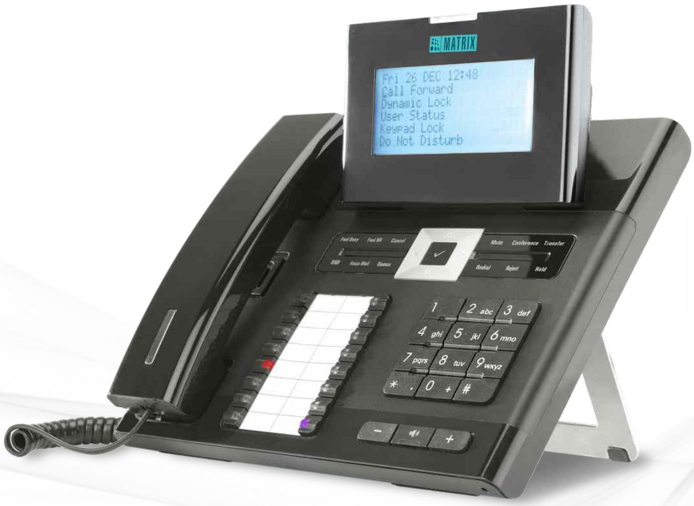


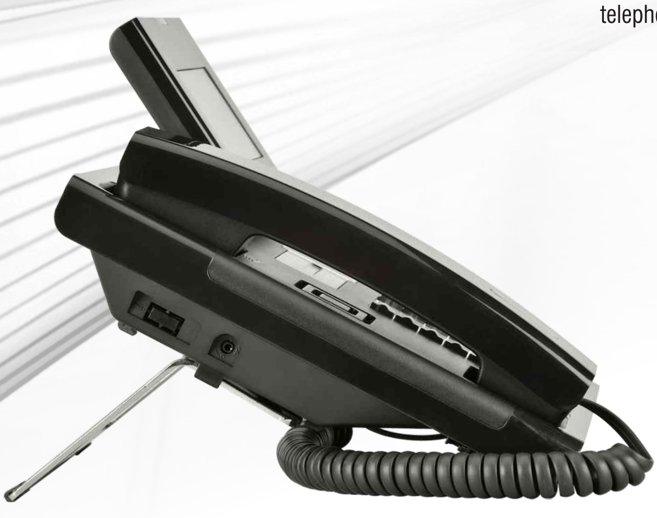
SPARSH VP

The High-definition Edge to Your IP Communication

Telecommunication has undergone many technological phases starting from analog to digital, wired to wireless and gradually evolved to VoIP technology which is rapidly gaining ground against traditional telephone technology. Though continuous advancement in communication technology had significantly improved the way of communication, however, voice quality is still restricted to nominal standards of narrow-band audio codecs. The adoption of VoIP has provided the opportunity to move beyond the set standards of PSTN “toll-quality voice” and get introduced to a new world of high-definition voice. High-definition Voice offers wide-band audio making communication more clear, effective and lively.

Matrix SPARSH VP is a High-definition VoIP Phone built with superior acoustics and elegant design to provide unsurpassed audio quality and rich user experience. The wide-band technology adds a new dimension to voice based communication.

Embrace the new and exciting world of High-definition IP telephony with Matrix SPARSH VP!



MATRIX
TELECOM SOLUTIONS

The SIP based VoIP phone can be used with existing IP infrastructure. It works as an intelligent terminal of IP-PBX, SIP Registrar or Proxy. SPARSH VP can also be used in a stand-alone mode in point-to-point and point-to-multi-point applications.

Making an outgoing call is as easy as it is from a normal telephone. Call progress tones like Dial Tone, Ring Back Tone and Busy Tone are fed to the caller as per the called number status. The user can make outgoing calls on one of the three SIP accounts. In addition, number based automatic SIP account selection is provided to select the most economical SIP account for a given outgoing number.

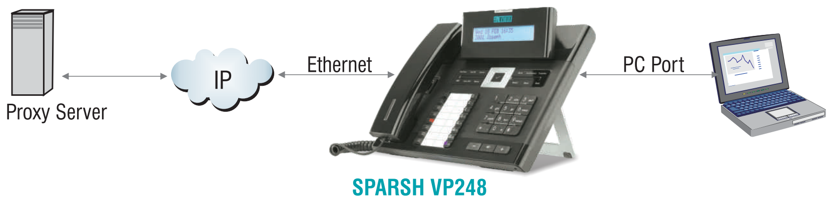
An incoming call from a SIP account can land on SPARSH VP. Multiple calls can be received and managed simultaneously.

Once a call is established, features like Call Hold, Call Toggle, Call Transfer, Call Wait and Conference are supported to manage multiple calls. Call Forward in different conditions and Do-Not-Disturb are also provided.

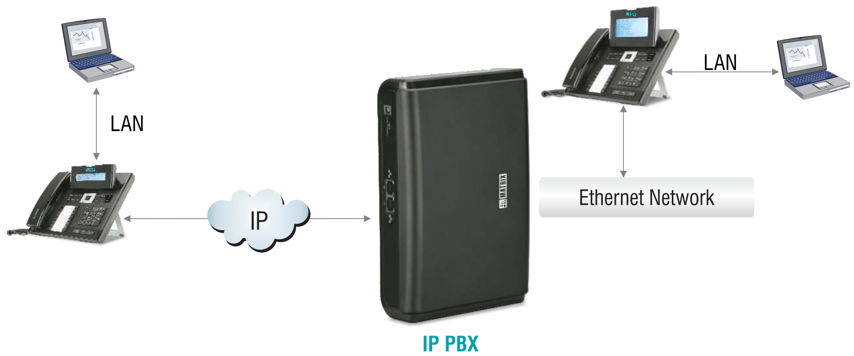
SPARSH VP is specially designed for busy executives to offer intuitive user experience, saving valuable time and thus increasing productivity. Full-duplex Speaker Phone, Headset Interface, Programmable Keys, Auto Answer, Speed Dial and many such useful features are supported.

Matrix SPARSH VP is easy to install and operate. It can be configured using its built-in web pages served by the internal HTTP server. Auto Configuration is also supported for service providers or IP-PBX to control multiple phones from a centralized server.

APPLICATIONS OF SPARSH VP248



Stand-alone Application

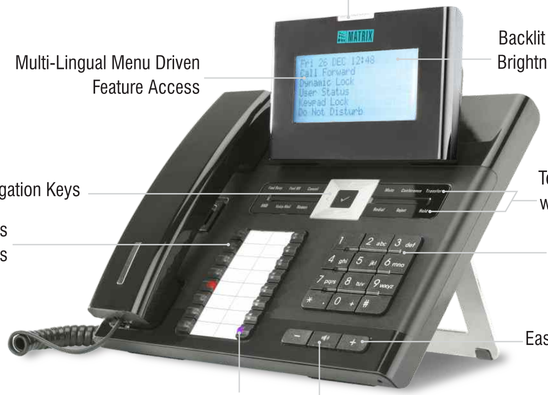


Business Application

MATRIX SPARSH VP248 PRODUCT RANGE

- SPARSH VP248S with 2 lines x 24 Characters LCD Display
- SPARSH VP248P with 6 lines x 24 Characters LCD Display
- SPARSH VP248SE with 2 lines x 24 Characters LCD Display and PoE
- SPARSH VP248PE with 6 lines x 24 Characters LCD Display and PoE

KEY FEATURES



3-Party Conference
This unique feature supports 3-party conference without affecting the speech levels. The user can also toggle between two remote parties or connect them and withdraw himself from the conference.

Adjustable LCD Display
The SPARSH VP's LCD display angle can be adjusted as per the user's convenience. The LCD's brightness and contrast can also be adjusted to improve readability.

Anonymous Call Rejection
SPARSH VP facilitates the user to reject incoming calls that do not have a Name or Number. Such calls are treated as unknown calls, which the user would not like to receive.

Auto Answer
SPARSH VP allows answering of the call without the user's intervention, on expiry of auto answer timer which is programmed. If any incoming call is received, SPARSH VP simulates the off-hook condition by activating the headset to answer the call automatically.

Auto Call Back
When the dialed number is busy, the user can apply the Auto Call Back feature so that his call gets in queue. When the called number is free, the user gets a call back notification on the LCD or LED. This helpful feature spares the user the effort of repeatedly dialing the same number, if it is supported by ITSP or IP-PBX.

Auto Configuration
SPARSH VP can be configured automatically from a central location. The configuration details like Registrar Server Address, Authentication User ID and User Password are stored in the central server. When the user connects SPARSH VP to the network, it automatically downloads its configuration using TFTP. This plug-n-play feature requires the user to enter only the server address provided by the service provide

Black Listed Callers' List
SPARSH VP allows the blocking of all incoming calls from specific addresses on SIP accounts, thus shielding users from unwanted junk calls.

Call Appearances
This intelligent feature gives the user a notification about 4 calls. While the user is attending a second call by putting the first call on hold, if more incoming calls land on the user's station, SPARSH VP gives indication for the third and the fourth call as a call waiting. The user can attend these calls by terminating any of the previous active calls or putting the previous call on hold.

Call Forward
Matrix SPARSH VP provides the user flexibility to forward his calls to a desired destination number. The calls can be forwarded in conditions like All Call, When Busy or When No Reply. The user can forward the calls on one number during the When Busy condition and to another number during a No Reply condition.

Call Waiting
This feature notifies the SPARSH VP user about another incoming call during an active call.

Call Hold
Matrix SPARSH VP allows the user to put the active call on hold and retrieve the second call which is in queue. The user can retrieve the first call again during the second call or on completion of second call. SPARSH VP notifies the user by indicating the call which is kept on hold.

Call Log
SPARSH VP is capable of keeping the log of 100 calls in its internal memory. 20 calls each of Dialed Calls, Answered Calls, Unanswered Calls, Rejected Calls and Forwarded Calls with details like Call Number, Called Party Name, Call Day-Date-Time and Call Duration are stored in SPARSH VP.

Call Mute
SPARSH VP allows the user to set a one-way speech and to avoid transmission of speech packets to the remote party.

Call Progress Tones and Rings
SPARSH VP provides Call Progress Tones like Dial Tone, Ring Back Tone and Busy Tone. Country-specific tones can be selected to match the tones of the country where it is installed. Similarly, ring cadences can also be selected.

Call Toggle
The user can handle up to four calls simultaneously, switching over from one active call to another held.

Call Transfer
The SPARSH VP allows the user to transfer the call after or without talking to a remote party to whom the call is being transferred. SPARSH VP supports two types of call transfers namely Unattended (Blind) Transfer and Attended Transfer.

Calling Options
Matrix SPARSH VP incorporates advanced calling functionality. The various options provided for making calls are - Using Handset, Using Speaker phone, Using Headset and Hot Pad Dialing. During an active call, the user can switch between handset, headset and speaker mode.

CLIP
CLIP based on information received from SIP message is supported by SPARSH VP.

CLIR
SPARSH VP offers the facility of calling line identity restriction. Selected users can deny disclosure of their identity to others.

Compact and Sturdy
Matrix SPARSH VP is an integrated IP Phone which can be installed on a wall or any table surface.

Day Light Saving
SPARSH VP synchronizes with the NTP server's clock timing to move forward or backward as per duration set in the system, automatically in tune with the Day Light Saving requirement of the country where it is installed. The options like Week-Day-Month or Date-Month are provided to move the clock forward and backward automatically on the specified day, date and time.

Dial Plan
SPARSH VP provides a list of 10 programmable numbers or part numbers with economical SIP account. When a user dials a number, it gets the matching entry from the list using "best-fit" logic and uses the SIP account given therein. This ensures lowest cost for all the outgoing calls.

Do Not Disturb Number List
This feature allows the user to receive calls from specific numbers only. Thus Matrix SPARSH VP provides the user privacy, shielding him from unwanted calls.

Hotline
if a particular number is dialed frequently, it can be programmed as a hotline number. SPARSH VP, waits for some time for the user to dial a number, if the user does not dial any number, SPARSH VP dials out the hotline number automatically. Option is also provided to set SPARSH VP to dial the number immediately without waiting for the user to dial the number.

Key Assignment
SPARSH VP allows the user to assign certain features to be accessed by the touch of a single key. The user can use specific feature by pressing the keys assigned to them. Multiple keys are provided to be programmed for a specific feature that is to be performed by them.

Keypad Lock
SPARSH VP can be prevented from unauthorized use by locking the keypad. The options for keypad lock are (i) Manual Locking and (ii) Auto Locking. The user password is required to unlock the keypad in both cases.

LDAP Client
Lightweight Directory Access Protocol (LDAP) is a widely implemented protocol for communication between directory client applications and directory servers about data in directories. SPARSH VP248 supports LDAP client (LDAP V3) which can be used to access and search the phone book managed in the centralized LDAP server.

Least Cost Routing
The SPARSH VP selects a port that offers the least cost for an outgoing call. It supports dialed number (Dial Plan) based LCR algorithm to select the most cost-effective route for making the call.

MAC Cloning
When replacing the existing hardware with another, you can simplify the installation process by copying the MAC address of your existing PC. In such a case, you do not need to delay the SPARSH p process by informing your service provider for newly installed equipment.

Multi-Lingual Menu driven Feature Access
SPARSH VP248 provides the flexibility to select language while accessing Menu features from Integrated LCD display. SPARSH VP248 supports languages like English(Default),French, German, Italian, Portuguese and Spanish.

Network Port Parameters
Considering various network conditions and application usage, SPARSH VP provides flexibility to program network parameters as per the requirement. Parameters like PPPoE, IP Address, DHCP, STUN, NAT, DNS, Gateway and Router's Public IP can be programmed through Web Jeeves or Phone.

Notifications
SPARSH VP has the facility to provide an indication to the user about specific operation or event occurred. In the event of a Call Transfer, Voice Mail, Call Forward and Auto Call Back, SPARSH VP displays the status on LCD Display. Also the notification for Voice Mail and Auto Call Back is provided by LED.

PPPoE
SPARSH VP supports PPPoE client and hence can be used with xDSL modems.

Password
The configuration of SPARSH VP is user specific and allows itself to be programmed by the user by providing two passwords: configuration password and user password. The configuration password is used for programming the SPARSH VP and the user password is used for unlocking the keypad.

Peer-to-Peer Calling
SPARSH VP can make and receive calls from other VoIP users without any Registrar or Proxy servers. Numbers and IP addresses can be assigned to other VoIP users to provide direct access across the network. SPARSH VP provides two dialing options for Peer-to-Peer calling: (i) Peer-to-Peer Number Dialing (ii) IP Address Dialing. Organizations having multiple locations like branch offices and factories can use this feature to provide direct dialing between these end-points.

Phone Book
Frequently used numbers with names can be programmed in the internal phone book with 100 entries. On using "Search Contact" functionality, SPARSH VP displays the entire phone book by names in alphabetical order on the LCD screen. The user can select the desired number and dial out the same.

Quality of Service (QoS)
QoS is the method of prioritization of voice packets over IP network. Matrix SPARSH VP supports TOS and Diff Serv to facilitate improved voice quality.

SIP Accounts
Three SIP accounts can be programmed and outgoing calls can be routed on a SIP accounts based on LCR, Dial Plan or Manual Selection by the user. Dynamic allocation of SIP account is also possible by using a dial plan.

STUN
This capability allows Matrix SPARSH VP to work behind asymmetrical NAT

Speed Dial
SPARSH VP facilitates the user to dial a number at the touch of a single key instead of dialing the entire number. Multiple numbers can be programmed as Speed Dial Numbers and each number can be assigned a specific key. When the user presses the speed dial key, the number is displayed on the LCD screen and then it will be dialed out.

Voice Mail
SPARSH VP can retrieve voice mail messages at the touch of a single key from the VMS server located at ITSP or IP-PBX. SPARSH VP notifies the user about new messages in the mailbox in the form of LED and text prompt (VM) on the LCD display.

Volume Setting
Matrix SPARSH VP allows the user to set receive and transmit gain to improve the quality of speech. The Speaker Volume, Handset Volume, Headset Volume and Ringer Volume, can be adjusted for better audio quality.

V-LAN Tagging
A Virtual Local Area Network (VLAN) may be defined as a group of LANs that have different physical connections, but which communicate as if they are connected on a single network segment. VLANs increase overall network performance by grouping users and resources that communicate most frequently with each other.

Web Jeeves
Matrix SPARSH VP incorporates built-in HTTP server and web pages for configuration parameters. This web based programming feature helps the user to configure SPARSH VP from any part of the world once it is connected with the IP network.

FEATURES

Software Features
- 100 Rel / PRACK (RFC 3262)
- 3-Party Conference
- 4-Call Appearance (VP248PE)
- Anonymous Call Rejection
- Auto Answer
- Auto Call Back
- Auto Configuration
- Black Listed Callers' List
- Busy Lamp Field (BLF)
- Call Appearances
- Call Log
- Call Mute
- Call Progress Tones and Rings
- Call Toggle
- Calling Options
- CLIP
- CLIR
- Comfort Noise Generation
- Compact and Sturdy
- Day Light Saving
- DHCP Client
- Dial Plan
- DND List
- Do-Not -Disturb (DND)
- Echo Cancellation
- Forward Error Correction (FEC)
- G.722 Wideband Audio Codec
- Hotline
- Key Assignment
- Keypad Lock
- LDAP Client
- Least Cost Routing
- LED Indications
- MAC Cloning
- Secondary DNS Server Redundancy
- Multiple Gateway
- Multi-Lingual Menu driven Feature Access
- Notifications
- Password Protection
- PCAP Trace
- Peer-to-Peer Calling
- Phone Book
- PPPoE
- Quality of Service (QoS)
- Receiving a Call (Handset, Headset Speaker)
- Rejecting a Call
- Remote Programming
- Search Contact
- Selective SIP line Access
- SIP Accounts
- SIP Over TCP
- Speed Dial
- STUN Support
- Syslog Client
- Symmetric RTP
- Supplementary Services
  - Call Forward On Busy
  - Call Forward On No Reply
  - Call Hold
  - Call Waiting
  - Call Transfer-Blind
  - Call Transfer-Attended
  - Conference 3-Party
  - Making Second Call
- VLAN Tagging
- Voice Activity Detection
- Voice Mail
- Volume Setting
- Web Jeeves

Hardware Features
- 1 LAN Port for Network Connectivity
- 1 PC (LAN) Port for PC Connectivity
- 1 Speaker Key (with LED Indication)
- Volume Adjustment Keys
- Programmable Keys
- 5 Navigation Keys
- Full Duplex Speaker Phone
- Head set interface
- Ringer LED
- Swivel Backlit LCD display
- Touch Sense Keys

TECHNICAL SPECIFICATIONS

VoIP

VoIP Protocols	: SIP v2, SDP, RTP, RFC 2833
Network Protocol	: IPv4,TCP, UDP DHCP, SNTP, STUN, HTTP, PPPoE
SIP	: 3 SIP Accounts, Out Bound Proxy Support, Main and Secondary DNS Server Support
NAT	: STUN and NAT Keep Alive
Voice Codecs	: G.711 A-Law, m-Law, G.723-5.3, G.723-6.3, G.726-16, G.726-24, G.726-32, G.726-40, G.729AB , G.722 Wideband
Call Progress Tones	: Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Waiting Tone
Voice	: Dynamic Jitter Buffer (Adaptive), Comfort Noise Generation and Voice Activity Detection
Quality of Service	: Layer 3 DiffServ and TOS
Data Network	: LAN Port (RJ45), 10/100 Base T with and without PoE, PC Port (RJ45), 10/100 Base T
Security	: Password Protected Configuration

Power Supply

Input	: 5VDC @2A through External Adaptor (90-265VAC, 47-63Hz) and Power-on-Ethernet (PoE)
Power Consumption	: 5W (Typical)
Connector	: DC Power Jack

Mechanical

Dimensions (WxHxD)	: 24x20x9.9 cm (9.4”x7.9”x3.9”) 24x23x9.9 cm (9.4”x9.1”x3.9”)
Unit Weight	: 1.18 Kgs (2.6 lbs) Approx.
Shipping Weight	: 1 Kgs (4.6lbs) Approx.
Material	: ABS Plastic HB Grade
Installation Mounting	: Wall Mount and Table-Top (30° and 50°)
Color	: Standard Black

Environmental

Operating Temperature	: 10˚C to + 50˚C (14˚F to + 122˚F)
Storage Temperature	: -40˚C to + 85˚C (-40˚F to + 185˚F)
Operating Humidity	: 5-95% RH (Non-Condensing)
Storage Humidity	: 0-95% RH (Non-Condensing) at 40˚C

COMPLIANCES

EMI/EMC

Conducted Emission	: CISPR 22 Class A
Radiated Emission	: CISPR 22 Class A
Harmonic Current Emission	: IEC 61000-3-2
Voltage Flicker	: IEC 61000-3-3
Electro-static Discharge	: IEC 61000-4-2
Radiated Susceptibility	: IEC 61000-4-3
Electrical Fast Transient	: IEC 61000-4-4
Surge	: IEC 61000-4-5
Conducted Immunity	: IEC 61000-4-6
Power Frequency Magnetic Field	: IEC 61000-4-8
Voltage Interruption & Dips	: IEC 61000-4-11

FCC

Conducted Emission	: FCC Part 15 Sub Part B Class A
Radiated Emission	: FCC Part 15 Sub Part B Class A

EC Directives

R&TTE 1999/5/EC
LVD 73/23/EEC
EMC 89/336EEC

Safety

IEC 60950-1	: 2001 First Edition

PRODUCT RANGE

ETERNITY IP-PBX	The IP-PBX with Universal Connectivity and Seamless Mobility
SAPEX	All-in-One Embedded IP-PBX Server
VYOM CCX	High-Density SIP Gateway
ETERNITY	The Universal Telephony Gateway
SETU VGFX	Multi-Port SIP based VoIP to GSM-FXO-FXS Gateway
SETU VFXTH	Multi-Port SIP based VoIP to FXO-FXS Gateway
SETU VFX	Multi-Port SIP based VoIP to FXS Gateway
SETU ATA211G	SIP based Analog Telephone Adaptor with 1 FXS, 1 GSM and 2 Ethernet Ports
SETU ATA211	SIP based Analog Telephone Adaptor with 1 FXO, 1 FXS and 2 Ethernet Ports
SETU ATA2S	SIP based Analog Telephone Adaptor with 2 FXS Ports and 2 Ethernet Ports
SETU ATA1S	SIP based Analog Telephone Adaptor with 1 FXS Port and 2 Ethernet Ports
SPARSH VP248PE	The High-definition IP-Phone with 6 Lines x 24 Characters LCD Display and PoE
SPARSH VP248SE	The High-definition IP-Phone with 2 Lines x 24 Characters LCD Display and PoE
SPARSH VP248P	The High-definition IP-Phone with 6 Lines x 24 Characters LCD Display
SPARSH VP248S	The High-definition IP-Phone with 2 Lines x 24 Characters LCD Display

ABOUT MATRIX

RoHS COMPLIANT

Established in 1991, Matrix is a leader in Telecom and Security solutions for modern businesses and enterprises. An innovative, technology driven and customer focused organization; Matrix is committed to keep pace with the revolutions in the telecom and security industries. With around 30% of its human resources dedicated to the development of new products, Matrix has launched cutting-edge products like IP-PBX, Universal Gateways, VoIP Gateways and Terminals, GSM Gateways, Access Control and Time-Attendance Systems, Video Surveillance System and Fire Alarm Systems. These solutions are feature-rich, reliable and conform to the international standards. Having global foot-prints in Asia, Europe, North America, South America and Africa through an extensive network of more than 500 channel partners, Matrix ensures that the products serve the needs of its customers faster and longer. Matrix has gained trust and admiration of customers representing the entire spectrum of industries. Matrix has won many international awards for its innovative products.

For further information contact:

MATRIX
TELECOM SOLUTIONS

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