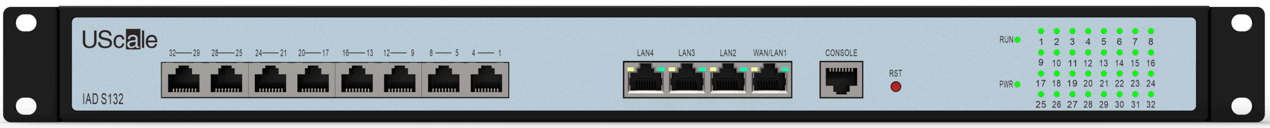
1. UScale OIAD Technical Specifications

1.1 UScale OIAD Overview

The UScale OIAD series is a high-density analog voice gateway / IPPBX access platform designed for medium and large-scale voice access and convergence scenarios. It provides flexible FXS access options with 32 / 96 / 224 ports, supports both RJ45 and DB68B physical interface types, and interoperates with mainstream SIP / VoIP ecosystems while handling multiple codecs in parallel. The platform supports IPv4 / IPv6, VLAN / QoS, secure access, remote O&M, and rich northbound management capabilities, making it an ideal choice for TDM–IP voice migration, call center PSTN aggregation, and branch office voice-to-cloud deployments. The UScale OIAD family includes three models: UScale OIAD S132, UScale OIAD S196, and UScale OIAD S1224. Each higher model provides more analog subscriber channels and supports step-by-step expansion from small to large deployments. The UScale OIAD S1224 supports up to 224 analog subscriber channels.

UScale OIAD S132 appearance is shown below:



UScale OIAD S196 appearance is shown below:

UScale OIAD S1224 appearance is shown below:

1.2 UScale OIAD Hardware Architecture

The UScale OIAD series analog gateways adopt a professional-grade hardware architecture with excellent reliability and scalability. The devices provide three density configurations with 32 / 96 / 224 ports, each port supporting FXS features and both RJ45 and DB68B interface types to maximize flexibility for different cabling scenarios. For network connectivity, the device provides 1 WAN port and 3 LAN ports, all supporting 10 / 100 / 1000 Mbps auto-negotiation to ensure high-speed and stable data transmission. In addition, there is 1 RS232 management port (RJ45 interface) that supports local serial configuration and debugging, so that device management is still possible in case of network failures. The hardware core is built on a high-performance processor combined with dedicated DSP chips, optimized for voice processing, echo cancellation, and codec transcoding to ensure voice quality and system stability. The device supports up to 3 km line transmission distance (based on 24 AWG cable) and provides 2 REN drive capability for remote FXS devices, fully meeting the long-distance deployment requirements of large campuses and factories.

1.3 UScale OIAD Performance Metrics

Parameter Category	Technical Specification	S132	S196	S1224
Physical Specifications	Number of Ports	32 FXS	96 FXS	224 FXS
	Interface Type	RJ45	DB68B	DB68B
	Network Interfaces	1×WAN, 3×LAN (10/100/1000M)		
	Management Interface	1×RS232 (RJ45)		
	Dimensions	440×44.4×318 mm	440×44.4×317 mm	440×86×317 mm
	Weight	2.44 kg	6 kg	10 kg
	Power Consumption	52 W	150 W	250 W
Power and Environmental	Power Input	100–240 V AC dual power supported or −48 to −60 V DC	100 / 240 V AC or −48 to −60 V DC	100 / 240 V AC dual power supported or −48 to −60 V DC
	Operating Temperature	0 °C to 45 °C		
	Operating Humidity	10% to 90% RH (non-condensing)		
	Storage Temperature	−20 °C to 70 °C		
	Certification	CE		
Voice Processing	Codecs	G.711A, G.729A, G.722, G.726, G.711U, iLBC		
	Echo Cancellation	ITU-T G.168 compliant, tail length up to 128 ms		
	Voice Gain	Programmable TX / RX gain control, ±12 dB range		
	Jitter Buffer	Adaptive dynamic jitter buffer, 20–200 ms		
	Line Driving	Maximum 3 km loop length, 2 REN drive capability		

Parameter Category	Technical Specification	S132	S196	S1224
Fax Features	Fax Protocol	T.38 and G.711 Pass-through dual modes		
	Fax Rate	Up to 14.4 kbps		
	Modulation Standards	V.17, V.21, V.27ter, V.29		
	ECM Support	Supports Error Correction Mode (ECM)		
Call Handling	Dialing Modes	DTMF, pulse dialing (10 / 20 PPS)		
	Caller ID	DTMF / FSK standards		
	DTMF Transport	SIP INFO, RFC4733, INBAND modes		
	Call Features	Call waiting, transfer, hold, three-party call, DND		
Network Protocols	SIP Protocol	SIP v2.0 RFC 3261, UDP / TCP / TLS transport		
	Media Protocols	RTP / RTCP (RFC 2833, RFC 1889)		
	Network Protocols	IPv4 / IPv6 dual stack support		
	QoS Support	DiffServ, ToS, 802.1P / 802.1Q		
	NAT Traversal	STUN, rport (RFC 3581)		
	Security Protocols	TLS, SRTP, HTTPS, 802.1x		
Management and Maintenance	Management Interfaces	Web (HTTP / HTTPS), SSH		
	Monitoring Protocols	SNMP v1 / v2c		
	Logging System	8 log levels (EMERG–DEBUG)		
	Time Synchronization	NTP client, daylight saving time support		
	Configuration Backup	Supports configuration import / export		
	Firmware Upgrade	Web online upgrade, TFTP / HTTP		

Parameter Category	Technical Specification	S132	S196	S1224
Service Features	Concurrent Calls	Full-port concurrent support		
	Ring Group	Supports ring group configuration		
	CDR Records	Detailed call detail records, export supported		
	Auto Provisioning	DHCP Option 66, TR-069		
	VPN Support	OpenVPN client		
Compatibility	IP-PBX	Asterisk, Issabel, 3CX, FreeSWITCH, BroadSoft, VOS		
	Special Devices	Modem, POS terminal compatibility		
	Telephone Sets	Standard analog phones, fax machines		

2. UScale OIAD Supported Services

2.1 Basic Service Functions

2.1.1 Basic Connectivity Functions

- **Multi-port FXS interface support**: The device provides flexible configurations of 32 / 96 / 224 FXS ports, supporting analog phone access and PSTN line connection to meet the basic communication requirements of users of different scales.
- **Gigabit Ethernet interfaces**: Equipped with 1 WAN port and 3 LAN ports, all supporting 10 / 100 / 1000 Mbps auto-negotiation, providing a high-speed and stable network connection foundation.

2.1.2 Basic Call Functions

- **DTMF and pulse dialing**: Supports both DTMF dual-tone multi-frequency and pulse dialing modes, compatible with modern and legacy telephone devices to ensure normal dialing operation.
- **Caller ID**: Supports both DTMF and FSK caller ID standards, accurately receiving and parsing calling numbers and forwarding them, achieving interoperability of caller ID between TDM and IP networks.
- **Busy tone detection**: Intelligently identifies PSTN busy tones and automatically releases calls when busy tones are detected, avoiding invalid occupation of trunks and improving line utilization.
- **Polarity reversal detection**: Monitors polarity reversal signals on the line in real time to accurately determine off-hook and on-hook status changes, ensuring accurate call state tracking and billing.

2.1.3 Basic Voice Processing

- **Multi-codec support**: Supports multiple audio codecs such as G.711A / U, G.729A, and G.722, and can automatically select the optimal codec according to network conditions.
- **G.168 echo cancellation**: Provides echo cancellation compliant with ITU-T G.168 with a maximum tail length of 128 ms, effectively suppressing echo and improving voice quality.

- **Programmable gain control**: Supports independent gain adjustment for transmit and receive audio, allowing optimization according to line characteristics and ambient noise to ensure the best call quality.

2.1.4 Basic Protocol Support

- **SIP v2.0 protocol support**: Fully supports the SIP v2.0 protocol stack and RFC 3261 standard, with UDP or TCP transport, achieving standardized interoperability with mainstream IP-PBX systems.
- **RTP / RTCP support**: Supports RTP for real-time transport of audio data and RTCP for monitoring transport quality, compliant with RFC 2833 and RFC 1889.
- **IPv4 basic support**: Provides basic IPv4 network protocol support to ensure normal communication and data transmission in existing network environments.

2.2 Supplementary Service Functions

2.2.1 Supplementary Call Processing Functions

- **Call waiting**: When there is a new incoming call during an ongoing call, the device plays a call waiting tone instead of busy tone. The user can choose to answer or reject the new call, enabling efficient use of a single line.
- **Attended transfer**: Supports attended transfer, where the user first establishes a call with a third party to confirm availability, then completes the transfer, improving transfer success rate and user experience.
- **Blind transfer**: Allows the user to transfer a call directly to a third party without first confirming availability, suitable for scenarios requiring fast call handoff.
- **Call forward on busy**: When the called party is busy, incoming calls are automatically forwarded to a preset number to avoid the caller hearing busy tone and degrading experience.
- **Call forward on no answer**: If the called party does not answer within a configured time, the call is automatically forwarded to a specified number. Administrators can flexibly configure the no-answer timeout.
- **Unconditional call forward**: All incoming calls to the number are unconditionally forwarded to a preset number, commonly used for temporary leave or vacation to ensure important calls are not missed.
- **Call hold**: Users can place the current call on hold to handle other tasks and then resume the call. While on hold, music-on-hold can be played.

2.2.2 User Convenience Functions

- **Hotline**: Configures specific numbers as hotlines so that picking up the handset automatically dials the preset number without manual dialing, suitable for emergency help and customer service hotlines.
- **Message Waiting Indicator (MWI)**: When a user has unheard voicemail, the MWI lamp on compatible phones lights up to remind the user to check and handle the messages in time.
- **Do Not Disturb (DND)**: Users can enable DND mode to block incoming calls. Callers hear an announcement or are sent to voicemail, suitable for meetings or rest periods.
- **Speed dial**: Supports speed-dial functionality by mapping frequently used long numbers to short codes, enabling quick dialing with fewer digits.
- **Three-way calling**: Supports three-party or multi-party calling, allowing the user to invite a third party

into an ongoing call to form an ad hoc conference and improve collaboration efficiency.

2.2.3 Supplementary System Management Functions

- **Ring group**: Multiple ports can be grouped into a ring group so that all phones in the group ring simultaneously for an incoming call, and any member can answer, suitable for attendant or hotline desks.
- **Web GUI configuration**: Provides an intuitive web-based management interface that supports graphical configuration of all features, allowing non-expert personnel to manage and configure the device.
- **Data backup and restore**: Supports configuration backup and restore. Configuration files can be exported for backup or imported in bulk, enabling rapid deployment and data protection.
- **CDR statistics**: Records detailed call detail records, including calling and called numbers, call duration, start and end time, and supports export for billing and analysis.

2.2.4 Network Enhancement Functions

- **IPv6 support**: Supports both IPv4 and IPv6 and can operate in pure IPv4, pure IPv6, or dual-stack environments, following network protocol evolution trends.
- **VLAN support**: Supports IEEE 802.1P service quality tagging and 802.1Q VLAN tagging to implement voice traffic isolation and priority differentiation.
- **NAT traversal support**: Supports rport as per RFC 3581 and other NAT traversal techniques to resolve connectivity issues behind NAT and improve applicability in complex networks.
- **Session Timer support**: Supports Session Timer per RFC 4028 to periodically refresh session state and promptly detect and clear hung calls.

2.3 Value-Added Service Functions

2.3.1 Enterprise-Level Management Functions

- **SNMP monitoring support**: Supports SNMP v1 and v2c, allowing integration with NMS platforms for device status monitoring, performance statistics, and fault alarms.
- **Auto provisioning**: Supports DHCP Option and TR-069-based auto provisioning mechanisms, enabling zero-touch deployment and greatly simplifying large-scale rollout.
- **Cloud-based management**: Supports centralized management via a cloud platform for unified configuration, monitoring, and maintenance of multiple devices, ideal for distributed deployments with centralized O&M.
- **Multi-level logging system**: Supports multiple log levels such as Debug, Info, and Error. Administrators can adjust log verbosity to facilitate troubleshooting and system tuning.

2.3.2 Security Value-Added Functions

- **Web ACL access control**: IP-based web management access control lists restrict access to the management interface to authorized IP addresses only, improving system security.
- **OpenVPN support**: Built-in OpenVPN client capabilities establish encrypted VPN tunnels to remote networks, ensuring secure transmission of voice traffic.

2.3.3 Business Integration Functions

- **Action URL**: Sends HTTP requests to a specified URL when specific events occur, such as call start and end, to notify third-party systems and facilitate integration.
- **API support**: Provides standardized application programming interfaces for third-party systems to exchange data and invoke functions, enabling customized integration and development.
- **Call routing policies**: Supports flexible routing configuration based on time, source, destination, and other conditions to implement least-cost routing, load balancing, and time-based routing.
- **Number manipulation**: Supports flexible caller and callee number manipulation, allowing modification, masking, or replacement of displayed numbers to meet privacy requirements.

2.3.4 Advanced Voice Functions

- **T.38 fax and pass-through support**: Supports both T.38 and pass-through fax modes. T.38 digitizes fax signals, while pass-through preserves the original waveform, ensuring fax reliability under different conditions.
- **Modem / POS support**: Optimized for modems and POS terminals to ensure dial-up and data transmission operate correctly over IP networks.
- **Multi-registrar support**: Supports simultaneous registration to multiple SIP servers, enabling load balancing and redundancy. When the primary server fails, the device automatically switches to a backup server.

2.3.5 Carrier-Grade Functions

- **Long-loop support**: FXS ports support loop lengths up to 3 km based on 24 AWG cable, meeting long-distance cabling requirements in large campuses and factories.
- **Advanced self-switching**: Calls between internal ports can be switched locally within the device without relying on an external PBX, reducing system complexity and cost.
- **Network packet capture**: Supports capturing network packets, including SIP signaling and RTP media streams, for in-depth troubleshooting and protocol analysis.
- **Ping / Traceroute tests**: Built-in network connectivity test tools such as Ping and Traceroute help quickly diagnose link issues and trace routing paths.

2.3.6 Industry Customization Functions

- **PPPoE support**: Supports PPPoE so that the device can act as a client to obtain a dynamic IP address, suitable for ADSL and other broadband access scenarios.
- **Digit manipulation rules**: Supports flexible digit manipulation rules for adding, deleting, or replacing digits to adapt to different dialing habits and numbering formats.
- **Programmable call progress tones**: Allows customization of dial tone, ringback tone, busy tone, and other call progress tones in terms of frequency and cadence to match country-specific standards.
- **Multi-mode DTMF support**: Supports SIP INFO, RFC4733, and in-band DTMF transport, automatically choosing the appropriate method according to peer compatibility.

3. UScale OIAD Compliance Standards

Standard ID	Description
ITU-T G.711 A-law/μ-law	64 kbps high-quality voice codec, full-duplex
ITU-T G.722	7 kHz wideband audio at 48 / 56 / 64 kbps
ITU-T G.723.1	5.3 / 6.3 kbps low-bit-rate codec for narrowband
ITU-T G.726	ADPCM at 16 / 24 / 32 / 40 kbps
ITU-T G.729A/B	8 kbps codec with excellent quality and efficiency
iLBC	13.3 / 15.2 kbps codec with packet loss recovery
OPUS	6–510 kbps variable bit rate with high quality
AMR/AMR-WB	4.75–23.85 kbps mobile network speech codecs
ITU-T T.30	Fax session establishment and control procedures
ITU-T T.38	Real-time IP transmission of fax signals
ITU-T V.17	7200 / 9600 / 12000 / 14400 bps fax data rates
ITU-T V.21	300 bps duplex communication for control signals
ITU-T V.27ter	2400 / 4800 bps fax data transmission
ITU-T V.29	7200 / 9600 bps high-speed fax data transmission
RFC 3261	SIP v2.0 core signaling, message format, sessions
RFC 3262	Reliable provisional responses
RFC 3263	SIP server discovery and selection
RFC 3264	Offer / answer model for media negotiation
RFC 3265	SUBSCRIBE / NOTIFY event framework
RFC 3311	Session parameter modification within a dialog
RFC 3515	Call transfer
RFC 2976	INFO method for mid-session signaling
RFC 4028	Session Timer to prevent hung sessions
RFC 3581	rport parameter for symmetric response routing
RFC 2806	TEL URI format
RFC 1889	RTP real-time media transport
RFC 1889	RTCP quality monitoring and control

Standard ID	Description
RFC 2833	DTMF and telephony events in RTP payload
RFC 2198	RTP payload for redundant audio data
RFC 2327	SDP for media session description
IPv4	Basic network layer connectivity and routing
IPv6	Next-generation protocol with extended address
TCP	Reliable connection-oriented transport
UDP	Connectionless low-latency transport
ICMP	Network diagnostics and error reporting
DHCP	Automatic IP address assignment
DNS	Domain name resolution
NTP	Network time synchronization
SNMP v1/v2c	Network device monitoring and management
HTTP/HTTPS	Web management interface access
Telnet	CLI-based remote management
SSH	Encrypted remote shell access
TFTP	Configuration and firmware file transfer
IEEE 802.1P	8-level QoS priority markings
IEEE 802.1Q	VLAN tagging and separation
DiffServ	IP layer QoS marking and classification
ToS	IP header type-of-service field
TLS v1.0/1.1/1.2	SIP signaling encryption
SRTP	End-to-end RTP media encryption
IPSec	Network-layer encryption and authentication
IEEE 802.1X	Port-based network access control
HTTPS	Encrypted web management access
STUN	NAT detection and traversal
PPPoE	Broadband dial-up access protocol
OpenVPN	Secure remote network connectivity

Standard ID	Description
ITU-T G.168	Echo cancellation with up to 128 ms tail
ITU-T G.711 Appendix II	Silence detection and comfort noise generation
ITU-T G.729 Annex B	Silence suppression and bandwidth optimization
CE	EU safety, health, and environmental compliance
FCC	US EMC and RF compliance
ITU-T	Compliance with ITU-T telecom standards