SETU ATA2S

SIP VoIP Adaptor with Two FXS and One WAN Port

Internet Telephony offers intrinsic benefits of cost and flexibility. At the same time legacy telephony infrastructure and habits cannot be replaced overnight. People desire the best of both worlds - lower cost of VoIP and convenience of using existing telephony products and methods.

Matrix Setu ATA2S is designed to meet this requirement of converting VoIP network to traditional telephony interfaces and vice-versa. It handles all the complexities of VoIP technology internally and provides simple telephone interfaces to make and receive calls.

Let Matrix Setu ATA2S be your bridge to the new world of IP Telephony!

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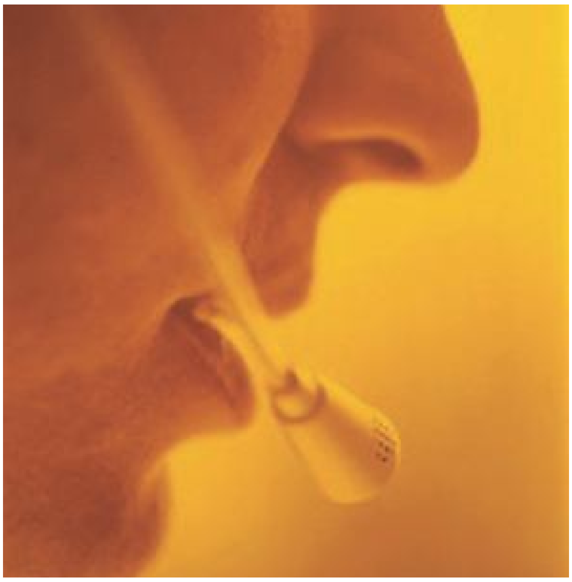

Matrix Setu ATA2S is a SIP based Analog Terminal Adaptor (ATA) with 2 FXS Ports and 1 WAN (Ethernet) Port. It interfaces legacy telephone devices with IP-based networks. It is specially designed for SOHO users to offer them the advantages of low-tariff Internet Telephony for long distance and international calls. It can be used with any existing PBX providing users access to VoIP trunks. It can also be used in a stand-alone mode.

Matrix Setu ATA2S converts the voice traffic into data packets for transmission over the Internet. When a telephone number is dialed by a user, Matrix Setu ATA2S converts it into an IP call using the SIP protocol and initiates a call to the dialed number in any part of the globe. Using an appropriate VoIP service provider, long distance or inter-office call charges can be reduced significantly or eliminated.

Making an outgoing call is as easy as 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 FXS ports can make outgoing calls on a common or two different SIP accounts. In addition, number based 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 be routed to any one or both FXS ports. All different CLIP protocols are supported so that the user can identify the caller before answering the call.

Once a call is established, features like Call Hold, Call Toggle, Call Transfer, Call Wait and Conference are supported to manage two calls from the same FXS port. The features Call Forward in different conditions and Do Not Disturb are also provided.

Matrix Setu ATA2S provides one Ethernet port for WAN or LAN connectivity. The user can connect xDSL modem or LAN to this port.

Matrix Setu ATA2S can also be used with any PBX without changing its existing infrastructure. PBX users can make voice calls on IP to availof the low-tariff of VoIP calls. The users can continue to make and receive calls without worrying on which network their calls are routed. Matrix Setu ATA2S is easy to install and operate. It can be configured using its built-in web pages served by the internal HTTP server.

Setu ATA2S

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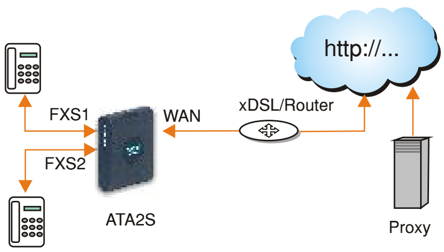
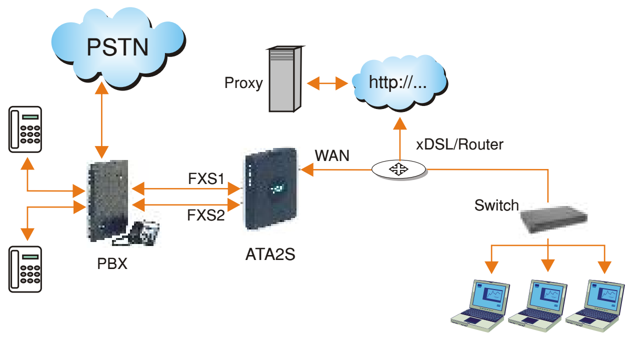
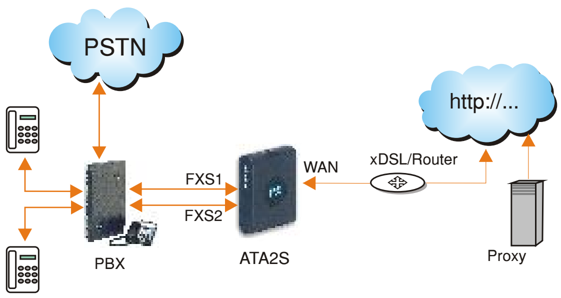
APPLICATIONS OF SETU ATA2S



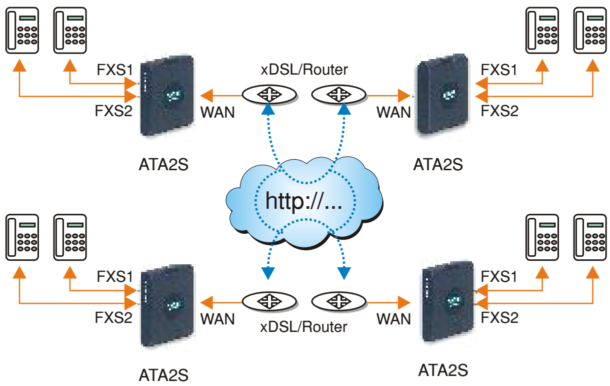
Setu ATA2S in Residential Application



Setu ATA2S in Business Application



Setu ATA2S in SOHO Application



Setu ATA2S in Peer-to-Peer Calling Application

KEY FEATURES

Auto Configuration

Setu ATA2S can be configured automatically from a central location. The configuration details like Registrar Server Address, Authentication User ID, User Password are stored in the central server. When the user connects Setu ATA2S 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 provider.

Calling Party Control (CPC)

CPC is required to prevent hanging of the FXS port when it is connected to a device like an answering machine, a voice mail system, etc. When a call is released from the other side of the Internet, the Matrix Setu ATA2S can propagate this call release on the FXS in the form of Calling Party Control (CPC) signal. The device senses this signal and frees the FXS port.

Call Progress Tones and Rings

Matrix Setu ATA2S supports programmable tones and rings to match those of the country where it is installed.

CLIP

Setu ATA2S allows users to program the FXS ports for any of the three CLIP protocols - DTMF, FSK ITU-T V.23 and FSK Bellcore 212A.

Compact and Sturdy

Matrix Setu ATA2S is an all-integrated equipment. It can be installed on a wall or any table surface.

Dial Plan

Matrix Setu ATA2S provides a list of 10 programmable numbers or part-numbers withthe preferred SIP account for each entry. When the user dials a number, the Setu ATA2S finds the matching number using the “best-fit” logic. It then uses the SIP account given against this matching number to make that call. This ensures lowest cost for all the outgoing calls.

Fax over IP (FoIP)

The Matrix Setu ATA2S user can send and receive Fax over SIP account, once the Fax machine is connected to its FXS port. The Setu ATA2S supports FoIP using T.38 vocoder and Pass Through technology.

Incoming Call Routing

A call arriving from any SIP account can be routed to either one or both FXS ports.

Jeeves (Web Based Programming Tool)

A flexible and user-friendly windows based software, Jeeves, helps in programming the features through web browser. This web based programming feature helps users to configure the Setu ATA2S from any part of the world, once it is connected with the IP network.

Peer-to-Peer Calling

Setu ATA2S can make and receive calls from other VoIP users without any Registrar or Proxy server. Numbers and IP addresses can be assigned to the other VoIP users to provide direct access across the network. For Peer-to-Peer calling, Setu ATA2S provides two options - (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 can be programmed in the internal phone book with 99 entries. The user can dial these numbers by using short codes in place of the complete, long numbers.

PPPoE

Matrix Setu ATA2S supports PPPoE client and hence can be used with any xDSL modem.

Quality of Service (QoS)

Matrix Setu ATA2S supports TOS and DiffServe to facilitate improved voice quality.

SIP Accounts

Two SIP accounts can be programmed and each FXS user can be assigned one of the SIP accounts for outgoing calls. Dynamic allocation of the SIP accounts is also possible using the Dial Plan.

STUN

This capability allows Matrix Setu ATA2S to work behind asymmetrical NAT.

Speech Volume Setting

Setu ATA2S allows users to set the transmit and receive gain to improve the quality of speech.

Supplementary Services

Setu ATA2S supports supplementary services like Call Hold, Call Waiting, Call Toggle, Call Transfer, Call Forward, Conference, Caller ID, DND and Making Another Call. These are the Service Provider dependent features.

Surface Mount Technology (SMT)

The Surface Mount Technology is the current semi-conductor packaging technology that offers reduction in real estate resulting in less heat generation and low power consumption. This is in turn improves reliability.

FEATURES

- Auto Configuration*
- Calling Party Control (CPC)
- CLIP (DTMF, FSK-ITU-T V.23, Bellcore 212A)
- CLIP to Caller
- Comfort Noise Generation
- DHCP Client
- Dial Plan
- Echo Cancellation (Programmable Tail Length- 8/16/32ms)
- Fax over IP-T.38 and Pass Through
- Flash Time (Programmable from 100-900ms)
- Flexible Incoming Call Routing
- Forward Error Correction (FEC)
- Full Duplex Audio
- LED Indications
- Password Protection
- Peer-to-Peer Calling
- Phone Book
- Polarity Reversal
- PPPoE
- Programmable Call Progress Tones and Rings
- Remote Programming
- Speech Volume Setting (Transmit and receive)
- Supplementary Services*
    - Call Forward On Busy*
    - Call Forward On No Reply*
    - Call Forward Unconditionally*
    - Call Hold*
    - Call Toggle*
    - Call Waiting*
    - Caller ID*
    - Call Transfer-Blind*
    - Call Transfer-Attended*
    - Conference 3 Party*
    - Do Not Disturb (DND)*
    - Making Second Call*
- STUN
- Voice Activity Detection

Features marked '*' are dependent on the Service Provider.

TECHNICAL SPECIFICATIONS

VoIP	
VoIP Protocols	: SIP v2, SDP, RTP, RFC 2833
Network Protocol	: IPv4, TCP, UDP, DHCP, SNTP, STUN, HTTP, PPPoE
SIP	: 2 SIP Accounts Out Bound Proxy Support Display Name, User Name, Password, URL, Proxy URL, Registrar URL, Registrar Interval
NAT	: STUN and NAT Keep Alive
Voice CODECS	: G.711 A-Law, μ-Law, G.723, G.729A, G.729B
Line Echo Cancellation	: G.168 with 8/16/32ms Tail Length
Call Progress Tones	: Dial Tone, Ring Back Tone, Busy Tone, Error Tone
Voice	: Dynamic Jitter Buffer (Adaptive), Comfort Noise Generation and Voice Activity Detection
Fax	: T.38 and Pass Through
Quality of Service	: Layer 3 DIFFServ and TOS
Data Network	: WAN Port (RJ45), Auto MDIX 10/100 BaseT
Security	: Password Protected Administration
FXS (SLT) Port	
Connection	: 2 nos. (RJ11)
Off Hook Impedance	: 600 Ω
Loop Limit	: 270 Ω (Max) Excluding telephone set
Loop Feed	: 39mA (Max)
Ringing Voltage	: 55Vrms @25Hz, 3REN
Pulse Dialing	: 10 PPS and 20PPS @ 1:2, 2:3 and 1:1
DTMF Dialing and Reception	: ITUT Q.23 and Q.24
Caller ID Presentation (CLIP)	: DTMF, FSK ITU-T V.23 and FSK Bellcore 212A
Call Maturity	: Polarity Reversal
Protection	: Solid state (Over Voltage and Over Current) built-in Secondary Protection
LED Indications	: 1 LED for Power 1 LED for each FXS port 1 LED for each SIP Account
Power Supply	
Input	: 12VDC @1.25A through External Adaptor (90-265VAC, 47-63Hz)
Power Consumption	: 5W (Typical)
Connector	: DC Power Jack
Mechanical	
Dimensions (WxHxD)	: 7.9x10.5x2.7cm (3.1”x4.1”x1.1”)
Unit Weight	: 0.45Kgs (1.10lbs) Approx.
Shipping Weight	: 1.00Kgs (2.20lbs) Approx.
Material	: ABS Plastic
Installation Mounting	: Wall and Table-Top
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

SYSTEM CAPACITY AND RESOURCES

Hardware	Application	No. of Ports	Connection
Maximum FXS Ports	Analog Phone Connectivity	02	RJ11
WAN Port	External Network Connectivity like xDSL or Router	01	RJ45 (10/100 BaseT)
SIP Account	To Call using IP Network	02	Through WAN Port
DC Jack	To connect Power Supply Adaptor	01	DC Power Jack

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

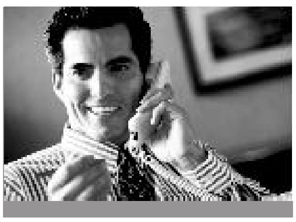
Safety

IEC 60950 3rd Edition (1999)

Setu ATA2S

VoIP PRODUCTS FROM MATRIX

Setu ATA2S	SIP based Analog Telephone Adaptor with 2-SIP Accounts, 2-FXS Ports and 1-WAN Port
Setu ATA2LL	SIP based Analog Telephone Adaptor with 2-SIP Accounts, 2-FXS Ports, 1-WAN Port, 1- LAN Port and 1-FXO (Lifeline) Port
Setu VFX88L Port,	SIP based VoIP gateway with 9-SIP Accounts, 8-Voice Channels, 8 FXS 1-WAN Port and 1-FXO (Lifeline) Port
Setu VGD10S	Integrated VoIP-GSM-PRI/T1/E1 Universal Gateway with 32-SIP Accounts, 120-VoIP Channels, 80-GSM Ports, 8-T1/E1/PRI Ports and 1-WAN Port
Setu VGD16S	Integrated VoIP-GSM-PRI/T1/E1 Universal Gateway with 32-SIP Accounts, 192-VoIP Channels, 128-GSM Ports, 8-T1/E1/PRI Ports and 1-WAN Port
Setu VP236S `	Executive IP-Phone with 2-SIP Accounts, 1-WAN Port and 24 Programmable Keys and 2 Lines x 24 Characters Backlit LCD Display
Setu VP236P `	Executive IP-Phone with 2-SIP Accounts, 1-WAN Port and 24 Programmable Keys and 6 Lines x 24 Characters Backlit LCD Display



ABOUT MATRIX

An ISO 9001 Company, Matrix is a leader in the VoIP, GSM, Key Phone System and PBX market. An innovative, technology driven and customer focused organization; the company is committed to keep pace with revolutions in the telecom industry. This has resulted in bringing forth cutting edge products like Digital and ISDN Key Phone Systems, Voice Messaging Products, GSM Gateways, VoIP Gateways, VoIP PBXs, Intercom Security Products and PLCC Switches. With over 1,000,000 line units installed and growing by over 1000 line units per day, the installed base of Matrix connects over 10,000,000 calls everyday. Thus, Matrix has gained the trust and admiration of users representing the entire spectrum of industries. Matrix has won many awards for its innovative products.

For further information contact:

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Due to continuous technology upgradations, product specifications are subject to change without notice.