

- Power over Ethernet (PoE)
- 2 SIP accounts
- Gigabit Ethernet ports
- High quality of sound
- Programmable keys
- Autoprovision (TR-069, DHCP)
- Headset connection
- Large LCD
- Modern design



**VP-17P** — up-to-date IP phone with integrated gigabit switch, which provides PC and IP phone with connection through one physical line. The device offers advanced functionality, PoE support, high quality and modern design.

### Business Solution

VP-17P is designed for companies with high requirements to reliability, usability and quality of voice data transmission.

### High Quality of Sound

The high performance of VP-17P IP phone is provided by the hardware based on advanced Realtek chips. All the main audio codecs used in VoIP networks (G.711, G.723.1, G.726, G.729) are supported by VP-17P. Echo cancellation, Silence Detector, DTMF signals reception and generation as well as traffic prioritization (QoS) ensure high quality of voice data.

### Redundancy

In case of main Softswitch connection failure, VP-17P is switched automatically to a redundant SIP server with main server state control.

### AutoProvision

AutoProvision (AP) subsystem is used for centralized configuration, update and state monitoring of subscriber's equipment. The subsystem allows transparent replacing of a phone (not only by another model, but by a device from another vendor as well). After replacing a phone, the configuration will be adapted automatically.

### Phone Desktop Assistant

Phone Desktop Assistant application is designed for phone services extension. The application allows answering a call, arranging calls via click-to-call mechanism, viewing call history by searching and filtering according to subscriber's name, phone number, etc.

### Usability

Convenient key arrangement, intuitive menu and user-friendly web interface with multilingual support provide usability in corporate telephone networks.

## Features and capabilities

### Phone features

- 2 SIP accounts configured independently
- Support for BLF
- Programmable keys
- Flexible dialplan
- Displaying of caller name and number (CallerID)
- Mute
- Redial
- Hotline
- Ringtones upload/delete
- Support for speakerphone
- Support for headset

### Additional services

- Call Hold
- Call Transfer
- Call Waiting
- Call Forward on Busy (CFB)
- Call Forward on No Response (CFNR)
- Call Forward Unconditional (CFU)
- Do Not Disturb mode (DND)
- Caller Line Identification Restriction (CLIR)
- Local 3-Way Conference
- Support for remote conference according to RFC4579
- Hotline/Warmline
- Answering an intercom call
- Automatic Call Answer
- Call Pickup

### Display, indicators and keys

- 3.2 inch (81 mm) backlit monochrome display with 128x64 resolution
- Interface language selection (English and Russian)
- Functional keys with LED indication: message, headset, microphone mute, speakerphone
- Functional keys without LED indication: conference, call hold, call transfer, redial, volume adjustment
- 6 programmable keys

### VoIP protocol

- SIP

### Audio codec support

- Codecs: G.711 a-law,  $\mu$ -law, G.723.1, G.726, G.729
- DTMF: In-band, RFC2833, SIP INFO
- Voice Activity Detection (VAD)
- Acoustic Echo Suppression (AES)
- Comfort Noise Generation (CNG)

### Phonebook

- Local phonebook for 1000 phone numbers
- XML and LDAP remote phonebooks
- Searching in phonebooks
- Call history: dialed, received and missed calls

### Integration with IP PBX

- Geographical redundancy of SIP server (support for up to 4 redundant servers)
- Operation without SIP server
- Short text messages transmitting and receiving (SIP MESSAGE)
- Voice mail counters viewing
- Message Waiting Indicator (MWI)
- Displaying of watched subscriber line status (BLF)

### Diagnostics

- Monitoring of device status via web interface
- Displaying of debugging information in Syslog

### Monitoring and management

- Management web interface (English and Russian)
- SSH
- Autoprovision (TR-069, DHCP)
- Factory reset, reboot

### Network protocols and security

- Data network connection (Static, DHCP, No IP)
- Time and date synchronization via NTP
- QoS 802.1p, DSCP traffic marking
- Support for NAT-traversal: STUN mode, Public IP
- Support for LLDP, LLDP MED
- Support for SIP over TLS
- Support for SRTP
- LLDP/DHCP VLAN

### Specifications

- RFC 3261 SIP 2.0
- RFC 3262 SIP PRACK
- RFC 4566 Session Description Protocol (SDP)
- RFC 3263 Locating SIP servers for DNS lookup SRV and A records
- RFC 3264 SDP Offer/Answer Model
- RFC 3311 SIP Update
- RFC 3515 SIP REFER
- RFC 3891 SIP Replaces Header
- RFC 3892 SIP Referred-By Mechanism
- RFC 4028 SIP Session Timer
- RFC 2976 SIP INFO Method
- RFC 2833 RTP Payload for DTMF Digits, Flash event
- RFC 3108 Attributes ecan and silenceSup in SDP
- RFC 4579 SIP Call Control – Conferencing for User Agents
- RFC 3361 DHCP Option 120
- RFC 3550 RTP A Transport Protocol for Real-Time Applications
- RFC 3611 RTP Control Protocol Extended Reports (RTCP XR)
- RFC 3842 A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
- RFC 4235 An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP)

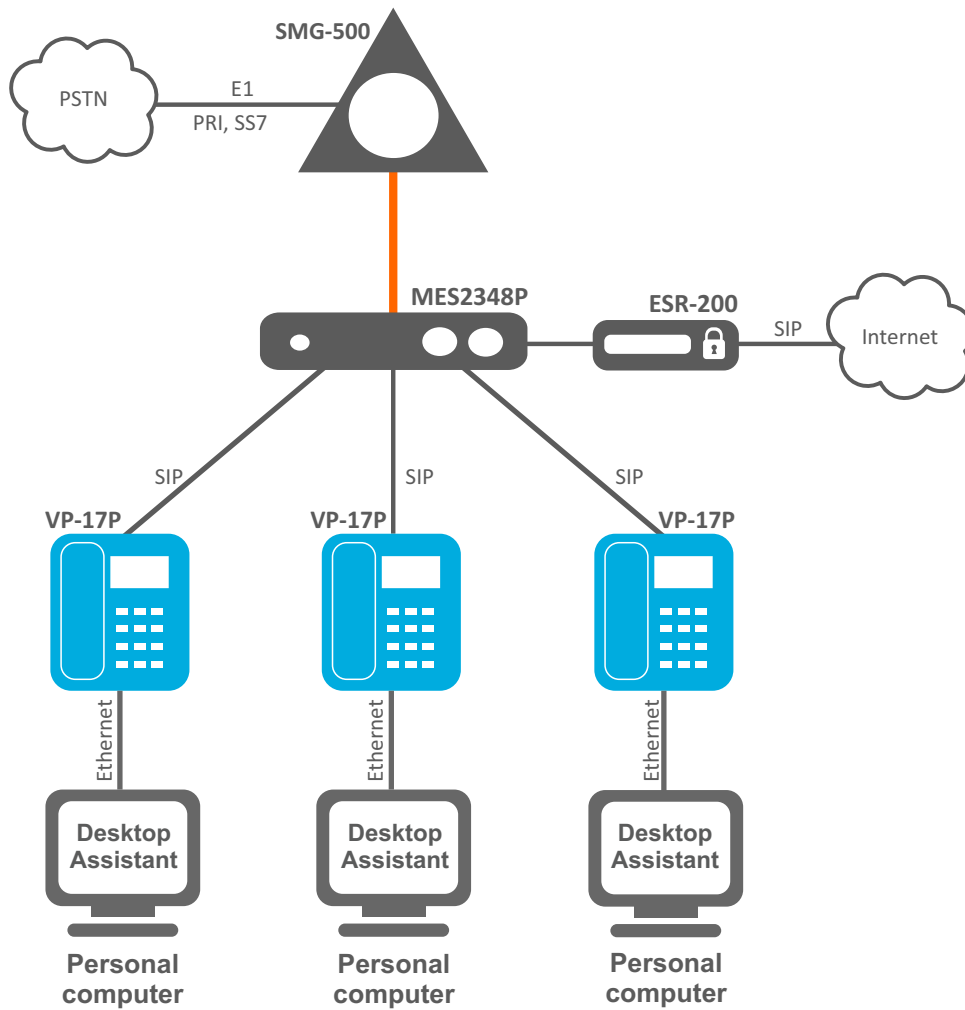
### Technical features

- Realtek 9607DQ processor
- SDRAM 512 MB
- SPI Flash 256 MB
- OC Linux 4.4
- Software redundancy

### Physical features

- 2xRJ-45 10/100/1000 Mbps Ethernet ports
- 1xRJ-9 (4P4C) for handset connection
- 1xRJ-9 (4P4C) for headset connection
- Power over Ethernet 802.3af
- Power consumption: no more than 4 W (maximum current consumption 0.8 A)
- Power adapter: 5 V DC, 2 A (optional)
- Operating temperature: from +5 °C to +40 °C
- Relative humidity at 25 °C: up to 80%
- Desktop implementation
- Dimensions: 205x210x86 mm
- Weight: 0.83 kg

Use case



Ordering information

Name	Description
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**VP-17P** IP phone VP-17P: 2 SIP accounts, 2 x 10/100/1000BASE-T (RJ-45) ports, LCD, PoE

Related products

<b>ECSS-AP-10</b>	Option ECSS-AP-10 for 10 devices auto-configuration
<b>ECSS-AP-50</b>	Option ECSS-AP-50 for 50 devices auto-configuration
<b>ECSS-AP-100</b>	Option ECSS-AP-100 for 100 devices auto-configuration

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