizing with specific Time Server

**Provisioning, Administration and Maintenance:**

- Auto Configuration
- Configuration Using Web Page, Phone-'Menu' and Programmable keys.
- LED Indications:
  - Ringer LED
  - Speaker LED
  - Programmable keys with LED
- Software Upgrade using FTP

**Security:**

- Password Protected Administrator.

**Power Supply:**

External Adaptor : 5.0VDC@2.5Amp. or through Power on Ethernet.

Power Consumption : 10W (Typical)

## Matrix

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### **Environmental:**

Operating Temperature : -10°C to 50°C (140° to 122°F)

Operating Humidity : 5-95% RH, Non-Condensing

Storage Temperature : -40°C to 85°C (-40°F to 185°F)

Storage Humidity : 0-95% RH, Non-Condensing

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