SETU ATA

VoIP Adaptors with FXO, FXS, GSM and Multiple SIP Accounts

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 ATA range of products 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. It is specially designed for small business and residential users to offer them the advantages of low tariff Internet telephony.

Let Matrix SETU ATA be your bridge to the new world of IP Telephony!

MATRIX
TELECOM SOLUTIONS

VoIP ATAs with Unique Integration of Technology Features and Reliability

Matrix SETU ATA is a family of SIP based Analog Terminal Adaptors (ATA). It interfaces legacy telephone devices with IP-based networks. In addition to the standard connectivity of IP and PSTN, it also offers connectivity between IP, GSM and PSTN networks. An ATA provides a user with the facility of using a standard telephone instrument yet make VoIP and GSM calls.

While routing calls over IP network, the ATA converts voice traffic into data packets for transmission over the Internet. When a telephone number is dialed by a user, Matrix SETU ATA 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 through peer-to-peer calling on the IP network. The ATA significantly saves on communication cost between multi-branch locations, relaying calls over IP and using the local switched network to place a call on a specific network.

Making an outgoing call is as easy as from a normal telephone. The ATA automatically translates the dialed number, matching to the format that can be understood by the destination (IP or GSM or PSTN) network. The users can continue to make and receive calls without worrying on which network their calls are routed. Call progress tones like Dial Tone, Ring Back Tone and Busy Tone are fed to the caller as per the called number status. The ATA ensures a call is always routed through the most cost-effective network. In addition, number based SIP account selection is provided to select the most economical SIP account for a given outgoing number.

An incoming call on GSM, FXO or any of the SIP accounts can be received on an existing terminal easily. The calls landing on the GSM port can be routed through a SIP account. Incoming calls on the FXO line can be routed through any of the SIP accounts, to avail the low tariff of IP telephony. While, an incoming call on a SIP account can be routed through the GSM port, for the benefits of cost and QoS (Quality of Service).

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. Call forward in different conditions and Do Not Disturb is also provided. All different CLIP protocols are supported so that the user can identify the caller before answering the call. Matrix SETU ATA provides two Ethernet ports - one for WAN and the other for LAN. The user can connect his PC on the LAN port and browse the internet or check his emails while talking on VoIP calls.

Matrix SETU ATA is easy to install and operate. It can be configured using its built-in web pages served by the internal HTTP server.

SETU ATA1S
1-FXS Port, 1-WAN Port,
1-LAN Port and 2-SIP Accounts

SETU ATA2S
2-FXS Ports, 1-WAN Port and
2-SIP Accounts

SETU ATA211
1-FXO Port, 1-FXS Port, 1-WAN Port,
1-LAN Port and 3-SIP Accounts

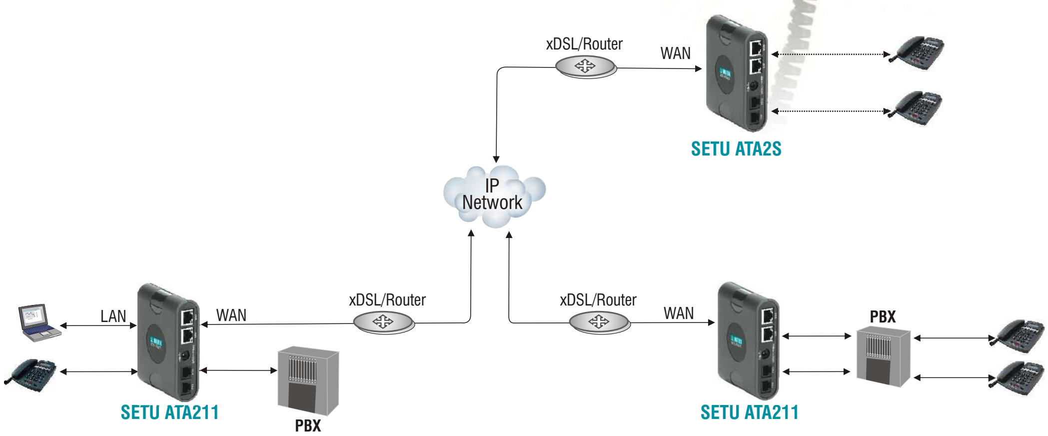
SETU ATA211G
1-FXS Port, 1-GSM Port, 1-WAN Port,
1-LAN Port and 3-SIP Accounts

PRODUCT APPLICATIONS

RESIDENTIAL APPLICATION

BUSINESS APPLICATION

PEER-TO-PEER CALLING APPLICATION



KEY FEATURES

Auto Configuration
SETU ATA 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 user connects SETU ATA 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.

Auto PSTN Fall back
PSTN can be interfaced to the SETU ATA using FXO Port. This port is used to dial out numbers to the PSTN Network. When Routing the calls from PSTN number to SIP trunk, it may happen that the Ethernet Link may go down or the SIP Account used is not registered. So the call will not be routed through SIP and you will get error tone. To avoid this you can use this feature to automatically route the call through the FXO Port.

Automatic Number Translation
SETU ATA supports multiple port types like FXO, FXS, SIP and GSM. Whenever a number is dialed from any of these ports, gateway routes the call to the desired destination port as per the routing mechanism defined for that port. In certain cases, the dialed number string is not understood by the network through which the call is to be routed, so by using Automatic Number Translation the dialed number string is translated into a number that is understood by the network or ITSP to reach the desired destination port.

Call Progress Tones and Rings
Matrix SETU ATA supports programmable tones and rings to match those of the country where it is installed.

CLIP
SETU ATA allows users to program the FXS ports for any of the three CLIP protocols - DTMF, FSK ITU-T V.23 and FSK Bellcore 202A.

Dialed Number Table
Matrix SETU ATA provides a list of programmable numbers or part-numbers with the preferred SIP account for each entry. When the user dials a number, the SETU ATA 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.

Disconnect Signaling
When a call is released from the other side of the network, the Matrix SETU ATA can propagate this call release on the FXS in the form of disconnect signal. The device senses this signal and frees the FXS port.

Fax over IP (FoIP)
The user can send and receive Fax over SIP account, once a Fax machine is connected to SETU ATA. The SETU ATA supports FoIP using T.38 UDPTL and Pass-Through.

FXO
SETU ATA FXO port should be connected to the PSTN or PBX so that the user can make PSTN calls from the FXO port.

International Mobile Equipment Identity (IMEI) Number
International Mobile Equipment Identity (IMEI) number provided on SETU ATA211G GSM engine is a unique 15 digit code to identify the GSM port. This number can be used to associate and tie the equipment with a particular GSM network.

Jeeves (Web based Programming Tool)
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 ATA from any part of the world once it is connected with the IP network.

MAC Cloning
When replacing the existing hardware with a new one, one can simplify the installation process by simply copying the WAN associated MAC Address to the new hardware. Thereby, eliminate the delay involved in the setup process by informing the service provider of the newly installed equipment.

Network Selection
SETU ATA211G provides flexibility to register with a GSM network manually or automatically. This is useful when the installation is close to a state or national border where local and foreign GSM networks overlap. Programming the ATA to work only with selected network prevents it from registering with an overlapping foreign network.

Multi-Stage Dialing
Multi-Stage Dialing is useful for ATAs connected to a SIP Server used for networking PBX of multiple sites. The user can dial the entire number string, both the destination number and extension number of the destination PBX together. The ATA will split the string into two stages, and dial out the destination number first and on receiving the answering signal it dials the extension number. This ensures hassle free access to PBX extension.

Peer-to-Peer Calling
SETU ATA 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 ATA 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 the dispersed end-points.

PIN Authentication
Incoming calls on FXS, FXO, SIP or GSM ports of the ATA can be restricted to a specific caller. The caller has to first prove his authentication before calling to ATA. This feature is used to avoid the possibility of malicious calls and to avoid misuse of its services.

PPPoE
Matrix SETU ATA supports PPPoE client and hence can be used with any xDSL modems.

Quad-Band Support
Matrix SETU ATA211G supports Quad-Band for 2G Network

Returned Calls to Original Caller (RCOC)
Matrix SETU ATA211G maintains records of all the unsuccessful calls on GSM and IP network. When a call is returned, it routes the call to the original caller comparing the called number, caller's number and the system port details to the entries stored in its RCOC table. Thereby a returned call can be landed to the extension which placed the call, hence, saving valuable time.

Router
Basic routing capabilities are provided so that LAN port packets can be transferred on WAN port. This allows the user to browse the internet and check his emails while making and receiving VoIP calls.

Signal Strength Indication
SETU ATA211G gives the indication of signal strength available for communication. Thus the possibility of network availability can be found. The signal strength indication is shown on the LCD of the telephone instrument supporting FSK CLIP.

SIM PIN
SETU ATA211G allows user to program 4 digits PIN number (Personal Identification Number) which prevents the SIM against unauthorized use. User has to enter PIN for making any operations. User can change the SIM PIN as and when required. SIM gets blocked if PIN is entered wrongly thrice in a row.

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

Speech Volume Setting
SETU ATA allows user to set transmit and receive gain to improve the quality of speech.

Speed Dialing
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.

Supplementary Services
SETU ATA supports supplementary service like Call Hold, Call Waiting, Call Toggle, Call Transfer, Call Forward, Conference, Caller ID, DND and Making another Call. These are the service provider dependant features.

Surface Mount Technology (SMT)
The Surface Mount Technology is the current semi-conductor packaging technology resulting in less heat generation and low power consumption. This is in turn improves reliability.

System Log
Syslog is one of the Built-in protocol used extensively for sending debug messages on IP network. This client/server protocol uses UDP as transport protocol for debugging process. Logging has several benefits which include easier and faster troubleshooting, improved security and a better system administration. Debug messages are sent to remote server on IP network for finding and reducing the number of bugs or defects from a system.

FEATURES LIST

- 100Rel/PRACK (RFC3262)
- Answer Signaling
- Auto Configuration
- Auto PSTN Fallback
- Automatic Number Translation
- Call Forward Unconditionally
- Call Forward on Busy
- Call Forward on No Reply
- Call Hold
- Call Toggle
- Call Waiting
- Caller ID
- Call Transfer-Blind
- Call Transfer-Attended
- Conference 3-Party
- Called Party Number Table
- CLIP (DTMF, FSK-ITU-T V.23, Bellcore 202A)
- CLIR (SIP/ GSM)
- Comfort Noise Generation
- DHCP Client/ PPPoE Client
- Dialed Number Table
- Digest Authentication
- Disconnect Signaling
- Do-Not-Disturb (DND)
- Echo Cancellation
- Fax over IP-T.38 and Pass-Through
- Flash Timer
- Forward Error Correction (FEC)
- Full Duplex Audio
- IMEI Number
- Incoming Call Routing
- LED Indications
- MAC Cloning
- Making Second Call
- Mobile Network Selection
- Multiple Gateway Support
- Multi-Stage Dialing
- Password Protection
- Peer-to-Peer Calling
- PIN Authentication
- Polarity Reversal
- Programmable Call Progress Tones and Rings
- Quad-Band Support
- Return Call to Original Caller (RCOC)
- Signal Strength
- SIM PIN
- SIP over TCP
- Speech Volume Setting (Transmit and Receive)
- Speed Dialing
- STUN
- Supplementary Services
- Symmetric RTP
- Syslog Client
- Voice Activity Detection
- VLAN Tagging
- Web-based GUI for Configuration

MATRIX SETU ATA FEATURES LIST

Features List	SETU ATA1S	SETU ATA2S	SETU ATA211	SETU ATA211G
3-Party Conference	●	●	●	●
Answer Signaling	●	●	●	●
Auto Configuration	●	●	●	●
Automatic Number Translation	●	●	●	●
Auto PSTN Fall Back	–	–	●	–
Called Party Number Table	–	–	●	●
Call Progress Tone and Rings	●	●	●	●
Class of Service (CoS)	●	●	●	●
CLIP	●	●	●	●
CLIR	●	●	●	●
Comfort Noise Generation	●	●	●	●
Call Progress Tone and Rings	●	●	●	●
DHCP Client	●	●	●	●
Dial Plan	●	●	●	●
Digest Authentication	●	●	●	●
Disconnect Signaling	●	●	●	●
Echo Cancellation	●	●	●	●
Fax over IP- T.38 and Pass Through	●	●	●	●
Flash Timer	●	●	●	●
Forward Error Correction	●	●	●	●
Full Duplex Audio	●	●	●	●
IMEI Number	–	–	–	●
LED Indications	●	●	●	●
Multiple Gateway	●	●	●	●
Multi-Stage Dialing	●	●	●	●
Mobile Network Selection	–	–	–	●
Password Protection	●	●	●	●
PCAP Trace	●	●	●	●
Peer-to-Peer Calling	●	●	●	●
Phone Book	●	●	●	●
PIN Authentication	–	–	●	●
Polarity Reversal	●	●	●	●
PPPoE Client	●	●	●	●
Quad-Band Support	–	–	–	●
Return Call to Original Caller (RCOC)	–	–	–	●
Speech Volume Settings	●	●	●	●
Speed Dialing	●	●	●	●
Signal Strength	–	–	–	●
SIM PIN	–	–	–	●
Supplementary Services	●	●	●	●
STUN Support	●	●	●	●
Symmetric RTP	●	●	●	●
Syslog Client	●	●	●	●
VLAN Tagging	●	●	●	●
Web based GUI for Configuration	●	●	●	●

– Unavailable ● Available

SYSTEM CAPACITY AND RESOURCES

Hardware	SETU ATA1S	SETU ATA2S	SETU ATA211	SETU ATA211G
FXS Ports	1	2	1	1
FXO Ports	–	–	1	–
GSM Ports	–	–	–	1
LAN Port	1	–	1	1
WAN Port	1	1	1	1
DC Power Jack	1	1	1	1

TECHNICAL SPECIFICATIONS

VoIP

VoIP Protocols	: SIP v2, SDP, RTP, RFC 2833
Network Protocol	: IPv4, TCP, UDP, DHCP, SNTP, STUN, HTTP, PPPoE
SIP	: Multiple Accounts Out BoundProxy 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, LAN Port (RJ45), Auto MDIX 10/100 BaseT
Security	: Password Protected Administration

FXS Port

Connection	: 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 202A
Call Maturity	: Polarity Reversal
Protection	: Solid State (Over Voltage and Over Current) built-in Secondary Protection

FXO Port

Connection	: RJ11
Off Hook Line Impedance	: 600 Ω
Loop Limit	: 1500 Ω
Pulse Dialling	: 10 PPS and 20 PPS @ 1:2, 2:3 and 1:1
DTMF Dialling and Reception	: ITU-T Q.23 and Q.24
CLI Reception	: DTMF, FSK ITU-T V.23 and FSK Bellcore 202A
Call Maturity	: Polarity Reversal
Protection	: Solid state (Over Voltage and over current) built-in secondary Protection

GSM Port

GSM Band	: Quad-Band: GSM850, EGSM900, DCS1800, PCS1900
SIM Card:	: One SIM
SIM Interface	: 1.8V, 3V
Transmission Power	: Class 4 (2W) at GSM 850 and EGSM900 MHz band Class 1(1W) at DCS1800 and PCS1900 MHz band
RF Sensitivity	: Better than -102dBm at GSM850/EGSM900/DCS1800/ PCS1900
External Antenna	: Gain: Dipole= 2.5 dBi
Type of Antenna	: Dipole/Whip Fixed/Omni Directional Antenna Roof- Top Antenna with flexible cable of 3 mtrs.(optional) Antenna Connector: SMA (Female), 50Ω Impedance
Speech Gain (Transmit- Receive)	: Programmable

Power Supply

SETU ATA1S	: Input- 12VDC @1.25A through External Adaptor (90-265VAC, 47-63Hz) Connector- DC Power Jack
SETU ATA2S	: Input- 12VDC @1.25A through External Adaptor (90-265VAC, 47-63Hz)Connector- DC Power Jack
SETU ATA211	: Input- 12VDC @1.25A through External Adaptor (90-265VAC, 47-63Hz) Connector- DC Power Jack
SETU ATA211G	: Input- 12VDC @ 0.75A through External Adaptor (90-265VAC, 47-63Hz)Connector- DC Power Jack

Power Consumption

SETU ATA1S	: 5W Approx.
SETU ATA2S	: 5W Approx.
SETU ATA211	: 5W Approx.
SETU ATA211G	: 6W Approx.

Mechanical

Dimensions (WxHxD)

SETU ATA1S	: 7.9x10.5x2.7cm (3.1"x4.1"x1.1")
SETU ATA2S	: 7.9x10.5x2.7cm (3.1"x4.1"x1.1")
SETU ATA211	: 7.9x10.5x2.7cm (3.1"x4.1"x1.1")
SETU ATA211G	: 7.9x10.5x2.7cm (3.1"x4.1"x1.1")

Unit Weight

SETU ATA1S	: 0.45Kgs (1.10lbs) Approx.
SETU ATA2S	: 0.45Kgs (1.10lbs) Approx.
SETU ATA211	: 0.45Kgs (1.10lbs) Approx.
SETU ATA211G	: 0.45Kgs (1.10lbs) Approx.

Shipping Weight

SETU ATA1S	: 1.00Kgs (2.20lbs) Approx.
SETU ATA2S	: 1.00Kgs (2.20lbs) Approx.
SETU ATA211	: 1.00Kgs (2.20lbs) Approx.
SETU ATA211G	: 1.00Kgs (2.20lbs) Approx.

Material

SETU ATA1S	: ABS Plastic
SETU ATA2S	: ABS Plastic
SETU ATA211	: ABS Plastic
SETU ATA211G	: ABS Plastic

Installation Mounting

SETU ATA1S	: Wall and Table-Top
SETU ATA2S	: Wall and Table-Top
SETU ATA211	: Wall and Table-Top
SETU ATA211G	: 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

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

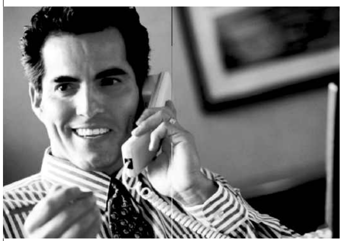
R&TTE 1999/5/EC
LVD 73/23/EEC
EMC 89/336EEC

Safety

IEC 60950-1	: 2001 First Edition

VoIP PRODUCTS FROM MATRIX

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
SETU VGFX	Multi-port SIP based VoIP to GSM-FXO-FXS Gateway
SETU VFXTH	Multi-Port VoIP to FXO-FXS Gateway
SETU VFX	SIP based VoIP Gateway with 4/8 FXS Ports, 1 FXO (PSTN Pass-Through) and 1 Ethernet Port
SETU ATA211G	SIP based Port VoIP to GSM and FXS Gateway
SETU ATA211	SIP based Analog Telephone Adaptor with 1 FXO, 1 FXS Port and 2 Ethernet Ports
SETU ATA2S	SIP based Analog Telephone Adaptor with 2 FXS Ports and 1 Ethernet Port
SETU ATA1S	SIP based Analog Telephone Adaptor with 1 FXS Port and 2 Ethernet Ports
SETU VP248PE	Executive IP-Phone with 6 Lines x 24 Characters LCD Display and PoE
SETU VP248SE	Executive IP-Phone with 2 Lines x 24 Characters LCD Display and PoE
SETU VP236SE	Executive IP-Phone with 2 Lines x 24 Characters LCD Display and PoE
SETU VP236S	Executive IP-Phone with 2 Lines x 24 Characters LCD Display



ABOUT MATRIX

An ISO 9001 Company, Matrix is a leader in Telecom and Security solutions for modern businesses and enterprises. An innovative, technology driven and customer focused organization; the company 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 telecom products like IP-PBX, Universal Gateways, VoIP Gateways and Terminals, GSM Gateways, Access Control and Time-Attendance Systems 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 more than 150,000 customers representing the entire spectrum of industries. Matrix has won many awards for its innovative products.

For further information contact:

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TELECOM SOLUTIONS

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