

**Q: What is Dial Plan used for ? How to configure it?**

**A:** Dial plan is used to convert the number dialed to the number actually sent based on “dial number”, “ddd prefix”, “idd prefix”, “ddd code”, “idd code”

these parameters. Parameter “use dial plan” contains 5 options:

1. “disable”

Number sent is the same as number dialed.

2. “enable”

If the number dialed is prefixed by “idd prefix”, the number sent will be the rest part of the number dialed dropping off the “idd prefix”.

Otherwise, if the number dialed is prefixed by “ddd prefix”, the number sent will be “idd code” + the rest part of the number dialed dropping off the “ddd prefix”.

Otherwise, the number sent will be “idd code” + “ddd code” + the number dialed.

For example: set “ddd code” = 10, “idd code”=86, “idd prefix”=00, “ddd prefix”=0.

If the number dialed is 00-1-4089821818, the number sent will be 1-4089821818.

If the number dialed is 021-82378008, the number sent will be 8621-82378008.

If the number dialed is 82378008, the number sent will be 861082378008.

3. “dial number”

Convert the number dialed to a new number according to “enable” dial plan, then prefix “dial number” to this new number. This new number is the number sent.

4. “prefix”

Effective only when “dial number” is not empty. This serves for those calling cards.

The number sent is just “dial number”. During the call connection, program will automatically send out a number made up with [“account” + “pin” + number converted by “enable” dial plan + ‘#’].

5. “hotline”

The number sent is “dial number”. Whenever the handset is lifted up, this number will be automatically dialed out.

**Q : What is “inner line”?**

**A :** Parameters “inner line” and “inner line prefix” help to distinguish whether the number dialed is an inner line number. “inner line” contains 3 options:

1. “disable”

No distinction among numbers. The number sent is the number converted according to parameter “use dial plan”.

2. “enable”

If the number dialed is prefixed by “inner line prefix”, the number sent is just the number dialed because this number is an inner line number.

Otherwise, the number sent is the number converted according to parameter “use dial plan”.

### 3. “omit prefix”

The only difference between 2 “enable” is that if the number dialed is prefixed by “inner line prefix”, the number sent is the number dialed dropping off this prefix.

### **Q : What is “digit map”? How to configure it?**

**A :** Digit map is a set of rules to determine when the user has finished dialing. With digit map, users don't have to press ‘#’ key or “call” key after dialing. Digit map comes from MGCP protocol. In MGCP, Call Agent will send a digit map to endpoint, so “use digit map” is always selected and users don't have to store a digit map file in the endpoint beforehand. With other protocols, a digit map file has to be stored in the phone beforehand and can be upgraded by pressing the [update digitmap] button or click the [upgrade firmware] at the bottom of the settings web page. The default digit map file used in our program is <http://www.aredfox.com/download/documents/stdmap.txt>. For detailed information, please refer to RFC3435 section 2.1.2. Here is a brief description of some items in stdmap.txt:

X represents any number between 0 and 9.

T represents a period of time after the user pressed the last number. Default value is 5 seconds.

[0-5]: Any number between 0 and 5. That is 0, 1, 2, 3, 4, 5.

13xxxxxxxx: Any 11 digits number starting with 13.

02[0-57-9]xxxxxxxx : Any number starting with 02, the second number between 0 and 5, the third number is between 7 and 9, and total length is 12.

\*x.T : Starting with \* key, and users have waited T seconds after pressing the last digits.

x.T : Any digit number and users have waited T seconds after pressing the last digits.

If the number dialed matches some item in the digit map file, or it doesn't match with any item, this number will be sent out immediately. Not like "use dial plan", "use digit map" won't change the number dialed, the number sent is the same as the number dialed.

"use digit map" can be combined with "use dial plan" and "inner line". First using digit map to determine when the user finished dialing, then convert this number to the number actually sent according to "use dial plan" and "inner line".

**Q : "dual mode" and "dual mode prefix"**

**A :** Only 1S1O gateway based on PA168V reference design needs to set "dual mode" and "dual mode prefix".

If "dual mode" is set to "pstn first", users can call any PSTN number directly; If users want to call an IP phone, please pat the hook once or press the flash key

and wait until the dial tone is heard, now the phone is ready to make an IP call. If “dual mode” is set to “IP first”, users can make an IP call directly; If users want to call a PSTN phone, please dial the number set in “dual mode prefix” field first and wait until the dial tone is heard, now the phone is ready to make a PSTN call.

In both cases, the phone will ring no matter the incoming call is from PSTN network or IP network. Users can pick up the handset to accept the call.

If “dual mode” is set to “disable”, this 1S1O gateway is reduced to a 1S port gateway.

If powered off, the gateway will be automatically switched to the PSTN network so that users can make PSTN calls as usual.

**Q: Why does it take so long before the phone logs in to the server? How to speed it up?**

**A:** Before registration, phone will first visit FTP server and check if the server has got new firmware or settings to upgrade, then it will visit the time server on the Internet to get current date and time. If the FTP server’s address is a domain name, program will visit DNS to resolve this domain name into IP address. Besides, if local IP address is allocated by DHCP, we will check if the allocated IP address conflicts with other devices. All these work will cost some time to complete, which causes the delay of registration. To speed it up, try the following two methods:

1. set “upgrade type” to “disable”, then phone will not check new upgrades from the FTP server.
2. set “sntp ip” to “255.255.255.255” , then phone will not visit timer server to get date and time. If one must know the current date and time, please set “sntp ip” to the IP address of a local time server which has fast access.

**Q: How to recover the phone if it crashed during upgrade?**

**A:** If user upgrades the phone with a firmware that doesn't match the hardware, the phone will crash and won't reboot. There are a lot of different reference designs based on PA1688, so one must make clear what the hardware type is and find the corresponding firmware before doing upgrade. Never set “debug” to “no check”, because this way program won't check if the firmware matches the hardware or not, which is very dangerous. If user really doesn't know the hardware type, please send email to [support@aredfox.com](mailto:support@aredfox.com) to get help. Once upgrade failed, follow the below steps to do safe-recovery:

1. Press and hold \* while power up, until you see \* begin to display on your LCD.
2. Power off and do 1 again, after the second time power up, our standard design IP phone's default IP address is 192.168.1.100. There are exceptions, like with Unisen ip8008, the default is 192.168.0.100, and with Vida iPH-E00 serial phones has default IP address of 192.168.1.10.
3. Put the ip phone and a Windows PC on the same LAN.

4. Run palmtool.exe on PC. You can try palmtool "Start Debug" button, a debug window will pop up, and if you press key on the IP phone, the key and the IP phone's IP address will be displayed on the debug window
5. If your PC's address is not 192.168.1.xxx (not on the same LAN of the IP phone), change its TCPIP settings to the same LAN.
6. Put 192.168.1.100 (or other default IP) in the palmtool "IP Address on Chip" field.
7. Find the correct upgrade file, and use palmtool "Update Program" button to update your IP phone.

**Q: How to set TOS field?**

**A:** The 8-bit TOS field specifies how the datagram should be handled and is broken down into five subfields: Let's assume that bit 0 is the most-important-bit and bit 7 is the least-important-bit.

Bit 0-2: Three PRECEDENCE bits specify datagram precedence, with values ranging from 0 (normal precedence) through 7 (network control), allowing senders to indicate the importance of each datagram.

Bit 3: Also called D bit. When set, the D bit requests low delay.

Bit 4: Also called T bit. When set, the T bit requests high throughput.

Bit 5: Also called R bit. When set, the R bit requests high reliability.

Bit 6-7: Unused

TOS is just a hint to the routing algorithm that helps it choose among various

paths to a destination. An internet does not guarantee the type of transport requested.