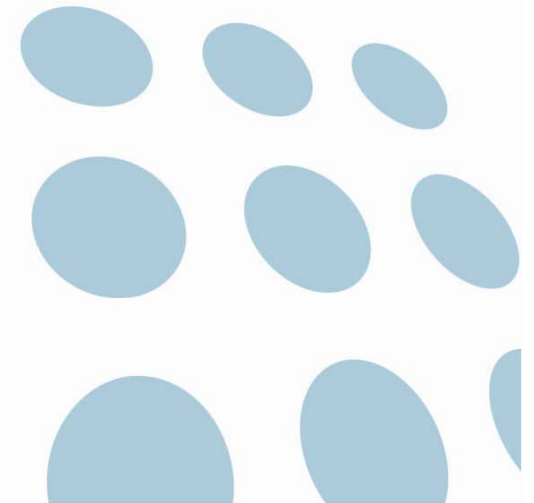
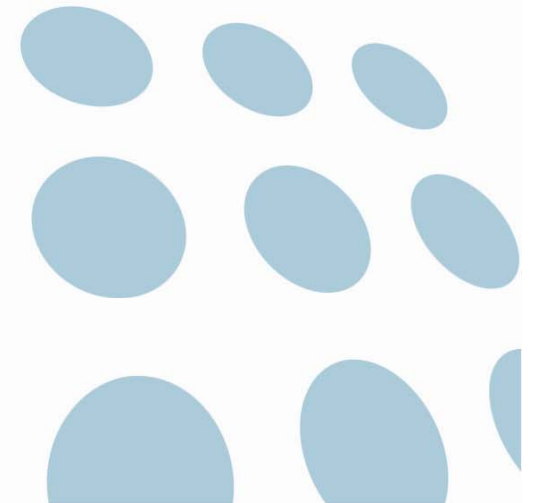


# Warm Welcome

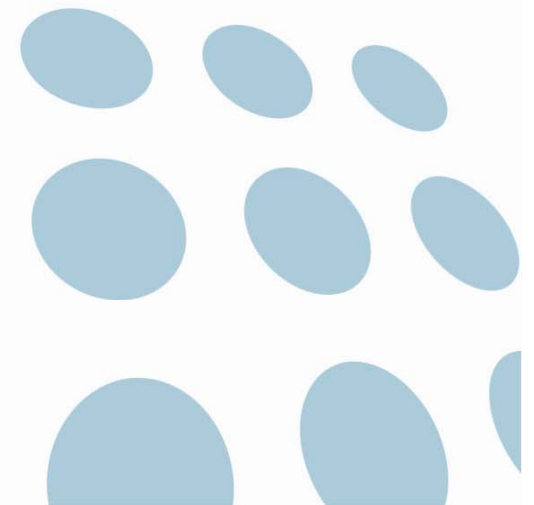


# Matrix SETU ATA211

VoIP Adaptors with FXO, FXS Ports and Multiple SIP  
Accounts



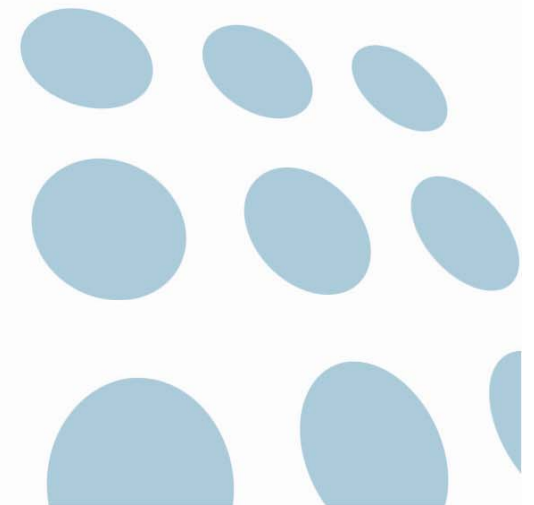
# Introduction





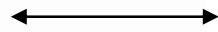
# Matrix SETU ATA Family

Model	Interfaces
SETU ATA1S	3 SIP Accounts, 1 FXS Port and 2 Ethernet Ports
SETU ATA2S	3 SIP Accounts, 2 FXS Ports and 2 Ethernet Ports
SETU ATA211	3 SIP Accounts, 1 FXO Port, 1 FXS Port and 2 Ethernet Ports
SETU ATA211G	3 SIP Accounts, 1 GSM Port , 1 FXS Port and 2 Ethernet Ports

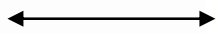


# SETU ATA211 Interfaces

FXO Port (RJ11)



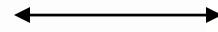
FXS Port (RJ11)



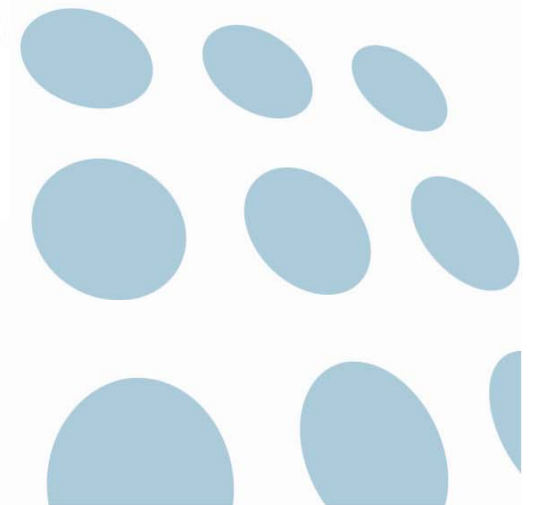
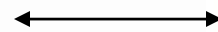
DC Input Jack



WAN Port (RJ45)



LAN Port (RJ45)

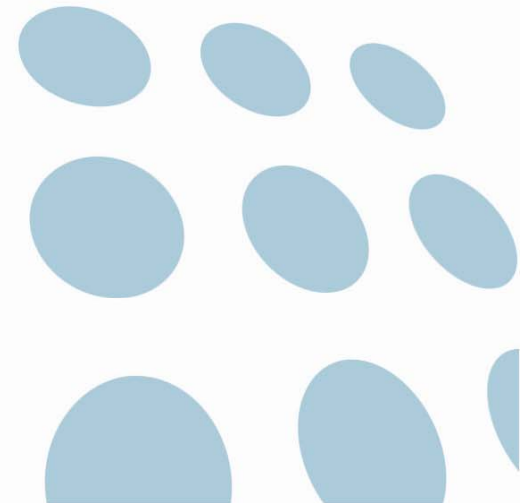


# Specifications

Description	SETU ATA211
FXS Port (RJ11)	1
FXO Port (RJ11)	1
Ethernet Port (RJ45)	2
DC Power Input Jack	1
VoIP Protocols	SIP v2
SIP Accounts	3 Accounts
Peer-to-Peer Table	500 Entries
Dial Plan	100 Entries
Phone Book	60 Entries
Fax over IP	T.38
Voice CODECS	G.723, G.729A, G.729B, PCM-A, PCM-μ

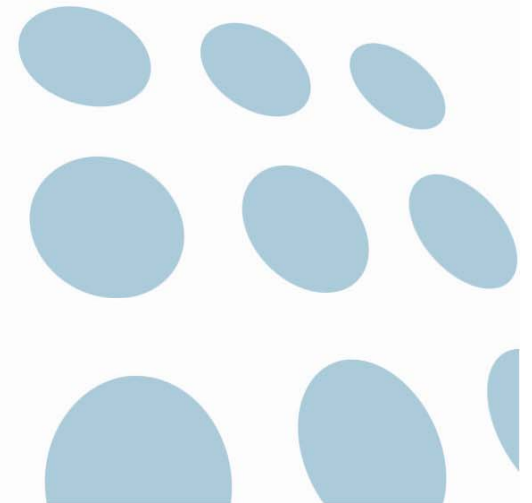
# Features

- 3-Party Conference
- Auto Configuration
- Automatic Number Translation
- Auto PSTN Fall Back
- Called Party Number Table
- CLIP
- CLIR
- Codec Selection: G.711, G.723.1, G.729AB
- DHCP Client



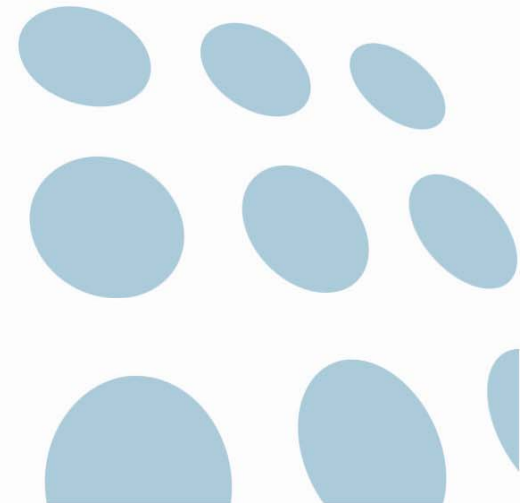
# Features

- Dial Plan
- Digest Authentication
- Echo Cancellation: G.168 up to 32ms
- Incoming Call Routing
- LED Indications
- MAC Cloning
- NAT and STUN
- Peer-to-Peer Calling



# Features

- Phone Book
- PIN Authentication
- PPPoE Client
- Programmable Call Progress Tones and Rings
- Syslog Client
- WAN and LAN pots-Surf and Talk Simultaneously

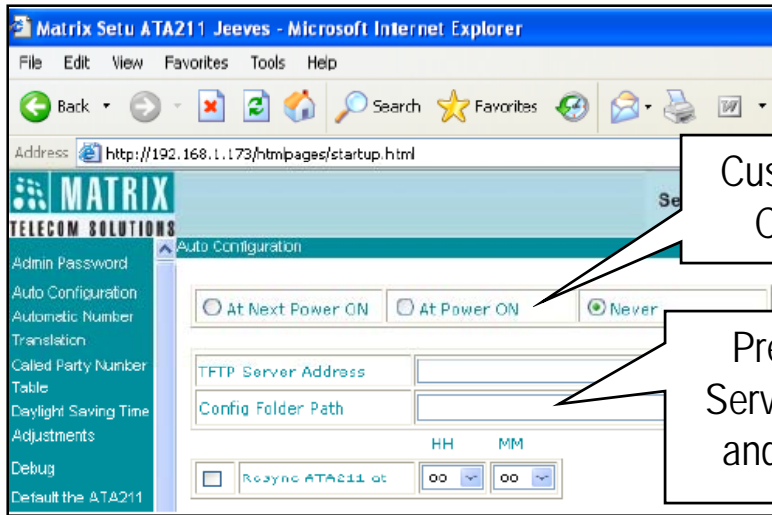


# Auto Configuration

- ATA's Need to be Configured for Certain Basic Parameters before Use
- ATA Users Mostly take an ITSP Connection to Make Calls
- It is Desirable to Configure ATA from a Central Location, Avoiding Technical Difficulties during Configuration Process
- Auto Configuration Allows ATA to be Configured from a Central Location (Service Provider Can Store the Configuration Files on a Central TFTP Server)
- ATA will Periodically Searches for Configuration Files on its Own



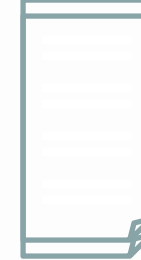
# Auto Configuration



Customer's Choice

Pre-Defined Server Address and File Path

Configuration File Downloaded via TFTP



TFTP Server

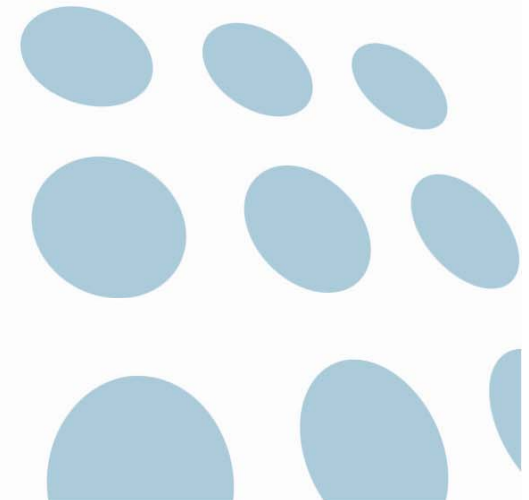
ITSP



Customer's Premises

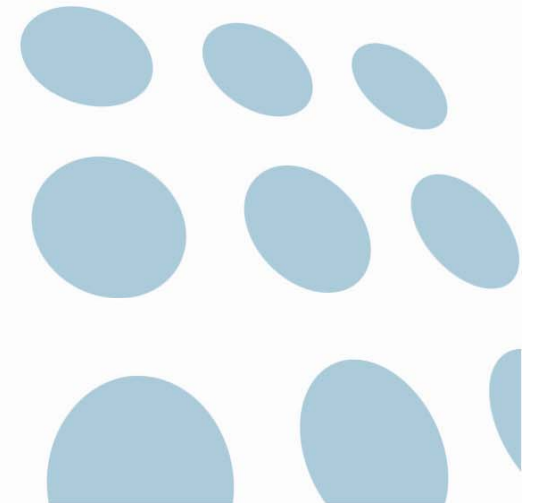


IP

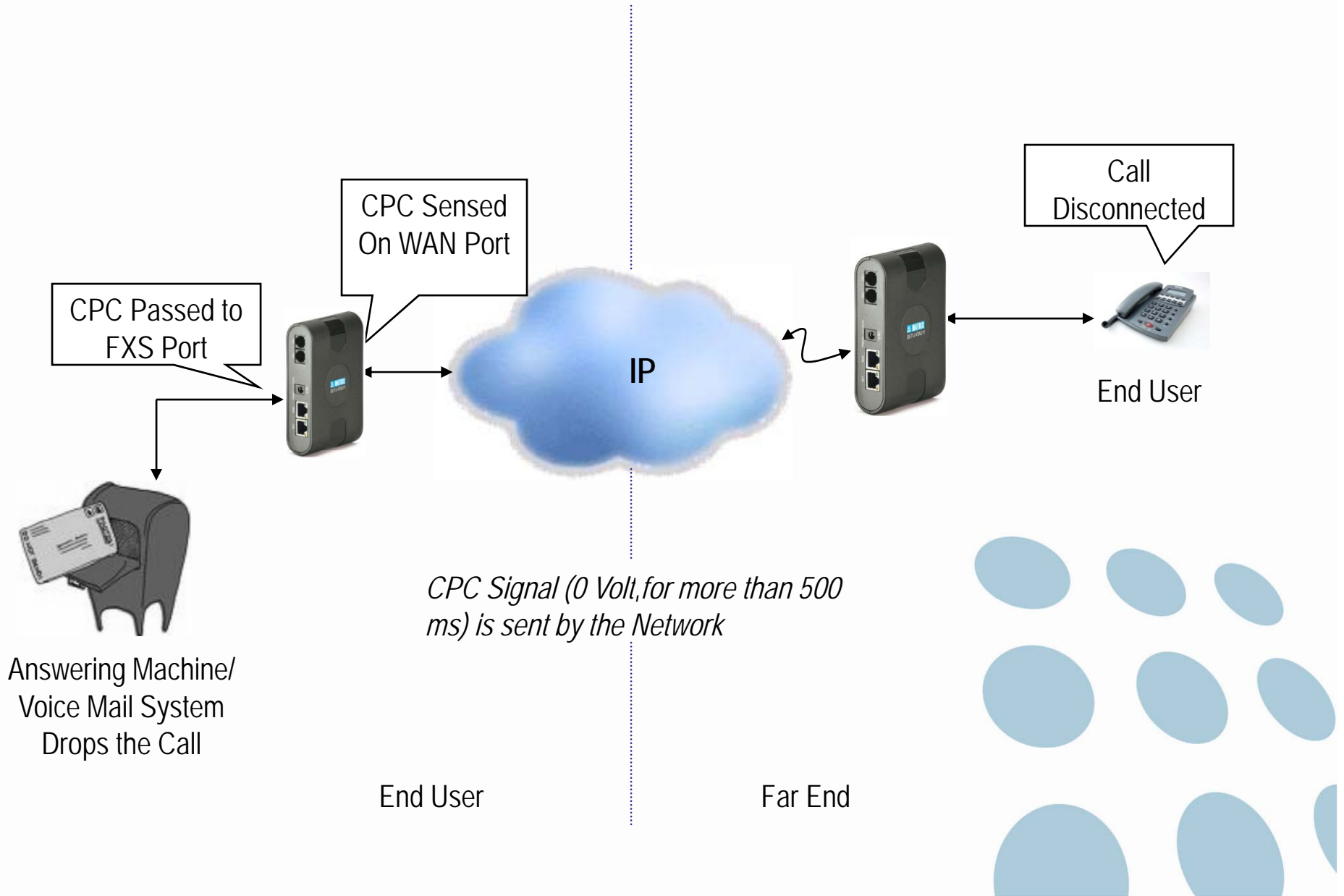


# Calling Party Control (CPC)

- The CPC signal Notifies to the ATA that
  - ✓ The Called Party has Hung-Up
  - ✓ Stop Recording to an Answering Machine or Voice Mail
  - ✓ Drop the Call
  - ✓ Prevents FXS Port from Hang-Up

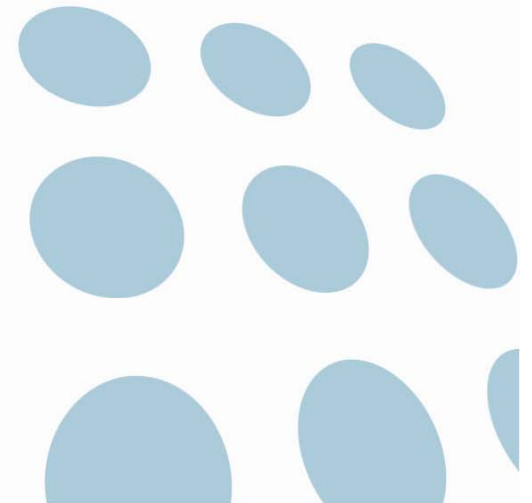


# Calling Party Control (CPC)



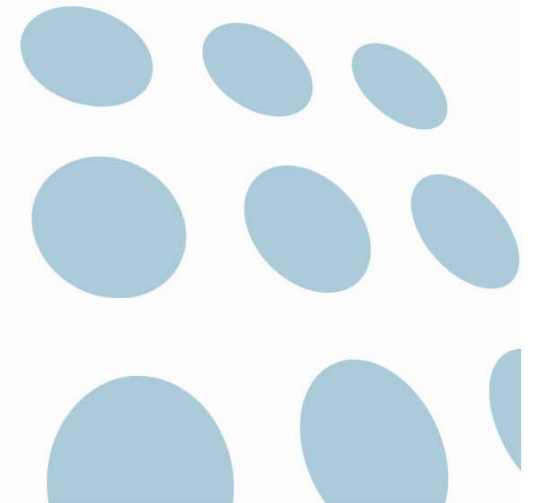
# Call Progress Tones and Rings

- Different Tones to Indicate the Progress of a Call
  - ✓ Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Feature Tone and Confirmation Tones, with Different Cadence, with Specific Frequency are Supported
- User can Select a Tone which Matches with the Tones Used in a Particular Region



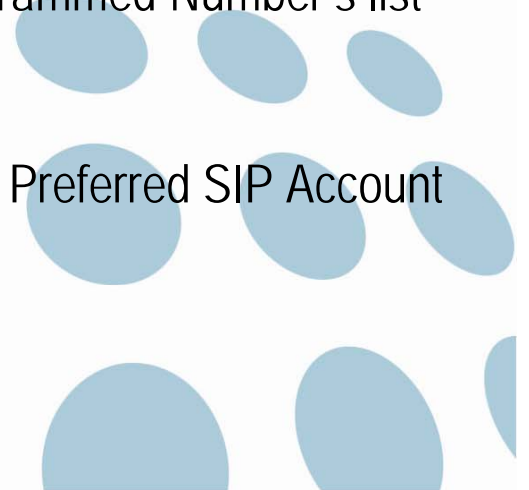
# CLIP

- Detection of DTMF and FSK ITU-T and FSK Bellcore
- CLIP for External Numbers, Internal Numbers
- FXS Port can be Programmed for CLIP Protocol
- This Feature can be Enabled / Disabled as Per Requirement



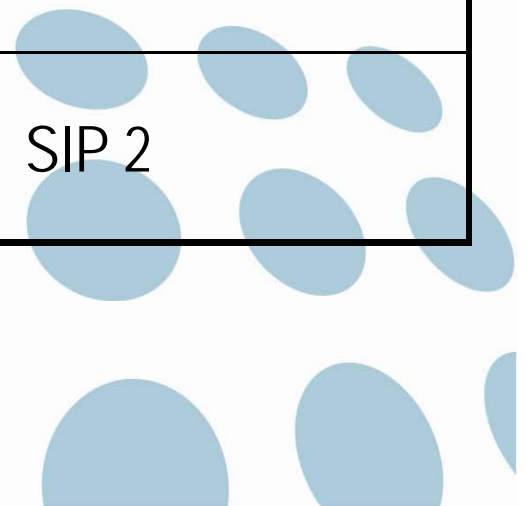
# Dial Plan

- Functions like LCR (Least Cost Routing)
- Numbers to be Dialed, are Pre-Programmed to Use the Most Economical SIP Account
- Basic Operation:
  - ✓ User Dials a Number
  - ✓ SETU ATA Searches for a Matching Number in the Programmed Number's list
  - ✓ A Number is then Dialed as Per the 'Best-Fit' Logic
  - ✓ 100 Numbers can be Programmed & Mapped along with Preferred SIP Account



# Dial Plan

Index	Number	SIP Account
01	001	SIP 1
02	0064	SIP 2
:	:	:
100	0061	SIP 2



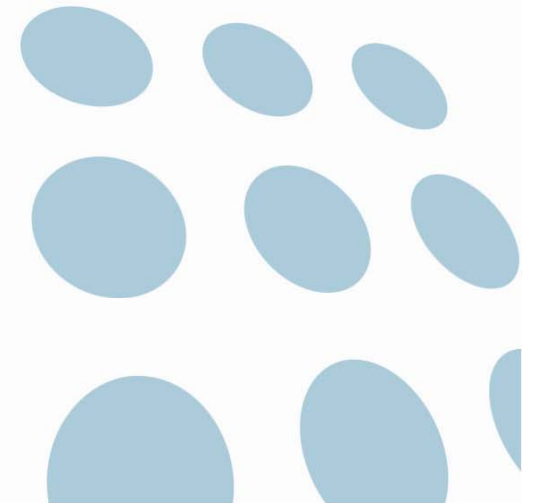
# Digest Authentication

- Feature Requirement
  - ✓ Industry Standard for Web Authentication
  - ✓ To Receive Calls from Specific Callers Only
- Basic Operation
  - ✓ Call is Received from Specific IP Device
  - ✓ Digest Authentication Transmits Credentials Across the Network as Message  
Digest to Authenticate the Caller
  - ✓ User is Authenticated for his User Name and Password
  - ✓ If Successful, Call is Allowed
  - ✓ Program such 100 Entries in "Digest Authentication Table"
  - ✓ Applicable to Incoming SIP Calls Only



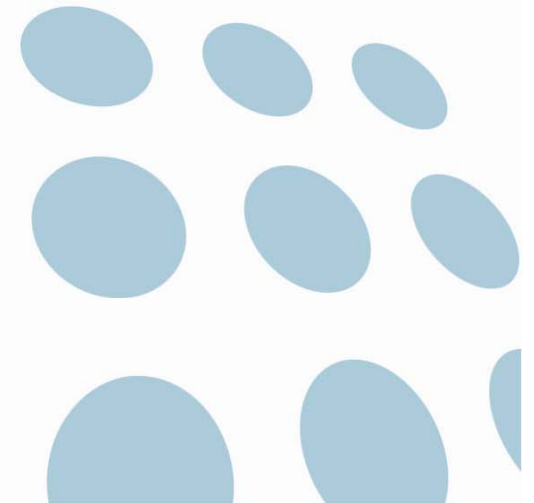
# FXO Port

- Can be Connected to POTS / GSM FCT for Making Calls
- A 'True' FXO Port
- Meant for INCOMING Calls as well as OUTGOING Calls



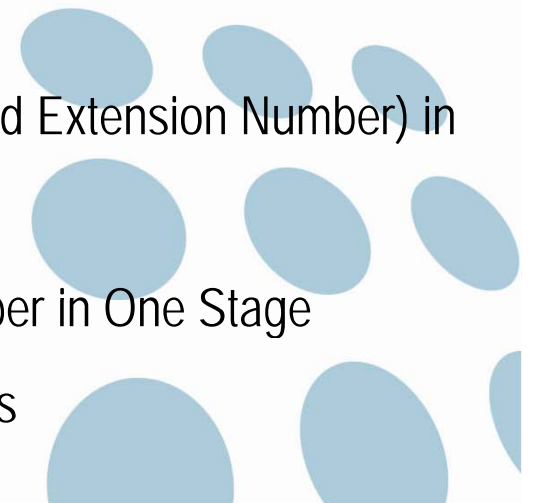
# MAC Cloning

- SETU ATA Facilitates Changing of MAC Address at the WAN Port
- Process of Changing MAC Address of a Device is Known as MAC Address Cloning
- Feature Benefit
  - ✓ Helps to Avoid Delays Involved In the Set Up Processes, wherein it is Required to Inform the Service Provider about the Newly Installed Equipment

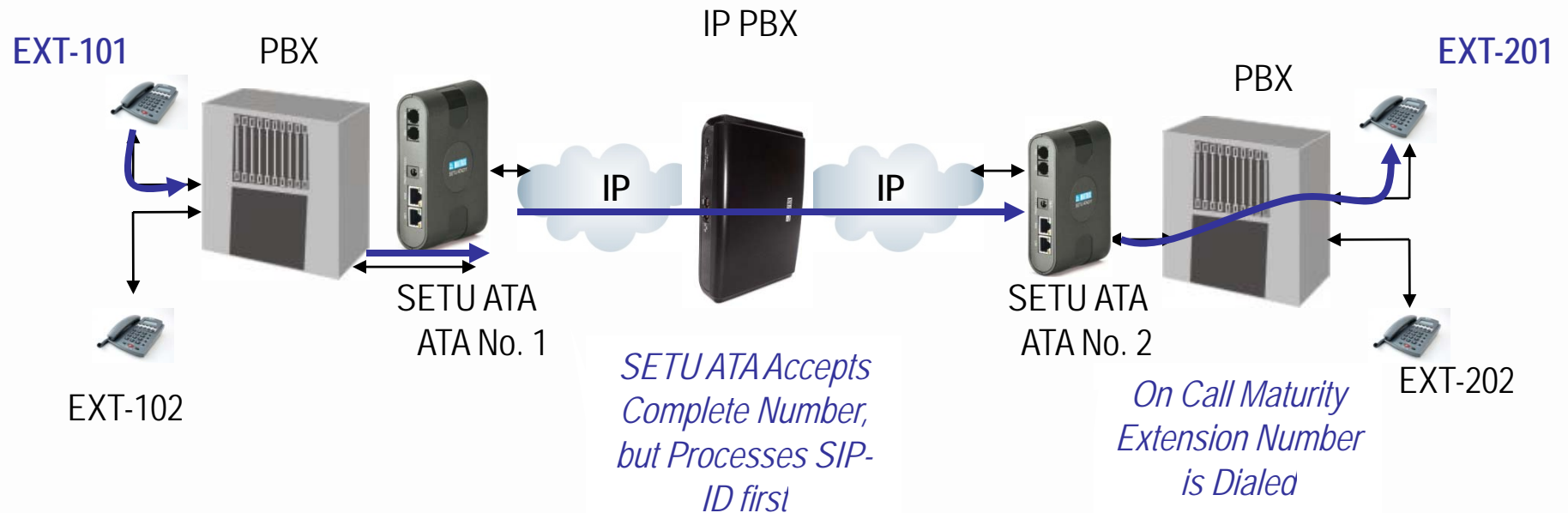


# Multi-Stage Dialing

- Feature Requirement:
  - ✓ Consider Cases when Calls are to be made between Multi-Location Office Site
- Normal Scenario:
  - ✓ While Calling an Extension in Another Location Caller Needs to Dial Number in Two Stages
  - ✓ SIP-ID of ATA , Followed by Extension Number
- Preferred User Scenario:
  - ✓ Caller wants to Dial Complete Number (SIP-ID of ATA and Extension Number) in One Stage Instead of Two
  - ✓ With Multi-Stage Dialing, Caller Can Dial Complete Number in One Stage
  - ✓ SETU ATA will Process the Dialed Number in Two Stages



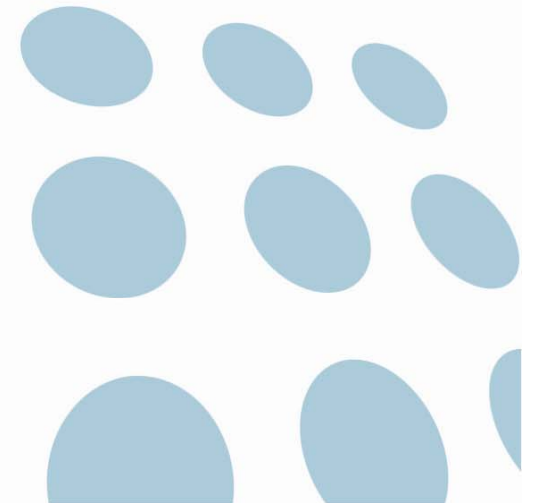
# Multi-Stage Dialing: Case Study



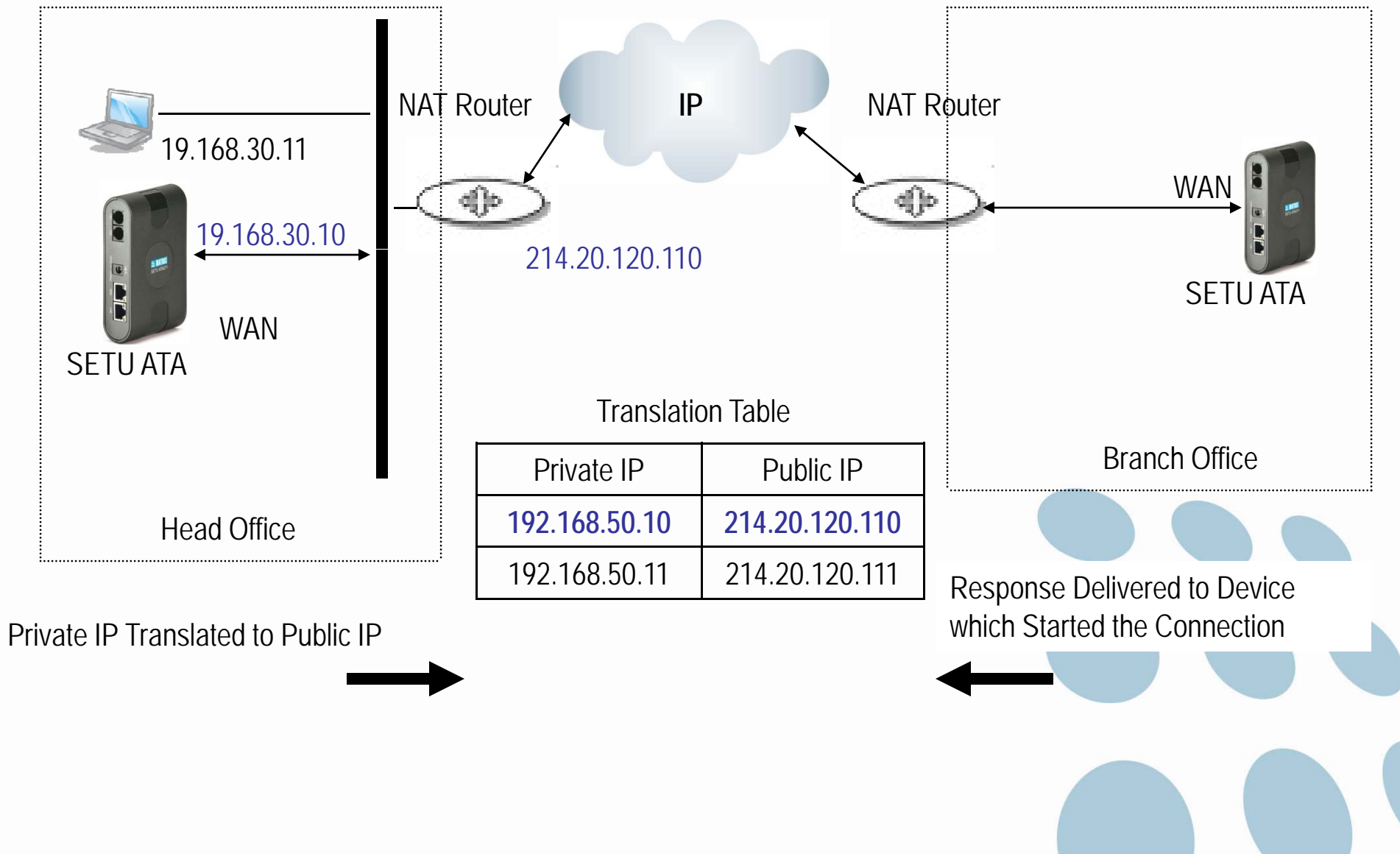
- ATA No. 1 and ATA No. 2 are Registered to the IP PBX
- EXT-101 Wants to Call EXT-201
- Caller Dials 22 (SIP-ID of ATA No. 2) and 201 (Extension to be Called) at One Go
- ATA No. 1 Accepts Complete Number but Dials SIP-ID of ATA No. 2 (22) First
- On Call Maturity, ATA No. 1 Sends the Remaining Number in DTMF to the Called Device

# NAT Support

- Network Address Translation
- A Technology used by Firewalls and Routers
- Feature Benefit
  - ✓ Allows Multiple Devices in a LAN to Share a Single Public IP Address
  - ✓ Enhances Security by Avoiding Direct Communication

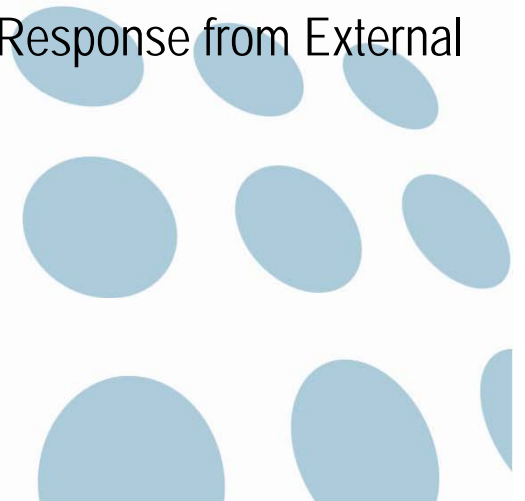


# NAT Router-Basic Operation

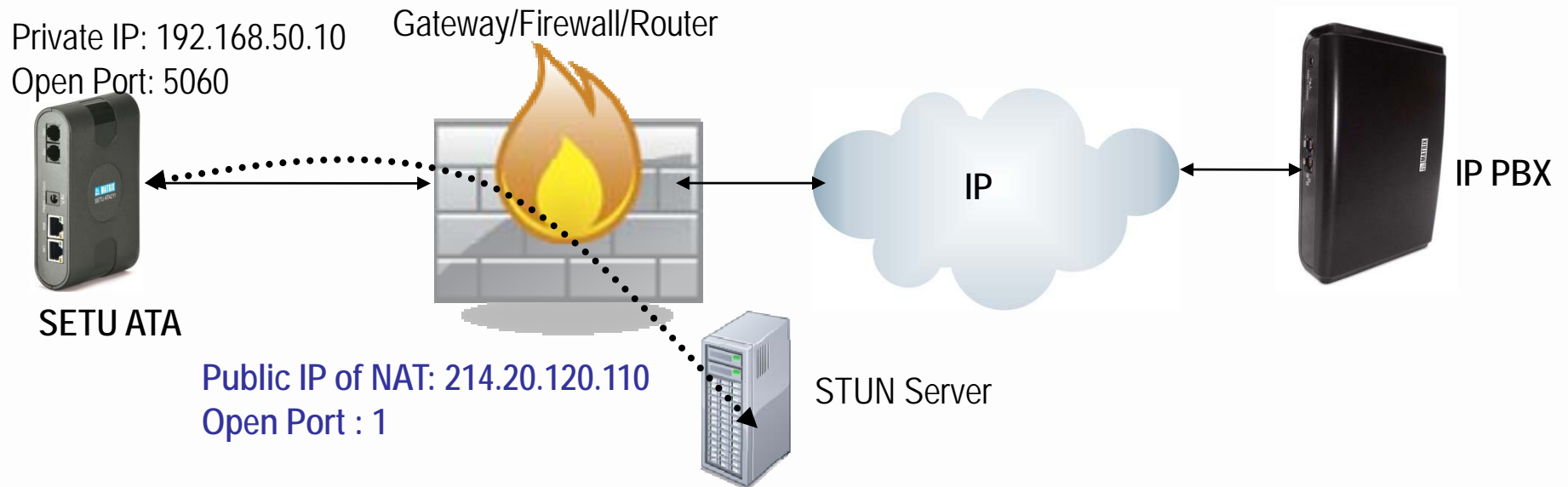


# STUN Support

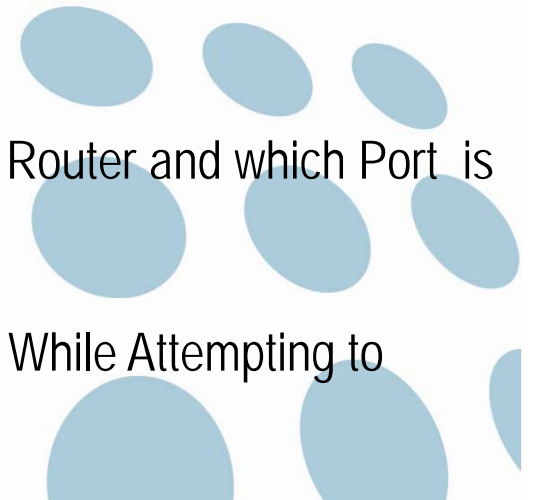
- STUN : Simple Traversal of User Datagram Protocol (UDP) through Network Address Translators (NAT's)
- A Client-Server Protocol
- STUN Allows a Client behind NAT to
  - ✓ Discover its Public IP Address
  - ✓ Discover the Type of NAT
  - ✓ Discover the Internet Side Port (Port on which Received Response from External SIP Terminals can be Mapped to its Own Open Port)



# STUN-Application Scenario

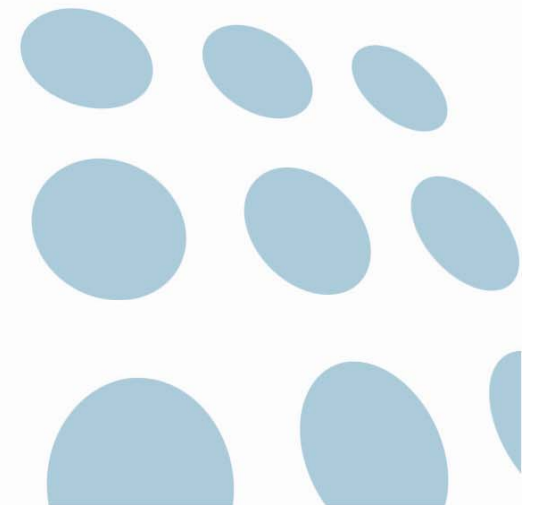


- STUN Client (like ATA) sends a request to STUN server
- The Server Reports Back with the Public IP Address of the NAT Router and which Port is Opened by NAT to Allow Incoming Traffic
- SETU ATA can Now Communicate the Public IP and Port details While Attempting to Register With the IP PBX



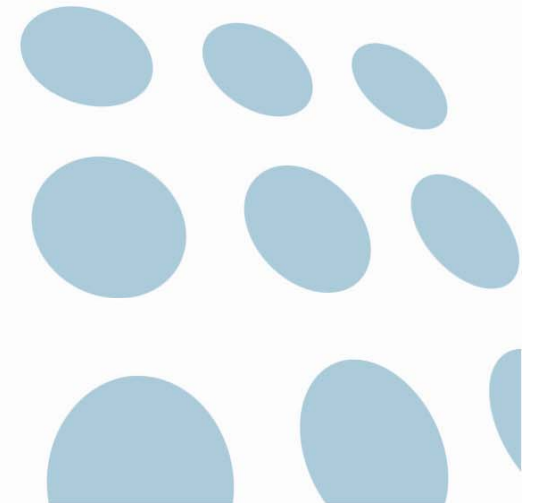
# STUN Support

- UDP: A Network Protocol for Transmitting Data (Without Acknowledgement/Connectionless Protocol)
- SIP Based VoIP Calls uses UDP as Transport Protocol
- STUN Allows ATA to work behind a Symmetrical NAT and Establish a VoIP Call

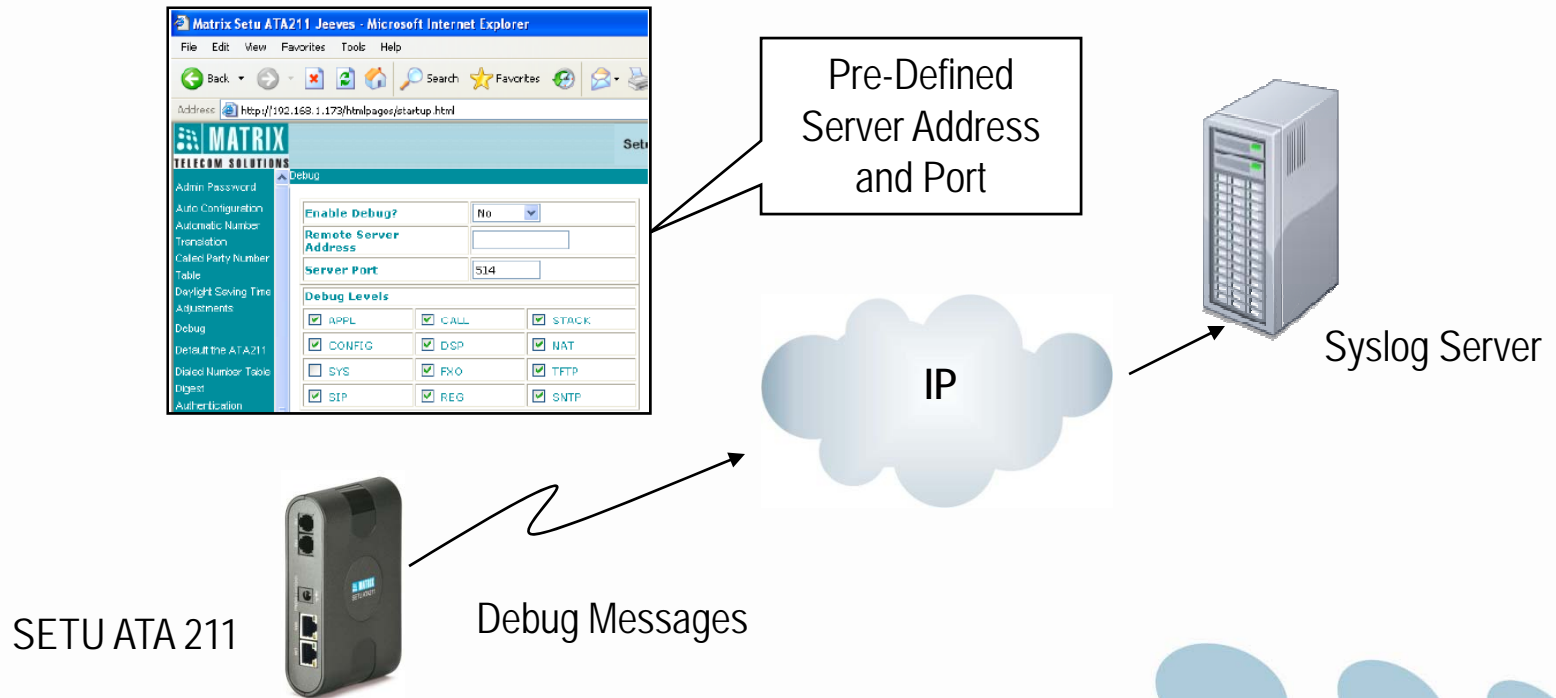


# Syslog Client

- Syslog: Protocol Used for Sending Debug Messages on IP Network
- A Client/Server Protocol
- Uses UDP as Transport Protocol for Debugging Process
- Logging has Several Benefits
  - ✓ Easier and Faster Troubleshooting
  - ✓ Security Enhancement
  - ✓ Better System Administration



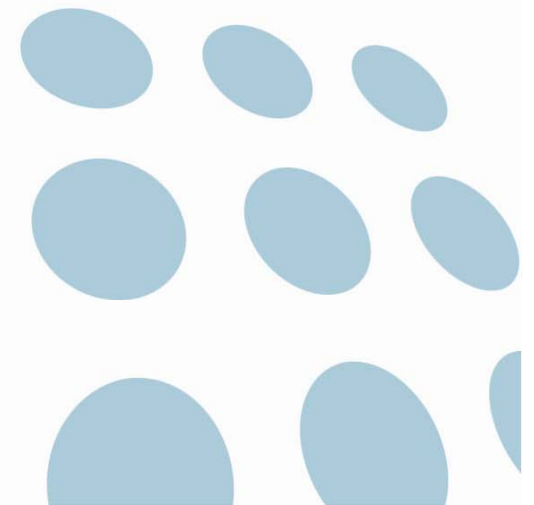
# Built-In Syslog Client



- SETU ATA has Built-In Syslog Client that Enables it to Send Debug Messages, to Remote Server, Via IP

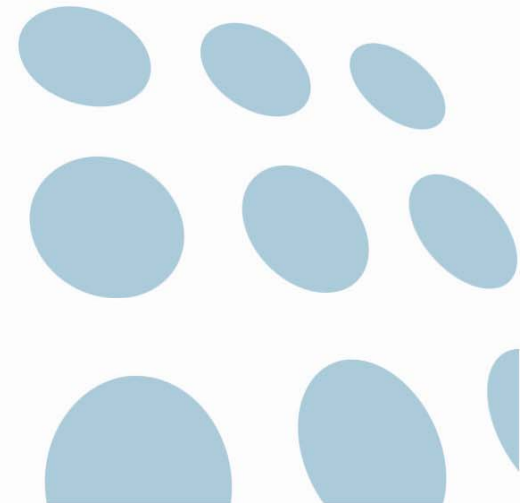
# PIN Authentication

- Used to Authenticate a Caller
- Caller has to Prove his Identity Before a Call is Routed from One Network to Another
- Feature Benefit
  - ✓ Avoids the Possibility of Unauthorized Use of Services



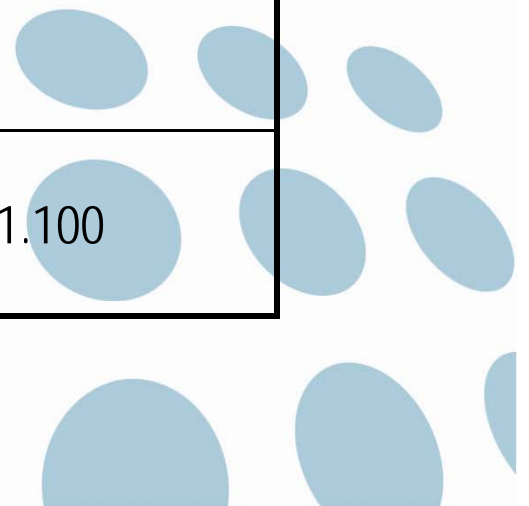
# Peer-to-Peer

- SETU ATA can Call another ATA's Extension or Soft phone (On Same LAN/Same Location or On Virtual LAN/Remote Location) without going through Proxy
- Extension Number of Remote ATA , along-with its IP Address is Programmed in Peer-to-Peer Calling Table
- Total 500 Numbers Can be Programmed



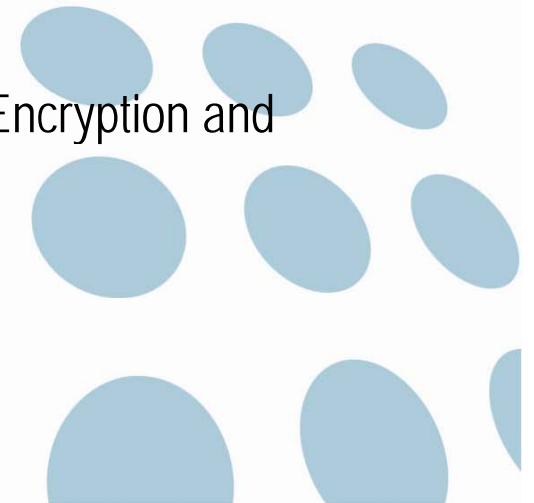
# Peer-to-Peer Table

Index	Number	IP Address
01	201	192.168.1.50
02	202	192.168.1.75
:	:	:
500	600	192.168.1.100



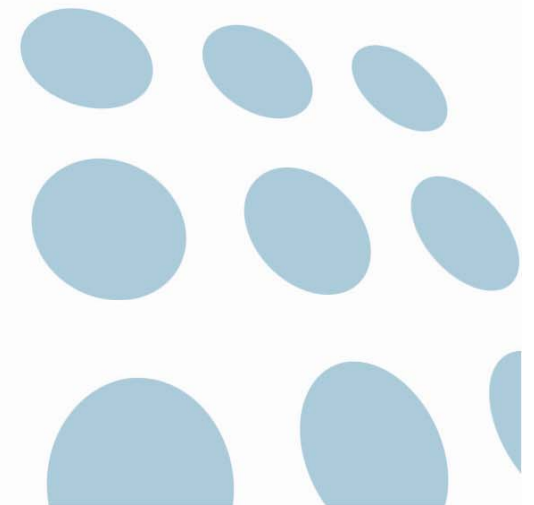
# PPPoE

- PPPoE : Point-to-Point Protocol over Ethernet
  - ✓ A Network Protocol for Encapsulating PPP Frames in Ethernet Frames
- Feature Requirement
  - ✓ It is used to Virtually Dial Another Ethernet Machine and Make a Point-to-Point Connection
  - ✓ Mainly used for xDSL Services using xDSL Modem
- Feature Benefit
  - ✓ Offers Standard PPP Features such as Authentication, Encryption and Compression



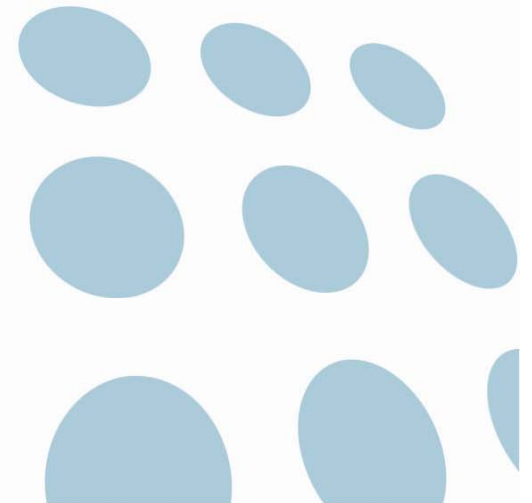
# Quality of Service (QoS)

- QoS : Capability of a Network to Provide Better Service
  - ✓ Types of QoS Supported : ToS and DiffServe
  - ✓ ToS (Type of Service): Server Decides Best Method of Delivering Data (Features like LCR, Fastest Route)
  - ✓ DiffServe (Differentiated Services) : Similar to Class of Service, Based on Application Type and Traffic Flow Services are Provided
- Feature Requirement
  - ✓ Decides Priority of Bandwidth Allocation
  - ✓ Controls Jitter and Latency
- Feature Benefit
  - ✓ Improves Voice Quality



# Voice Activity Detection

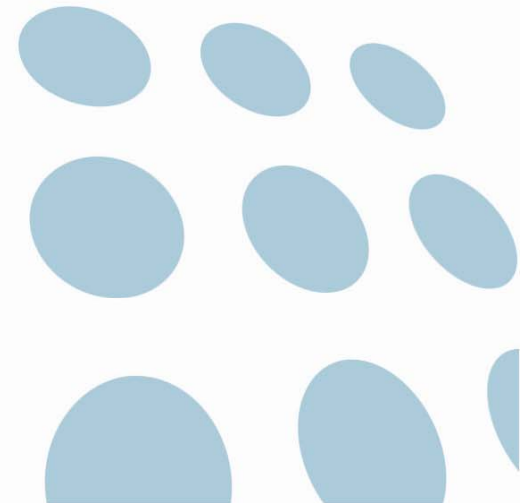
- It is a Software Application
- Detects Audio Absence during Speech
- Prevents the Transmission of these " Silence Packets" over the Network
- Saves Bandwidth



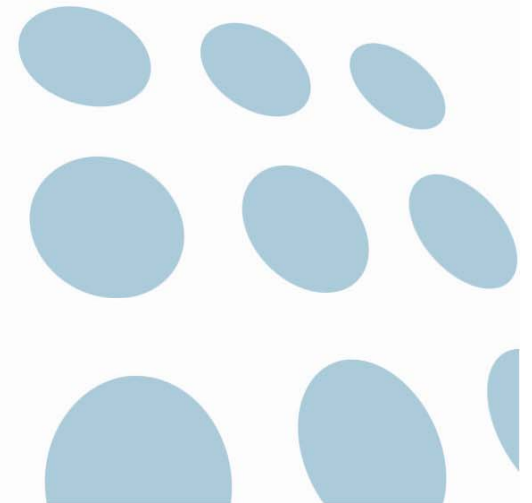


# Target Customers

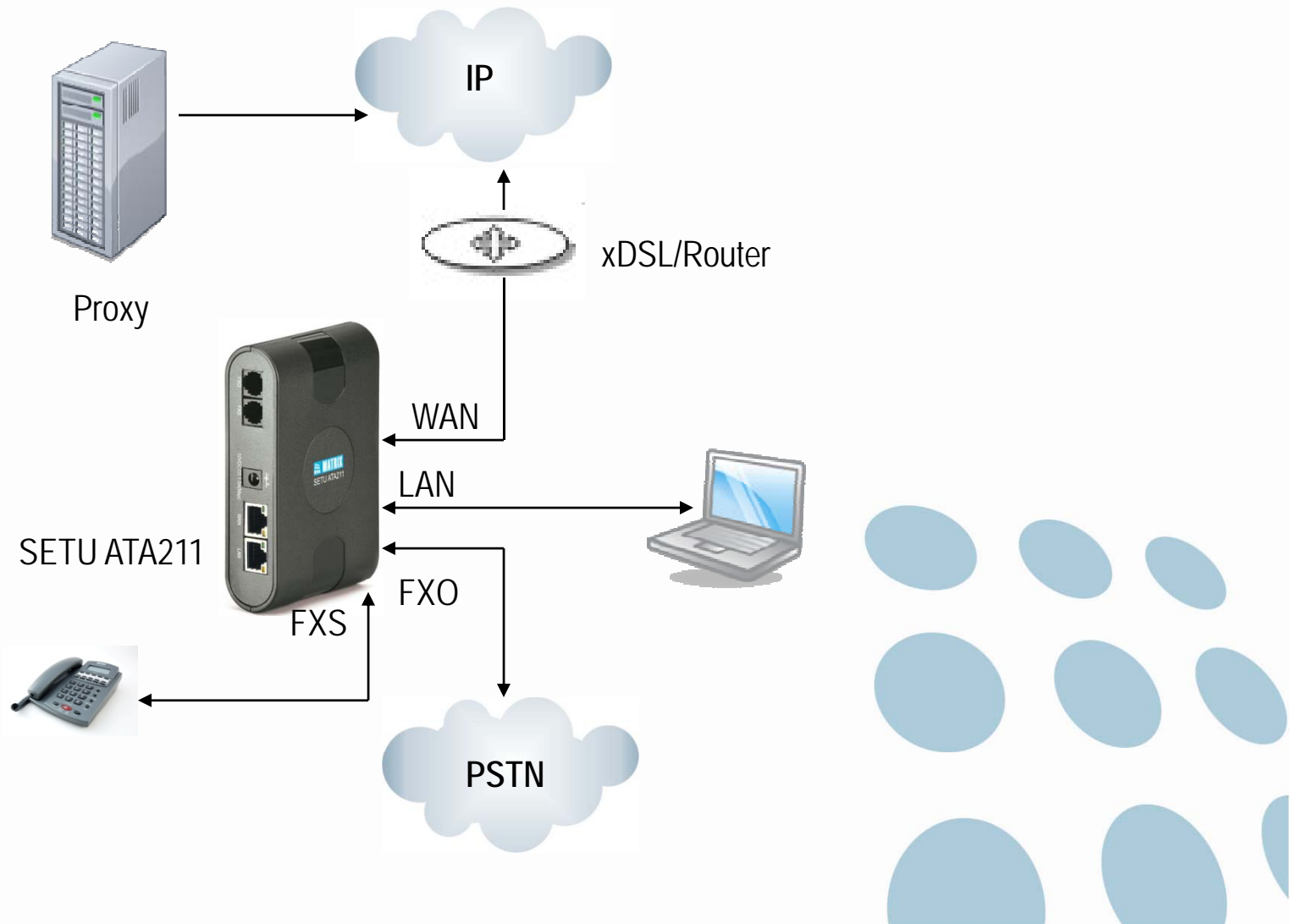
- Home Users
- Multi-Site Scenarios
- Businesses
- Call Shops



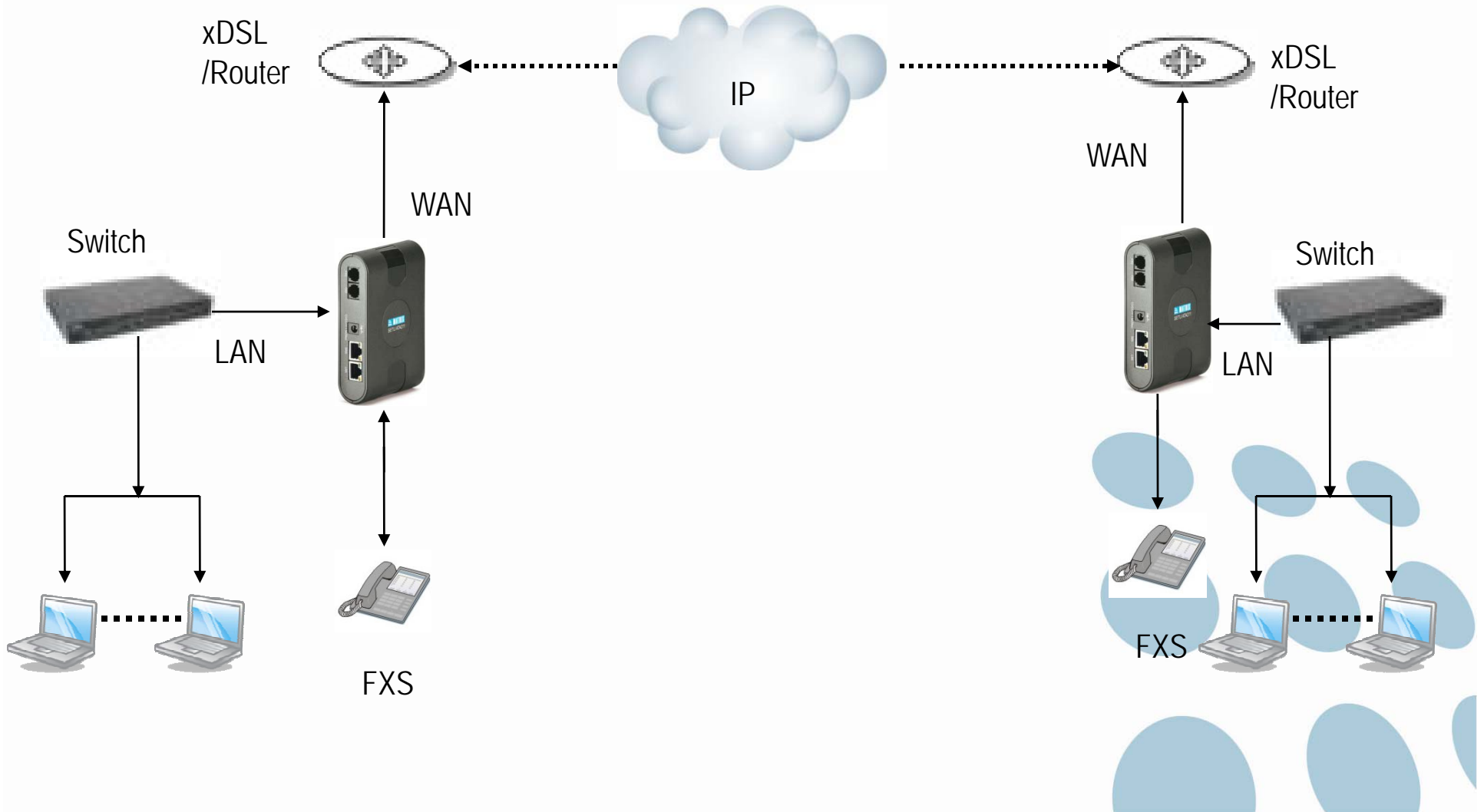
# Applications of SETU ATA211



# SETU ATA211: Residential Application



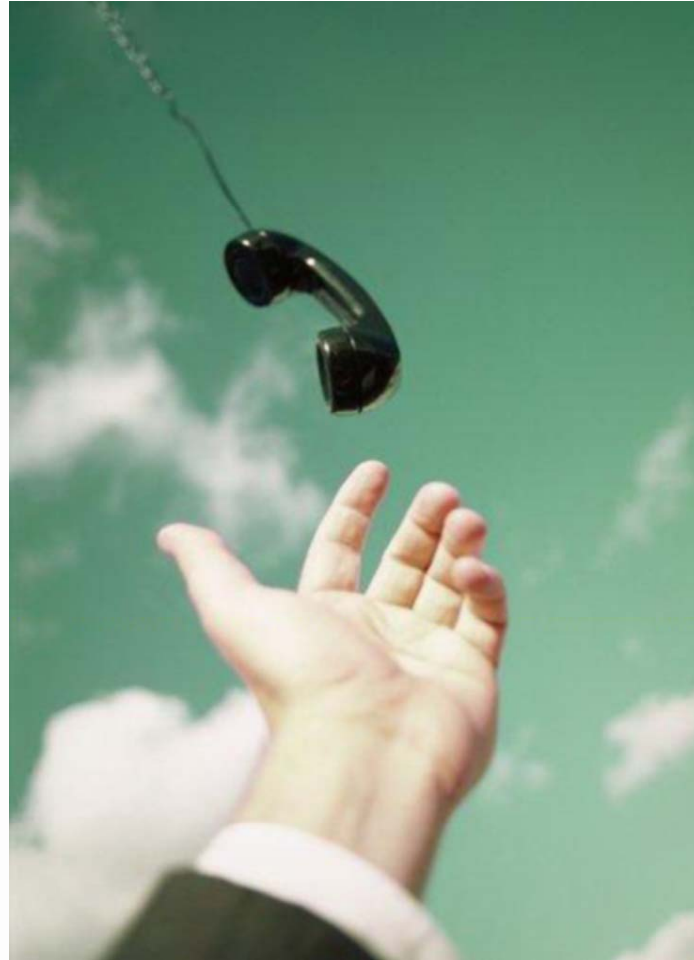
# SETU ATA211: Peer-to-Peer Application



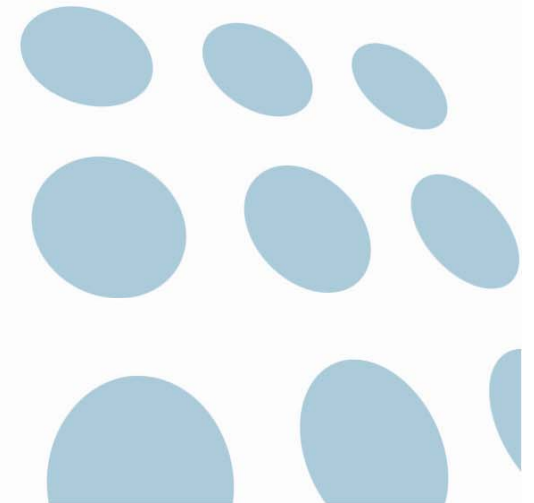


# Matrix VoIP Product Range

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 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 Analog Telephone Adaptor with 1 GSM, 1 FXS Port and 2 Ethernet Ports
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 2 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 VP248P	Executive IP-Phone with 6 Lines x 24 Characters LCD Display
SETU VP248S	Executive IP-Phone with 2 Lines x 24 Characters LCD Display



Thank You



- Type of Presentation: Product Presentation
- Number of Slides: 41
- Revised On: 6<sup>th</sup> August, 2010
- Version-Release Number: V1R2

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