Ultrative Communications
E: sales@ultrative.com
T: +86-755-29630367
F: +86-755-29685231
http://www.ultrative.com

UTT-4 VoIP IAD Series Datasheet

The UTT-4 Voice Gateway series offer high-quality, high-function, and low-density access devices used in residential, SOHO, and mobile-office VoIP applications. It provides a reliable, low-cost, and flexible means to deploy converged communication solutions for network operators and enterprises as well. The UTT-4 series can be configured as either 2-in-1 with connections to Ethernet and analog phones, or 3-in-1 with connections to Ethernet, analog phones, and CO lines.

Key Features

1. Connect analog telephone, PBX, facsimile machine and POS machine to the IP core network, or PSTN
2. Work with service platforms to provide various telephone supplementary services
3. Support protocols: SIP (SIP trunk), MGCP
4. Flexible configuration of phone/line interfaces
5. Support static IP address configuration , DHCP and PPPoE
6. Support G.711ALaw, G.711Ulaw, G.729 and G.729B
7. Support echo cancellation
8. Up to 500 routing rules can be stored in gateways
9. Intercom. Built-in 3-way calling.
10. Support digitmap
11. Support call-progress tones for various countries and regions
12. Support second-stage dialing or voice prompt
13. Support PSTN failover through line ports
14. Security: IP filter, encryption
15. Support routing table

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16. Support T.38 version 3 fax relay with V.34
17. Support polarity-reverse and busy-tone detection
18. Compatible with all standard SIP Platforms ; unified communication solutions, like CallManager and OCS/Lync
19. Support multiple local and remote-maintenance & management modes such as Web, Telnet, auto-provision, and TR069/TR104/TR106 client

Specifications

Item	Features
Port configurations	UTT-411: 1 FXS and 1 FXO ports, 2 Ethernet ports
	UTT-422: 2 FXS and 2 FXO ports, 2 Ethernet ports
	UTT-402: 2 FXS port, 2 Ethernet ports
Case	Desktop and wall mountable
	Size: 150 x 109 x 30mm
	Weight: 300g
	LED: power, Ethernet, FXS, FXO, device status
Connector type	Ethernet: RJ45
	FXS：RJ11
	FXO：RJ11
Hardware	ATMEL AT91SAM9G20B with 400MHz
	32MB SDRAM
	8MB Flash memory
Powe supply	AC/DC Adaptor with input: 100-240V AC and output: 9V/0.66A DC
	Power consumption: 6W
Operating system	Linux kernel 2.6.27
SIP protocol RFC3261 RFC3262 RFC3264 RFC3311 RFC3515 RFC3581 RFC 3966 RFC4028	SIP registration (per trunk, per gateway)
	Registration expire setting
	SIP trunk
	Backup SIP proxy (up to 10 proxies)
	Peer-to-peer communication
	SIP-to-SIP relay
	Hook flash relay (INFO)
FXS	Polarity reversal generation
	Caller ID generation (Bellcore and ETSI FSK, DTMF, before/after ring)
	Ring cadence setting
	Ring frequency setting
	Volume control
	Hook flash timing setting

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	Message waiting indicator (FSK, polarity inverse)
FXO	Hot line
	Relay to FXS extension
	Redirect to SIP server
	Interactive Voice Response (IVR) and second-stage dialing
	Outbound calls from either FXS or service network
	Battery reversal detection
	Caller ID detection (Bellcore and ETSI FSK, DTMF, before/after ring)
	Busy tone detection
	DTMF out-pulsing timing setting
	Volume control
	Ring parameter setting
	Busyout when FXS is not available
Codec/FAX/RTP	G.711ALaw, G.711Ulaw, G.729 and G.729B
	T.30 fax transparent, T.38 fax relay
	Echo cancellation
	Dynamic jitter buffer management
	Static jitter buffer
	DTMF relay (RFC2833, SIP/INFO, inband)
	Adjustable packetization period, 10/20/30/40/50/60ms
	Call progress tones compliant with multi-nation standards
Voice QoS	IEEE 802.1p tag
	DiffServ code point (TOS) bits
Call control	Blind transfer
	Explicit transfer
	Call forward on busy
	Call forward on no answer
	Call forward variable
	Call waiting
	Caller ID
	Caller ID blocking
	Caller ID on call waiting
	Distinctive ring
	Do not disturb
	Music on hold
	Color ring back tone
	Call progress tone (configurable)
	Release control (called party control, calling party control)
	Built-in 3-way calling
	Speed dialing

	Calling and called number based routing
	Call number modification (add, delete, and replace)
	Hunt group (sequential and circular selection)
	Ring group
	Digit map
	PSTN failover (upon IP network break or failure to reach SIP proxy, or power break)
Networking	DHCP client
	DNS/DDNS client
	PPPoE client
	NAT traversal (STUN)
Security	IP filtering list (IP table)
	SIP/RTP/Telent/HTTP/TFTP port screening
	Web-utility access privilege (admin and user)
System management	TR069-based management (TR069, TR104, and TR106)
	Standard SNMP agent, MIB v2
	Web-utility for configuration, data import/export, and firmware upgrade
	Auto-provisioning for configuration and firmware upgrade
	Log management (8 levels)
	Syslog
	Debugging and call trace
	TCP dump
	System status monitoring and statistics (TR069, SNMP, Web)