

Analog Telephone Adapter



UTA-1001 User Manual

CONTENT

1. UTA-1001 SIP Gateway Features.....	3
1.1 Product Appearance	3
1.2 Backside Illustration	4
1.3 Software	4
1.4 Protocol and standard.....	4
1.5 Interface features.....	5
1.6 Electric requirements.....	5
1.7 Operating requirement.....	5
1.8 Certificate:	5
1.9 Packaging.....	5
1.10 Installations	5
2. Settings 7	
2.1 Home.....	7
2.2 LAN	8
2.2.1 LAN status	8
2.2.2 LAN settings	9
2.2.3 PPPoE setting.....	9
2.3 SIP.....	11
2.3.1 SIP server Configuration	11
2.3.2 Extensions	12
2.3.3 User1 Configuration(User2 is the same as User1).....	13
2.3.4 OOB Signalling	15
2.3.5 ToS/DiffServ	16
2.3.6 Tone.....	17
2.3.7 Ring.....	18
2.3.8 Service Code.....	19
2.4 CODECS.....	20
2.5 System.....	21
2.5.1 Security,Timeout	21
2.5.2 Localization.....	22
2.5.3 Handset	23

2.5.4	<i>SNMP Configuration</i>	24
2.6	Download.....	25
2.6.1	<i>Download</i>	25
2.7	Configuration.....	26
2.7.1	<i>Backup and restore settings</i>	26
2.8	Reset.....	27
3.	Restore to factory default	28
4.	FAQ	29
Q1	What is the default account of UTA-1001?	29
Q2	How to use the IVR function of UTA-1001?.....	29
Q3	How can I know the IP address of UTA-1001?	30
Q4	How to update UTA-1001 firmware?	31
Q5	How to use dial plan?.....	31
Q6	How to use the value add service of UTA-1001?	32
Q7	How to configure UTA-1001?	33
Q8	How to change UTA-1001 LAN port MAC address?.....	33
Q9	Why does my UTA-1001 always drop off from the server?.....	34
Q10	How to use the speed dial function?	34

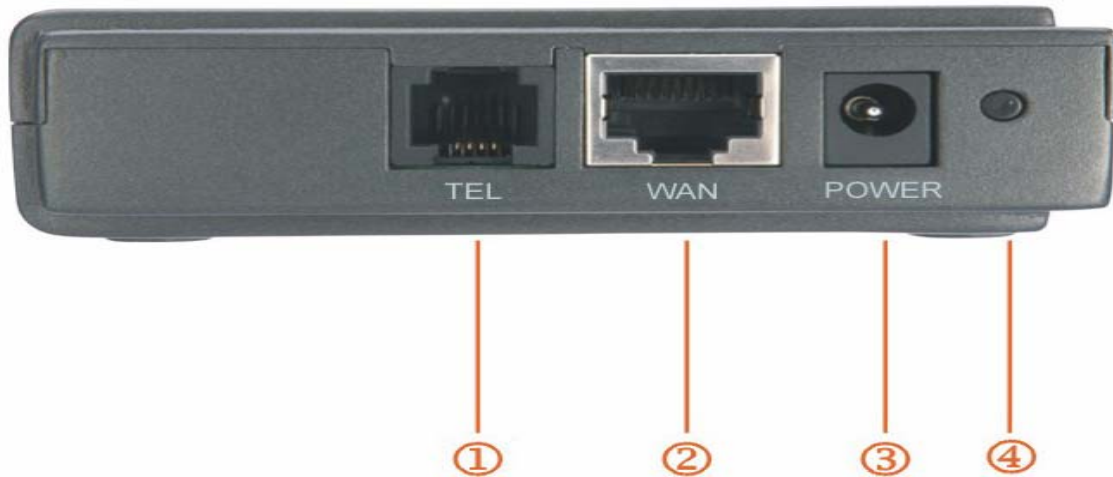
1. UTA-1001 SIP Gateway Features

1.1 Product Appearance



1. TEL: telephone status, the light is led ,when pick up the phone the light off
2. WAN: WAN port status ,when the network is ok the light will flash
3. SYS: Not registered in sip sever ,the light off , Registration lamp blinks registered, light off
4. PWR: power status led.

1.2 Backside Illustration



1. TEL: telephone interface
2. WAN: 10M/100M auto-negotiation, factory default is set to DHCP.
3. POWER: Power Jack, 9~12V,800mA
4. Reset: Please refer to "restore to factory default" for the usage of this button

1.3 Software

- Configured by HTTP web browser
- Support HTTP, TFTP upgrade.
- Support major G.7XX (G711,G729,G723,G726) Codec
- Dynamic voice jitter buffer, CNG (Comfort noise generation),VAD
- G.165 compliant 16ms echo cancellation
- Tone generation and Local DTMF re-generation according with ITU-T
- E.164 dial plan and customized dial rules
- Support T38 FAX
- Support adjustable user password and super password
- IVR (Interactive Voice Response)

1.4 Protocol and standard

- IEEE 802.3 /802.3 u 10 Base T / 100Base TX
- Major G.7XX;
- SIP RFC3261
- TCP/IP: Internet transfer and control protocol
- RTP: Real-time Transport Protocol
- RTCP: Real-time Control Protocol
- VAD/CNG save bandwidth
- TFTP: File Transfer protocol
- HTTP: Hyper Text Transfer protocol
- HTTPS: Secure Hypertext Transfer Protocol

1.5 Interface features

- WAN: 10M/100M auto-negotiation
- LAN: 10M/100M auto-negotiation
- FXS ports:
 - Line Feed Voltage: $\geq 42V$
 - Ring Voltage: $\geq 45V$.
 - Ring Current: $\geq 30mA$

1.6 Electric requirements

- Voltage: 9V ~ 24V
- Power adapter: output DC 12V/450 mA

1.7 Operating requirement

- Operation temperature: 0 to 40° C (32° to 104°F)
- Storage temperature: -30° to 65° C (-22° to 149°F)
- Humidity: 10 to 90% no dew

1.8 Certificate:

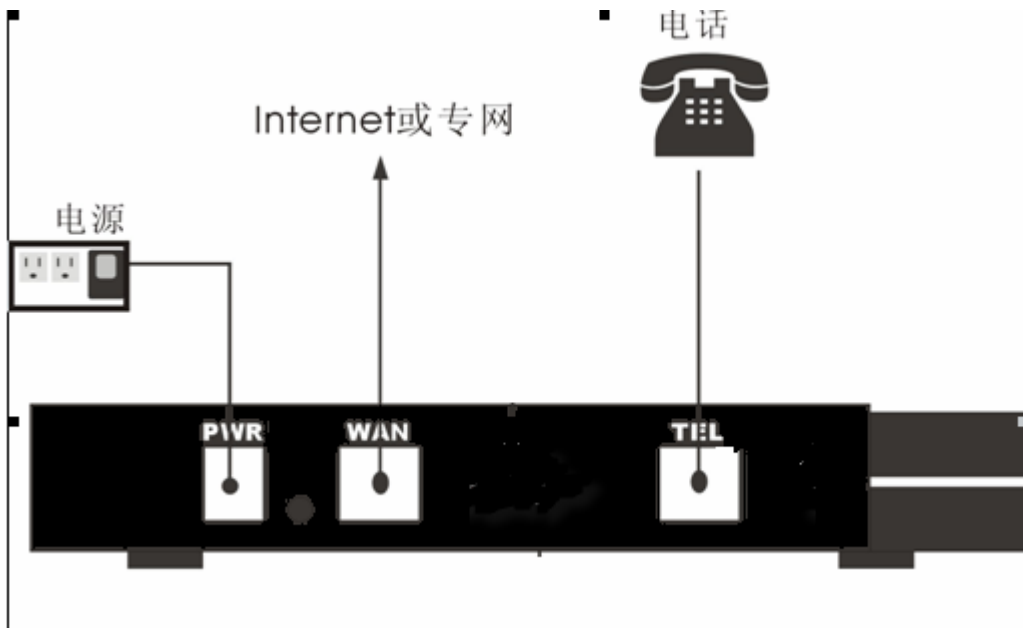
- CE, FCC Part15, RoHS

1.9 Packaging

- **Size:** 22.5cm × 17.0cm × 7.6cm
- **Packing list**
 - ✓ UTA-1001 gateway X 1
 - ✓ Power adapter X 1
 - ✓ CD X 1

1.10 Installations

Connect the UTA-1001 LAN port and you computer with the RJ45 cable, and then change your computer IP to static 192.168.1.xxx or use dynamic obtain IP, type 192.168.1.1 in your IE browser to access UTA-1001 and change its setting.



2. Settings

2.1 Home

Welcome to the VoIP ATA100(1FXS) download and configuration utility.
 Select from the configuration options in the menu on the top.

System Information

System Uptime:	0 days, 0h 0m 10s
NTP time:	NTP Time Not Available
LAN IP Address:	192.168.1.26 (Dynamic)
MAC Address:	00:0d:1a:00:00:01
Serial Number:	
Security:	Password installed
Application Code Version:	VR 4.2A (MSCS A10001) Build-Date: Apr 12 2007
Downloader Code Version:	VR 4.2A (MSCS A10001)

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1. System uptime: system running time
2. NTP time: NTP time and date
3. LAN IP Address: Gateway IP, factory default is 192.168.1.1
4. MAC Address: Gateway MAC address, the same as LAN port MAC address
5. Serial Number: device serial number
6. Security: password information
7. Application Code Version: firmware version
8. Downloader Code Version: downloader version

2.2 LAN

2.2.1 LAN status

LAN Status

LAN Settings

Interface Status

Enabled: Yes
 Protocol: Ethernet
 Interface Status: **Up**
 Link Status: 10M bps, Half Duplex

Network Settings

IP Address: **192.168.1.26**
 MAC Address: 00:0d:1a:00:00:01
 Subnet Mask: 255.255.255.0
 Default Gateway: 192.168.1.1
 Host Name:
 Domain Name:
 Priority Tag: Not set

Update

Interface Status:

1. Enable: Yes indicate LAN id ready for use
2. Protocol: Ethernet
3. Interface status: UP or Down
4. Link Status: link speed mode

Network Settings:

5. IP address: LAN port IP address, factory default is 192.168.0.1
6. MAC Address: LAN port MAC address
7. Subnet Mask: LAN port subnet mask
8. Default Gateway: default gate way IP
9. Domain Name: Domain
10. Priority Tag: Priority Tag value encoded in the Ethernet header in outgoing packets.

2.2.2 LAN settings

LAN Configuration

Use DHCP to obtain LAN configuration

Specify fixed LAN configuration

IP Address:

IP Netmask:

IP Gateway:

Automatically obtain DNS server settings

Manual DNS server settings

IP DNS Server:

IP DNS Server2:

Host Name:

Domain Name:

- 1、 Use DHCP to obtain LAN configuration:use DHCP to obtain ip address for Ethernet
- 2、 Specify fixed LAN configuration: fix ip address
 IP Address: ip address of ethernet
 Subnet Mask: mask
 IP Gateway: gateway
- 3、 Automatically obtain DNS server settings: Auto obtain DNS
- 4、 Manual DNS server settings: setting DNS server by yourself
 IP DNS Server : public DNS Server
 IP DNS Server2: private DNS Server

2.2.3 PPPoE setting

Home **LAN** SIP CODECS System Download Configuration Reset Logout

LAN Status
 LAN Settings
PPPoE

LAN PPPoE Configuration

Enable PPPoE:

Authentication

Username:

Password:

Settings

Idle Timeout: minutes

Service Name:

AC Name:

1. Enable PPPoE: use PPPoE to connect to the internet
2. Username&Password: PPP id and PPP pin from your ISP
3. Echo Timeout: The duration between PPP echo requests sending to server.
4. Echo Count: The number of unanswered PPP echo requests before PPP connection is closed.

2.3 SIP

2.3.1 SIP server Configuration

Server
 Extensions
 Digit Map
 User 1
 OOB Signalling
 ToS/DiffServ
 Tone
 Ring
 Service Code
 Phone Book

SIP Server Configuration

Primary Server Settings
 (Current Server: : 5060 ; Domain:)
 * Address: [] (IP or FQDN)
 * Port: []
 Domain Name: []
 Send Registration Request with Expire Time [1800]
 Outbound Proxy IP: [] (IP or FQDN)
 Outbound Proxy Port: []

Secondary Server Settings
 (Current Server: : 0 ; Domain:)
 * Address: [] (IP or FQDN)
 * Port: []
 Domain Name: []
 Send Registration Request with Expire Time []
 Outbound Proxy IP: [] (IP or FQDN)
 Outbound Proxy Port: []

RTP Port Number Setting(5000~65535) []~[] Transport type Setting UDP

NAT Traversal Settings
 NONE
 UPnP Control Point
 STUN Server IP: [] (IP or FQDN) STUN Server Port: []

1. Primary Server: Primary Server, UTA-1001 will auto switch to Secondary server if primary server is unavailable.
2. Secondary Server: secondary server (back up function)
3. Address: SIP server IP address
4. Port: SIP server port, the well know port is 5060
5. Domain Name: server domain
6. Send Registration Request with Expire Time: Register TTL (unit: seconds). Indicate the register period, if UTA-1001 always log off after some time, please set this time to a lower value.
7. Outbound Proxy IP: Outbound Proxy server IP address
8. Outbound Proxy Port: Outbound Proxy server port

RTP port Number Setting: RTP local port, the minimum value is 4

NAT Traversal

1. NONE: disable NAT traversal
2. UPnP: use UPnP, need the support of upper gateway
3. Stun Server IP: Fill in your stun server IP when using stun method
4. Stun Server Port: Fill in stun server port

Gateway Settings

1. Dial Plan: please refer to dial_plan
2. # use as a quick dial function: the number will send immediately after you press the # button

3. To enable # to be recognized as dial number: use # as a dial number
4. * use as a quick dial function: the number will send immediately after you press the * button
5. To enable * to be recognized as dial number: use * as a dial number

2.3.2 Extensions

Server

Extensions

Digit Map

User 1

OOB Signalling

ToS/DiffServ

Tone

Ring

Service Code

Phone Book

SIP Extensions

- Support PRACK method with provisional response reliability
- Encode SIP URI with user parameter
- Session Timer use UPDATE method
- Call Hold using c=0.0.0.0 (RFC 2543) in SDP
- enable Global Number support (E.164)
- send NOTIFY for REFER request
- send Message Waiting Indicator (MWI) SUBSCRIBE command
- No Authorization Header in re-REGISTER
- Check existence of To Tag in INVITE 2xx response

SIP Timers

- Send INVITE with Timer header value: Seconds
- SIP Session Timer value: Seconds
- SIP Keep Alive Timer value: Seconds
- Conditional Call Forwarding Timer: Seconds
- Inter Digit Timer: Seconds.
- SIP T1 Timer: Milliseconds
- SIP T2 Timer: Milliseconds
- SIP T4 Timer: Milliseconds

SIP Extensions:

1. Support PRACK method with provisional response reliability: enable SIP PRACK support
2. Encode SIP URI with user parameter: encode user=phone parameter in SIP URI
3. Session Timer use UPDATE method: enable SIP session timer function.
4. Enable Global Number support(E.164): enable E.164 support.
5. Call Hold using c=0.0.0.0 (RFC 2543) in SDP:using the call hold method described in RFC 2543. If unchecked, the call hold would follow RFC 3263 method
6. Send NOTIFY for REFER request: send out NOTIFY request to

transfer for unattended and attended call transfer.

Server

Extensions

Digit Map

User 1

OOB Signalling

ToS/DiffServ

Tone

Ring

Service Code

Phone Book

Gateway Settings

Dial Plan:

Name	Digits for matching	Operation	Digits for operation
Digit Map1	<input type="text"/>	dropped <input type="button" value="v"/>	<input type="text"/>
Digit Map2	<input type="text"/>	dropped <input type="button" value="v"/>	<input type="text"/>
Digit Map3	<input type="text"/>	dropped <input type="button" value="v"/>	<input type="text"/>
Digit Map4	<input type="text"/>	dropped <input type="button" value="v"/>	<input type="text"/>
Digit Map5	<input type="text"/>	dropped <input type="button" value="v"/>	<input type="text"/>
Digit Map6	<input type="text"/>	dropped <input type="button" value="v"/>	<input type="text"/>
Digit Map7	<input type="text"/>	dropped <input type="button" value="v"/>	<input type="text"/>
Digit Map8	<input type="text"/>	dropped <input type="button" value="v"/>	<input type="text"/>

*The fields must be set to 'null' if this field will do nothing.

use as a quick dial function
 To enable # to be recognized as dial number

* use as a quick dial function
 To enable * to be recognized as dial number

2.3.3 User1 Configuration(User2 is the same as User1)

Server

Extensions

Digit Map

User 1

OOB Signalling

ToS/DiffServ

Tone

Ring

Service Code

Phone Book

User 1 Configuration

	Line 1	Phone Number	CallerID Name	Port	User Name	Password
Primary Server		<input type="text"/>	<input type="text"/>	5060	<input type="text"/>	<input type="text"/>
Secondary Server		<input type="text"/>	<input type="text"/>	5060	<input type="text"/>	<input type="text"/>

Line1 AEC Control

Line1 Gain Control

Input Gain Control (-12 ~ 18)db db

Output Gain Control (-12 ~ 18)db db

Supplementary Service Subscription

- Enable Call Waiting (Reject second incoming call)
- Enable Caller-ID Display
- Reject anonymous call
- Block Caller-ID in outgoing call

Primary Server, Secondary Server

Phone Number: phone number.

CallerID Name: caller ID

Port Name: Local register port. (Note: please assign different port to different user)

User Name: user name.

Password: password.

Line1 AEC Control: enable AEC (Acoustic Echo Cancellation) function, if the other hear a significant echo, please check this option.

Line1 Gain Control:

Input Gain Control (-12 ~ 18) db: input volume control.

Output Gain Control (-12 ~ 18) db: output volume control.

Supplementary Service Subscription:

Enable Call Waiting (Reject second incoming call): enable call waiting.

Enable Caller ID: enable caller ID display.

Reject anonymous: reject anonymous call.

Block Caller ID in outgoing call: use anonymous Caller ID.

Distinctive Ring Settings: set distinctive ring to different user.

Speed Dial Setting: speed dial number setting.

2.3.4 OOB Signalling

RTP Telephone Event Configuration

Send DTMF Events

RFC2833 signalling using payload value:

Regenerate OOB DTMF tone

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

2.3.5 ToS/DiffServ

ToS/DiffServ

Call Signalling Packets: (2 Hex digit byte value)

RTP Packets: (2 Hex digit byte value)

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

Press “Save ToS/DiffServ Settings” to save these new settings.

2.3.6 Tone

Server

Extensions

Digit Map

User 1

OOB Signalling

ToS/DiffServ

Tone

Ring

Service Code

Phone Book

Tone Configuration

Dial Tone:	<code>3500-13+4400-13#ON(1000),R</code>
Recall Dial Tone:	<code>3500-13+4400-13#[ON(100),OFF(100)]3,ON(1000),R</code>
Confirm Tone:	<code>3500-13+4400-13#[ON(100),OFF(100)]3,OFF(1000),R</code>
Ring Back Tone:	<code>4400-19+4800-19#ON(2000),OFF(4000),R</code>
Busy Tone:	<code>4800-24+6200-24#ON(500),OFF(500),R</code>
Reorder Tone:	<code>4800-24+6200-24#ON(250),OFF(250),R</code>
Receiver-Off-Hook Tone:	<code>14000-3+20600-3+24500-3+26000-3#ON(100),OFF(100),R</code>
Message-Waiting Indicator Tone:	<code>3500-13+4400-13#[ON(100),OFF(100)]10</code>
Call-Waiting Indicator Tone:	<code>4000-14#ON(150)</code>

Set UTA-1001 ring tone for different region.

2.3.7 Ring

- Server
- Extensions
- Digit Map
- User 1
- OOB Signalling
- ToS/DiffServ
- Tone
- Ring**
- Service Code
- Phone Book

Ring Configuration

Default Ring:

Call-Waiting
Reminder Ring:

Distinctive Ring Configuration

Distinct Ring 1:

Distinct Ring 2:

Distinct Ring 3:

Distinct Ring 4:

Distinct Ring 5:

Distinct Ring 6:

Distinct Ring 7:

Distinct Ring 8:

Set different user's ring tone, co-work with "SIP→User→Distinctive Ring Settings"

2.3.8 Service Code

Server

Extensions

Digit Map

User 1

OOB Signalling

ToS/DiffServ

Tone

Ring

Service Code

Phone Book

Service Code Configuration

Conditional Call Forwarding:

Call Forwarding On Busy:

Call Forwarding On:

Call Forwarding Off:

Do Not Disturb On:

Do Not Disturb Off:

Call Transfer:

Call Return:

Speed Dial:

- use *XX# or #xx# format , xe=01-99

Save Service Code Settings

Please refer to [value_add_service](#) for the use of service code.

2.4 CODECS

Audio/CODEC Configuration

CODECS

Selected	Silence Suppression
G711U	OFF ▼
G711A	OFF ▼
<input type="checkbox"/> G723	OFF ▼
<input type="checkbox"/> G726	OFF ▼
<input checked="" type="checkbox"/> G729	OFF ▼

Packetization 20ms ▼

Jitter Buffer

Adaptive Jitter Buffer: 100ms ▼ (maximum playout delay in milliseconds)

Fixed Jitter Buffer: 40ms ▼ (fixed playout delay in milliseconds)

Automatically switch to Fixed Jitter Buffer upon fax/modem tone detection

Save CODEC Configuration

CODECS:

Support CODEC: G711U、G711A、G723、G726、G729。
 Silence Suppression: enable VAD.

Packetization:

Configure the packet sending increments

Jitter Buffer

configure the timing of the voice buffering.
 Selection between adaptive or fixed jitter buffer. Default = ADAPTIVE
 Set the adaptive jitter buffer maximum playout delay. Default = 100ms
 or Fixed jitter buffer playout delay. Default = 40ms
 Whether or not to automatically switch from an adaptive jitter buffer to a
 fixed jitter buffer upon fax/modem tone detection
 Click on “Save CODEC Configuration” to save the configurations made.

2.5 System

2.5.1 Security, Timeout

Security
Timeout
Localization
Handset
SNMP
Service Access

Set Security Password

Password is currently installed

Account: admin

Old password:

New password:

Confirm new password:

Change Password

Security
Timeout
Localization
Handset
SNMP
Service Access

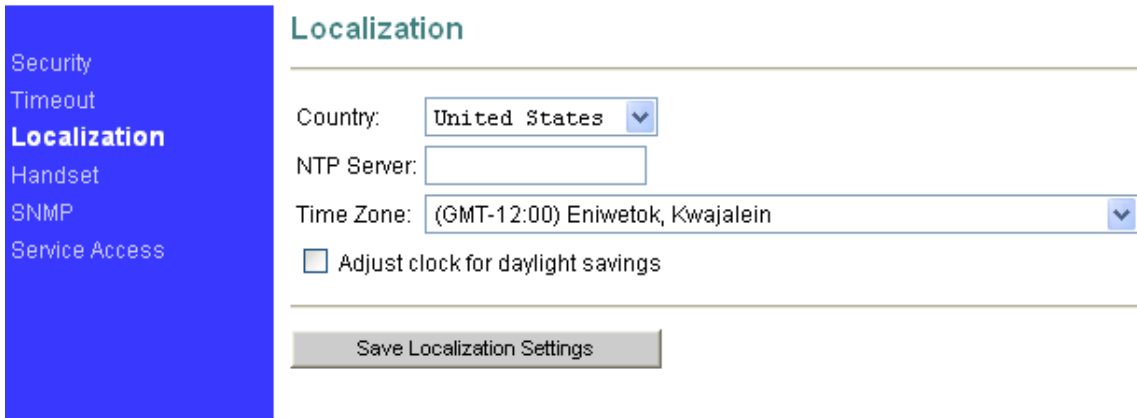
Set Web System Timeout

HTTP Authentication Timeout: (Seconds)

Change Time

Setting web security and authentication timeout

2.5.2 Localization



The screenshot shows a web-based configuration interface for the UTA-1001. On the left is a blue sidebar menu with the following items: Security, Timeout, Localization (highlighted in white), Handset, SNMP, and Service Access. The main content area is titled "Localization" in green text. Below the title is a horizontal line. The settings are as follows: "Country:" with a dropdown menu showing "United States"; "NTP Server:" with an empty text input field; "Time Zone:" with a dropdown menu showing "(GMT-12:00) Eniwetok, Kwajalein"; and a checkbox labeled "Adjust clock for daylight savings" which is currently unchecked. Below these settings is another horizontal line, followed by a grey button labeled "Save Localization Settings".

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

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

2.5.3 Handset

Security

Timeout

Localization

Handset

SNMP

Service Access

Media Hub Handset Configuration

Control Timer Values

Hook Flash Timer Min: Milliseconds

Hook Flash Timer Max: Milliseconds

***Please enter a multiple of 10.(ex:10,20,30....)**

Save Handset Settings

Hook Flash timing setting

Hook Flash Timer Min: minimum available time, unit: ms.

Hook Flash Timer Max: maximum available time, unit: ms.

When you press the flash during the time range you set, your action will act as hold function, and otherwise it will act as the hang up function. In some application, user wants to call another people immediately after he put down the handset, he can set the hook flash timer during the range: 10ms~20ms

2.5.4 SNMP Configuration

SNMP Trap Configuration

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

SNMP Community Configuration

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

SNMP System Configuration

System Description: Description of the unit (e.g. "John's phone")
 System Object Id: A vendor's enterprise ID

2.6 Download

2.6.1 Download

Download
AutoUpdate

Download
Warning: The download process will reset the unit into the download mode. This will terminate all network connections and reset your browser connection.

TFTP Download method (Select remote TFTP server IP address and filename)

TFTP Server IP:

Filename:

HTTP Download method (Select filename on local browser machine)

Filename:

URL Download method (Currently tftp://, http:// and https:// are supported)

URL:

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

Note: The available upgrade firmware is in .r0 extension; make sure you use the correct firmware before you update the firmware.

2.7 Configuration

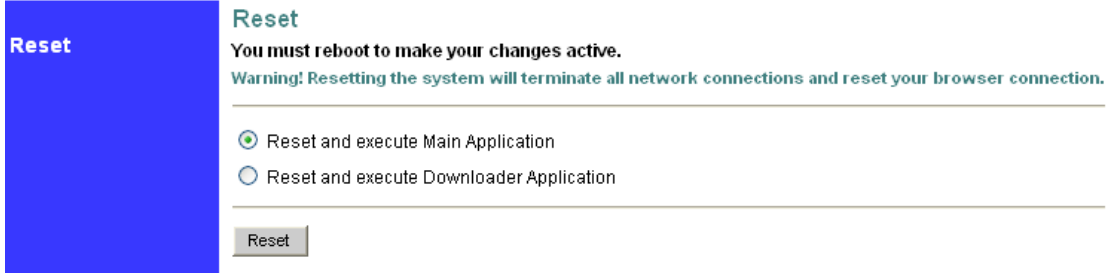
2.7.1 Backup and restore settings

The screenshot shows a web-based configuration interface. On the left, there is a blue sidebar with two sections: 'Backup' (with 'Restore' as a sub-option) and 'Restore' (with 'Backup' as a sub-option). The main content area is divided into three sections:

- Configure File Backup:** A section with a horizontal line above and below it. It contains a single button labeled 'Backup Configure File'.
- Configure Restore:** A section with a horizontal line above and below it. It contains the text 'Configure Restore method (Select filename on local browser machine)'. Below this is a 'Filename:' label followed by an empty text input field and a 'Browse...' button. Below the input field is a 'Start Download' button.
- Restore Factory Default:** A section with a horizontal line above and below it. It contains a single button labeled 'Start Restore Default Factory'.

Back up and restore the configure files.

2.8 Reset



UTA-1001 will save the current settings and reset by clicking the “reset” button

3. Restore to factory default

If your UTA-1001 settings is in chaos or you can't get the UTA-1001 IP to access it, you can reset the device to factory default:

- a) Power off
- b) Press reset button and power on
- c) The PWR led will light →Then PWR , SYS, WAN and LAN led will blink→ Then the PWR and SYS led light.
- d) Release the reset button after about 15~20 seconds,
- e) The PWR, SYS, WAN and LAN led will blink for a while and then the PWR and SYS led will light.

UTA-1001 will be reset to factory default after the above procedure, you can then access UTA-1001 through its LAN port, please refer to [access UTA-1001](#) for details.

4. FAQ

Q1 What is the default account of UTA-1001?

A1: The default account is:
 Administrator: user name: admin password: voip;
 User: user name: user password: voip

Q2 How to use the IVR function of UTA-1001?

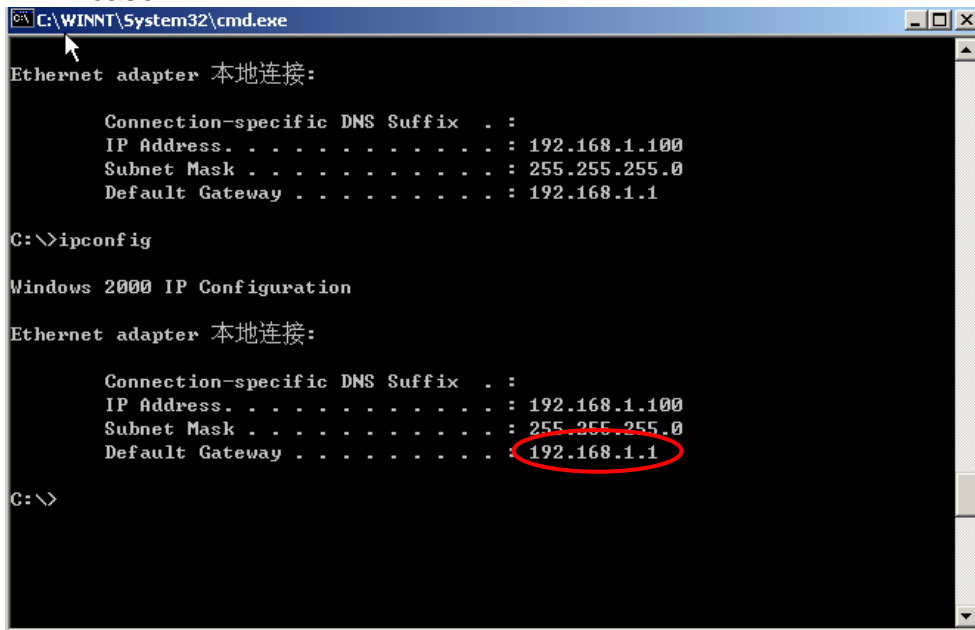
A2:
 The IVR function is record in G729 codec, so you have to choose G729 codec to active the IVR.
 You can use IVR function to observe and set the WAN port network parameters pick up the handset and dial **** to enter IVR mode.

Key	Function	Input
****	Enter main menu	Submenu
100#	Check network state	
110#	DHCP Settings	1# Enable DHCP function 2# Disable DHCP function or# back to main menu
120#	Static IP address Settings	Use "*" replace ".", and "#" as end. For example: 172*16*230*227# or # back to main menu
130#	Gateway IP settings	Use "*" replace ".", and "#" as end. For example: 172*16*230*1# or # back to main menu
140#	Subnet mask settings	Use "*" replace ".", and "#" as end. For example: 255*255*255*255# or # back to main menu

Q3 How can I know the IP address of UTA-1001?

A3: you can use the following methods to obtain UTA-1001's IP :

1. Use IVR function;
2. Observe the IP from the upper gateway;
3. Use the DHCP assignment function of UTA-1001
 - i. Reset UTA-1001 to factory default ,
 - ii. Directly connect your computer and UTA-1001' LAN port,
 - iii. Set your computer to dynamic obtain IP,
 - iv. Use command "ipconfig" to view your computer network status, and the gateway IP is the UTA-1001 LAN port IP. It is 192.168.1.1 in this case.



- v. Then you can use this IP address to access UTA-1001 through its LAN port the default account is admin/voip



Q4 How to update UTA-1001 firmware?

A4: Go to Download→Download, press “browse” in the http download method, and choose the correct firmware file (a 1.5M file in .r0 extension),and press the “Start HTTP Download” to perform updating.

Q5 How to use dial plan?**A5:**

A dial plan gives the unit a map to determine when a complete number has been entered and should be passed to the gatekeeper for resolution into an IP address. Dial plans are expressed using the same syntax as used by MGCP NCS specification.

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

```
Digit ::= "0" | "1" | "2" | "3" | "4" | "5" | "6" | "7" | "8" | "9"
Timer ::= "T" | "t"
Letter ::= Digit | Timer | "#" | "*" | "A" | "a" | "B" | "b" | "C" | "c" | "D" | "d"
Range ::= "X" | "x" -- matches any digit
| "[" Letters "]" -- matches any of the specified letters
Letters ::= Subrange | Subrange Letters
Subrange ::= Letter -- matches the specified letter
| Digit "-" Digit -- matches any digit between first and last
Position ::= Letter | Range
StringElement ::= Position -- matches any occurrence of the position
| Position "." -- matches an arbitrary number of occurrences
including 0
String ::= StringElement | StringElement String
StringList ::= String | String "|" StringList
DialPlan ::= String | "(" StringList ")"
```

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

Simple Dial Plan

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

Complex Dial Plan

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

Dial plan for this is:

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

Q6 How to use the value add service of UTA-1001?

Service Code Configuration

Conditional Call Forwarding:	<input type="text" value="*70#"/>
Call Forwarding On:	<input type="text" value="*72#"/>
Call Forwarding Off:	<input type="text" value="#72#"/>
Do Not Disturb On:	<input type="text" value="*74#"/>
Do Not Disturb Off:	<input type="text" value="#74#"/>
Call Transfer:	<input type="text" value="*98#"/>
Call Return:	<input type="text" value="*69#"/>
Speed Dial:	<input type="text" value="*68"/>

- use *XX# or #xx# format , xx=01-99

Save Service Code Settings

You need to set the service code for using the UTA-1001 value add service. For example, I set the service code as the above picture.

Condition Call Forwarding: (the call will transfer if no one answer)

- Set forwarding number: pick up the handset → press *70# → then you will hear the dial tone → press the forwarding number → then you will here three beeps indicating setting finish.
- Set the timeout: go to the “sip extensions → Conditional call Forwarding timer” and set the timeout before forwarding, unit: second, and then active this option.
- Then the call will automatically transfer to the forwarding number if no

one answers the call in the timeout.

Call Forwarding: (forwarding always)

- a) Enable call forwarding: pick up the handset → press *72# → then you will hear the dial tone → press the forwarding number → then you will here three beeps indicating setting finish, then all incoming call will forward to this number automatically.
- b) Disable call forwarding: pick up the handset → press #72# → then you will here three beeps indicating setting finish

Do not disturb: (DND)

- a) Enable DND: pick up the handset → press *74# → then you will here three beeps indicating setting finish → then the phone won't ringing when there is an incoming call.
- b) Disable DND: pick up the handset → press #74# → then you will here three beeps indicating setting finish

Call transfer:

- a) Unattended transfer: A call B → B press *98# and then enter C number → then B will hear three beeps indicating the transfer successfully.
- b) Attended transfer: A calls B → B push the hook flash to hold A → B then dial C number to talk with C → then B press *98# to transfer the call → then A can talk with C.

Call Return:

Pick up the handset → and then press *69# to dial the latest received call

3 way conference call:

A calls and talks with B → B push the hook flash to hold A B then dial C number to talk with C → B then push the hook flash again to enable three way conference call → C will leave the call is B push the hook-flash again.

Q7 How to configure UTA-1001?

A7 please refer to "UTA-1001 quick start guide"

Q8 How to change UTA-1001 LAN port MAC address?

A8 please access <http://UTA-1001ip/burn.htm> and change the MAC

address, after you have changed it, clap the reset button to save your setting.

Q9 Why does my UTA-1001 always drop off from the server?

A9

You can find the register TTL in the “SIP→server→ Send Registration Request with Expire Time”, if this time is longer then the system require register time, UTA-1001 will always drop off from the server, please set this time to a suitable value, (unit: seconds).

Q10 How to use the speed dial function?

A10

You need to set the speed dial number in the “SIP→User→ Speed Dial Settings”, and then set the operation code in the “Server→Service Code”, for example *68, then you can dial *681 to replace the speed dial number in the speed dial settings.

Speed Dial Settings

Speed Dial 1:	<input type="text" value="83018049"/>	Speed Dial 2:	<input type="text"/>
Speed Dial 3:	<input type="text"/>	Speed Dial 4:	<input type="text"/>
Speed Dial 5:	<input type="text"/>	Speed Dial 6:	<input type="text"/>
Speed Dial 7:	<input type="text"/>	Speed Dial 8:	<input type="text"/>

Save SIP User Settings