

# KTA2110 REFERCE GUIDE

## 1 Functions

### 1.1 Making Calls Through PSTN Port

**Notice:** In order to use this function you must connect the 'PSTN' port to your existing wall telephone jack.

If the current working status of KTA is PSTN, you can make a call directly, but if in VoIP status, please pick up the handset then dial '0\*' and then dial destination number. Your call will be placed on your regular PSTN line using your phone company's dialing rules.

### 1.2 Receiving Calls Through PSTN Port

**! Note:** If there is a call incoming on the PSTN port, the called telephone connected to Phone port will not ring immediately, the KTA will have to detect the ringing before switching it to the phone, this could take 3-4 seconds. VoIP calls will ring immediately.

### 1.3 Work as router/ bridge

This terminal can work as a family router/ bridge, please select the working status of the network as you want.

**! Note:** We recommend you use it as a secondary router.

### 1.4 Volume Adjust

On WEB page programming: Phone Setting ◊ Volume Settings.

**! Note: This feature can be limited by the phone device.**

### 1.5 Dial Plan

Dial plan is for:

1. Set auto dial out rule without delay and press # (dial now).
2. Change input number before send out and select a sip account to dial out.

How to set it: Web page programming: Phone Settings ◊ Dial Plan Settings:

Dial Plan

You could set the dial plan in this page.

Dial now:

Name	Digit for matching	Operation	Digit for separating	Routing rule
Digit Map1	.+1*	drop number		IP account1
Digit Map2	.+2*	drop number		IP account2
Digit Map3	.+3*	drop number		IP account3
Digit Map4		drop number		current sip account
Digit Map5		drop number		current sip account
Digit Map6		drop number		current sip account
Digit Map7		drop number		current sip account
Digit Map8		drop number		current sip account

Auto Dial Time:  (3-9 sec)

Use # as send key:  Yes  No

Use \* for 1st dialing:  Yes  No

**PART A**      **PART B**      **PART C**      **PART D**

In 'dial now', set a number here, if matched, the number will be auto dial out without delay and press #. E.g., set xxx+51 in 'dial now' means any 3 digital number or number 51 will be auto send out without delay and press #.

To change the input number:

**PART A:** can set a number use for matching. If device find the number is the same with the prefix of your input number, it can use **PART B** and **PART C** to change your number and then dial out.

**PART B** has the selection of:

Drop number: delete the number in **PART A**

Replace number by: use **PART C** to take place of the number in **PART A**

Disable

In **PART D:** select a sip account to send out this call, if select 'current sip account', means use your current using line to send out. (press 1\*, 2\*, 3\* can select 1st~3rd sip lines).

You can use following special characters in **Part A** or **Dial now:**

**+** : You can use it in the left blank to describe more than 1 prefix rule in **Place B**. Example: 88+77 in **Place B** means the number which has 88 or 77 as the prefix can be changed.

**x** : It can use to act as a number from 0~9. in **Place B** and **Dial now**. Example 1: 87xxxxx in **Dial now** means a number 87xxxxx will be send out automatically. Example

2: 5xxx in Place B means a number 5xxx can be changed.

Dial Now	If the number you dialed is the same with the number which you entered in the left blank, it will be automatically send out  <b>! Note: 0 cannot be set as the first character.</b>
Auto Dial Time	Use for auto dial out if your haven't any new input within this period
Use # as send key	If select it, press # to dial out, if not select, you should wait for the auto dial time to dial out
Use * for IP Dialing	To let * as .(DOT), when you want to dial a IP you need to select it

## 2 Service Features

**! Important Note:** These features need your VoIP service provider's support.

### 2.1 PSTN and VoIP Caller Number Pass Through

KTA can pass the caller ID number sent from the PSTN port or VoIP to the regular phone connected to its PHONE port. It now only supports FSK way to pass number, you can see the caller ID number on the LCD of the phone (if available).

Please set the related parameters for the caller ID display type in WEB page: Phone Setting◇ Caller ID.

### 2.2 VoIP Call Forwarding to VoIP

Please set this function in web page: Phone Setting◇ Call Forward.

Unconditional Call Forward: The incoming call will always forward to the destination number.

No Answer Call Forward: The incoming call will forward to the destination number if there is no answer within the time you have set.

Busy Call Forwarding: The incoming call will forward to the destination number if there is a call.

### 2.3 Call Hold / Retrieve

Operation: While on a call, depress hookswitch quickly (or press 'Flash' button on the phone) then dial the new number. To retrieve the previous call just presses hookswitch quickly (or press 'Flash' key on the phone.)

### 1.6 Hot line

In WEB page: Phone Setting◇ Hot line Settings.

The device will send out the number you set after you picked up the handset or use hand free model without any incoming call.

### 1.7 Wake up alarm

In WEB page: Phone Setting◇ Alarm.

The device will auto ring when reached the time you set here.

### 2.4 Second line and switch line

When the call is on hold, dial number and # to make a second line call. Press Flash key or presses hookswitch quickly can switch line to the other one.

While you are on 2 calls, to cancel one call: you can hang up this call and will hear a ring tone, pick up to make call with another line.

### 2.5 Call Transfer

- Blind Transfer

Operation: While on a call, press 'Flash' button or hookswitch to hold this call. Press #510# and then input the destination number and '#', then the call will be transferred to the destination number. Please wait till you heard the busy tone and then you can hang up.

- Attendant Transfer

Operation: While on a call, press 'Flash' button or hookswitch to hold this call. Press #511# and then input the destination number and '#', after someone picks up the call, hung up and the call will be transferred.

### 2.6 3-way calling

Operation: While on a call, press 'Flash' button or hookswitch to hold this call. Press #512# and then dial the 3<sup>rd</sup> party's number and '#', after 3<sup>rd</sup> party answers the call, you can press 'Flash' button or hookswitch to begin a

3-way calling.

## 2.7 Call waiting/Answer the waiting call

When on a call and a new call comes in, you will hear the call waiting tones. Press the hookswitch quickly or press the 'Flash' key on the phone to pick up the call and hold the last call.

## 2.8 Message Waiting Indicator

When there is a unread message on the server, the device will get a signal from server and you will hear the special tone when you pick up the phone.

## 2.9 Multiple lines

You can set 3 different account and server information in the device. When the device stand by, pick up the handset, press 1\*, 2\*, 3\* to switch to the 1st, 2nd and 3rd line, after you press the hookswitch, the setting will take

## 3 Upgrade

**Note: The new version may request you restore default settings, you may lose all the settings in the device especially including account and server information.**

How to use **Restore default settings**: Dial #198# or on web programming: Update ◊ Default Settings.

### 3.1 Local update

Put the new version file to your C:\ of your PC.

A. If the version file is a \*.gz file, use your IE explorer to open the upgrade web page: **http://x.x.x.x(your device IP)**, use normal user account login and go to the menu: **Update ◊ New Firmware**. In 'Method', select Local PC, in 'Codec Type', Select Risc. And then select the new firmware, then click 'Update' button.

Please keep the power on and do not change WEB page till it reboot automatically. Then upgrade is OK.

B. If the version file is a \*.rom file, use your IE explorer to open the upgrade web page: **http://x.x.x.x(your device IP)**, use root account login then change the URL to: **http://x.x.x.x(your device IP)/update.htm**, select Update type as 'ALL ROM' and the new version file to update.

Please keep the power on and do not change WEB page till it reboot automatically. After that, you should **Restore default settings**.

### 3.2 Remote upgrade

If this terminal has been pre-set the upgrade server address, it will automatically check new version from the server. If there is a new version, it will send a special voice when the user pick up the handset or use hands free.

User can dial #160# to begin remote upgrade.

After it restarts, the phone will check new version again.

**Warning: Do not let your device lose power during the upgrading process, or it may be damaged!**

## 4 Safety Warning

**Safety Warning: Please do not place this product near fire or high temperatures. Avoid any heavy impacts, and do not leave the product in rainy or highly humid environments! Do not touch the exposed metal component of the device.**

effective.

In the dial plan settings, if there are 1.+1\*, 2.+2\*, 3.+3\* be defined to use sip account1,2,3, that means you can directly dial 1\*/2\*/3\* + number to use 1<sup>st</sup>/2<sup>nd</sup>/3<sup>rd</sup> sip account to make out calls.

## 2.10 VoIP: Do not disturb

You can set in the web page: Phone Setting◊ Do Not Disturb Settings. If you set it on, all incoming calls will be cancelled. You can set the block duration of the terminal, if the end time is earlier than the begin time, it means the DO NOT DISTURB duration will from beginning time to next day's end time.

## 2.11 Call Forward Indicator

If you are using call forward function, you will get a indicating voice when you pick up.