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TELECOM SOLUTIONS

Hearty Welcome

Matrix SETU VFX
Multi-Port VoIP Gateway

Introduction

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Matrix SETU VFX Family

Interfaces	SETU VFX44L	SETU VFX88L
FXS Ports (RJ11)	4	8
PSTN-Pass Through Port	1	1
Ethernet Port (RJ45) 10/100 Base T	1	1
DC Power Input Jack	1	1

SETU VFX

SETU VFX44L

SETU VFX88L

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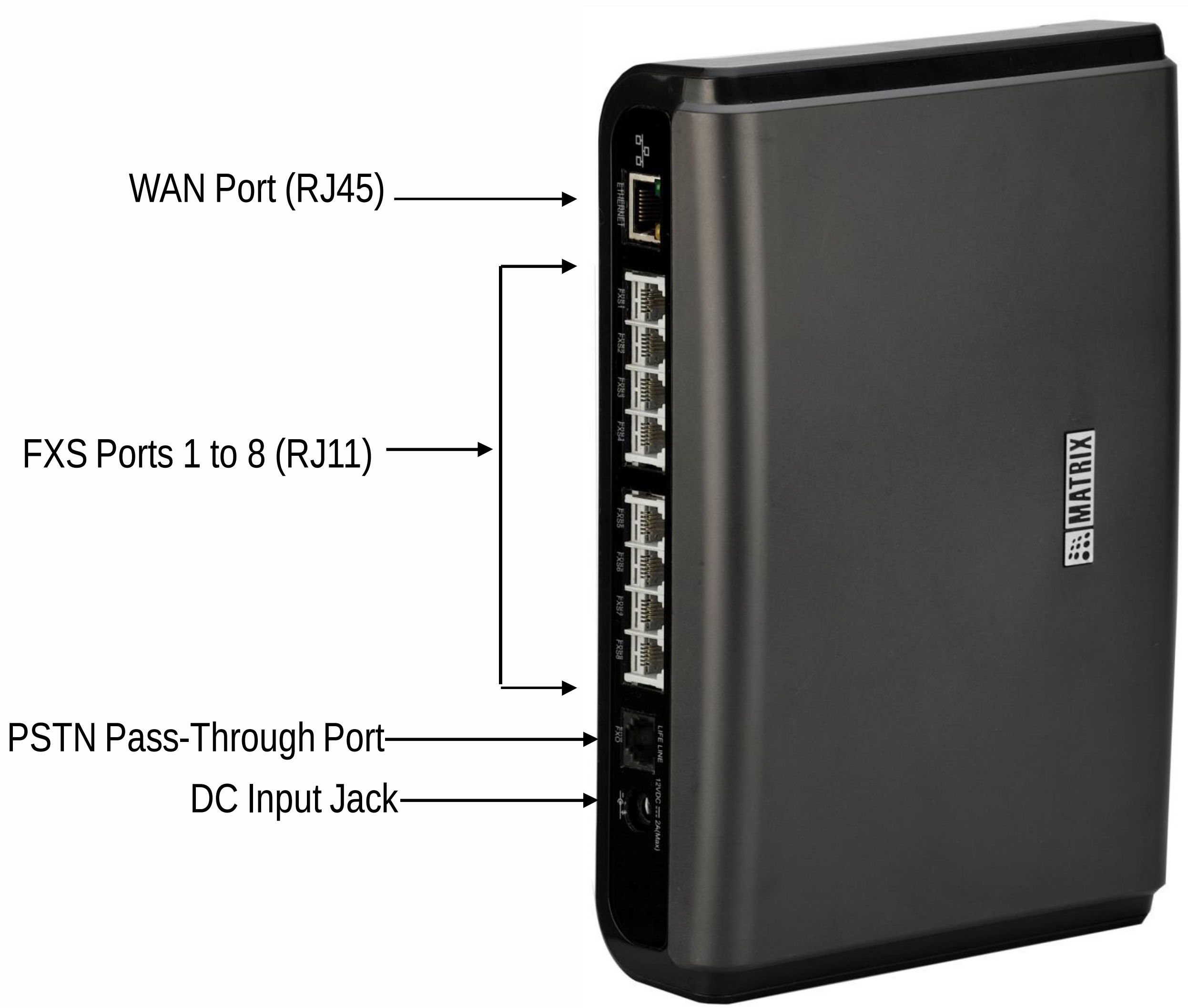
Interfaces


WAN Port (RJ45)
FXS Ports 1 to 8 (RJ11)
PSTN Pass-Through Port
DC Input Jack
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What is SETU VFX?

- SETU VFX44L/ VFX88L Links VoIP Networks to Traditional Telephony and vice-versa
- It has a 4/8 Channel (9 SIP Accounts) for VoIP Calls
- Allows 4/8 Simultaneous VoIP Calls using FXS Ports
- It can be Interfaced with PBX or GSM FCT having FXO Port
- Eliminates the Need to Upgrade an Existing PBX for Making IP Calls

Matrix SETU VFX–Hardware Features

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Matrix SETU VFX–Hardware Features

- Compact and Sturdy Design
- 1 FXO (PSTN Pass-Through) Port
- 1 WAN Port (10/100 BaseT)
- 4/8 FXS Ports
- DSP Technology
- Codec Technology
- Real Time Clock (RTC)
- Wall and Table-top Mounting

Compact and Sturdy Design

- SLIC and SMT Design
- Less Heat Generation
- Dual Protection
- Powder Coated Aluminum System Sub-Rack
- Wall and Table Top Mounting

DSP Technology

- DSP: Digital Signal Processor
- Executes Electronic Processing and Simulation of Signals
- Capable of Executing Numeric Processing Quickly
- Can Execute Real-Time Computations
- Consumes Less Power
- Occupies Less Space on PCB Die to its Compact Size

Codec Technology

- Voice is Encoded from Analog to Digital IP Packets
- Signals are Decoded back to Analog Voice at Receiver End
- Codec's Supported : G.711 A-Law, µ-Law, G.723.1, G.729A, G.729B,iLBC,GSM-FR

Mounting Options

- Two Mounting Options are Available:
  - ✓ It Can be Mounted on the Wall
  - ✓ It Can be Placed on the Table-Top

Matrix SETU VFX–Software Features

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SETU VFX–Software Features

- 9 SIP Accounts
- Access Codes
- Allowed and Denied List
- Automatic Number Translation
- Block Black Listed Callers
- Call Detail Records (CDR)
- Calling Party Control (CPC)
- CLIP
- Class of Service
- DHCP Client
- Day Light Saving Time (DST)
- Dial Plan
- Echo Cancellation
- Emergency Number Dialing
- Fax Over IP (FoIP)
- PSTN-Pass Through Port
- Hotline

SETU VFX–Software Features

- Incoming Calls Routing
- LED Indications
- Multiple Gateway
- Peer-to-Peer Calling
- Password Protection
- PPPoE Client
- Programmable Call Progress Tone
- Programmable Ring Tones
- Remote Programming
- 100 Rel / PRACK
- SIP Over TCP
- Supplementary Services
- STUN and NAT Keep Alive
- Voice Activity Detection
- Web Based Programming

SIP Account

- SIP–Session Initiation Protocol
- Nine SIP Accounts are Provided
- Each FXS can be Assigned One of the SIP Accounts for Making Outgoing Calls
- Using Dial Plan Feature, Dynamic (Dedicated) Allocation of SIP Accounts is Possible

Access Codes

- A String of Digits Dialed to Access Supplementary Features
- A String of Digits Used to Enable/Disable Features or Enter Programming Mode
- Access Codes of Your Choice can be Programmed
- Access Code for Programming Mode can not be Changed
- Access Code Can be Maximum of 3 Digits
- To Create an Access Code 0-9, *, # and ^ Signs are Allowed

Allowed and Denied Numbers

- Can be Programmed to Allow or Deny Numbers from being Dialed
- Avoid Misuse and Restrict Unproductive Calls
- Very Useful Feature to Control Cost
- Can be Programmed Separately for VoIP, GSM, FXO and FXS Ports
- 24 Number Lists can be Programmed
- 64 Entries per List are Supported

Automatic Number Translation

- Translates full Number or Part of a Dialed Number to Match with Numbering Plan of the Destination Network
- Automatic Number Translation (ANT) is Supported on SIP
  - ✓ User Dial 001-xxxx to Reach a Number in the USA, But as the ITSP Understands 1-xxxx so ANT Replaces the Number 001-xxxx with 1-xxxx to let the ITSP Understand the Dialed Number String

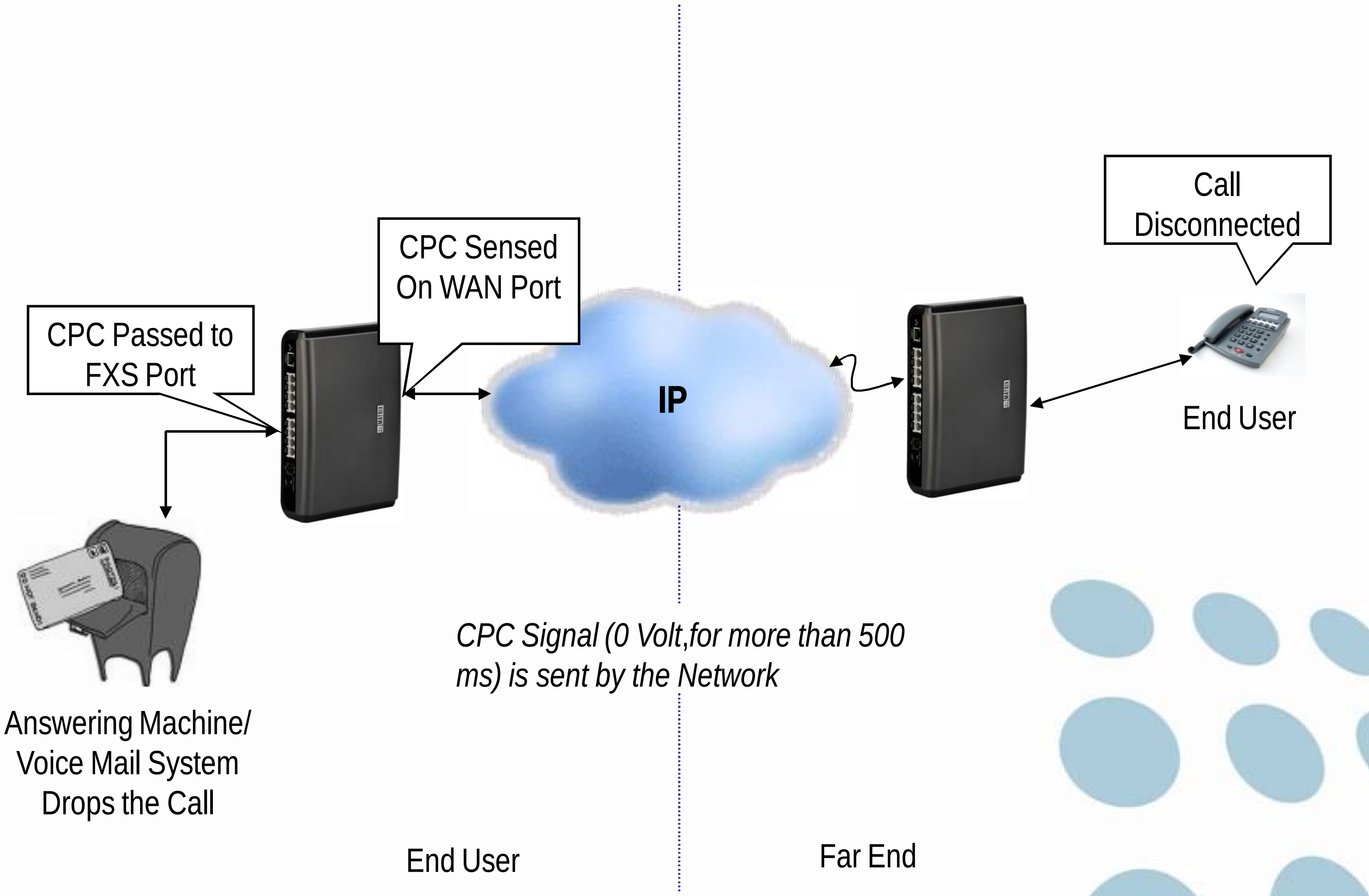
Call Detail Records (CDR)

- SETU VFX stores up to 2000 Calls
- Report can be Generated with Various Filters like:
  - ✓ Port
  - ✓ Number
  - ✓ Date
  - ✓ Time and Duration

Calling Party Control (CPC)

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

Calling Party Control (CPC)



Caller Line Identification Protocol

- Detection of DTMF,FSK ITU-T V.23 and FSK Bellcore 212A
- CLIP for External Numbers, Internal Numbers
- FXS Port can be Programmed for CLIP Protocol
- This feature can be Enabled/Disabled as per Requirement

Call Progress Tones and Rings

- Different Tones can be Programmed to indicate the Progress of a Call Activity
  - ✓ Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Feature Tone and Confirmation Tone are Examples of CPT
- Tones with Different Cadence, with Specific Frequency are Offered
- User can Select a Desired Tone which Matches with the Tones used in the Concerned Region

Class of Services

- This Feature Allows Provision of Specific Feature to be used by Specific Users only, while Restricting Others from using the Same

DHCP Client

- DHCP: Dynamic Host Configuration Protocol
- DHCP is a Client-Server Networking Protocol
- DHCP Provides Mechanism for Dynamic Allocation of IP Address to Host (Computers) on the Network
- Basic Operation
  - ✓ DHCP Client Requests Server for Specific Information required to Participate on the Internet Network
  - ✓ DHCP Server Provides Configuration Parameters Specific to the DHCP Client

Daylight Saving

- Daylight Saving is the Procedure of Setting Clocks Ahead at a Particular Time of Year to Make way for Additional Hour of Daylight in the Evening
- Real Time Clock (RTC) Moves Backward or Forward Automatically in Tune with the Daylight Saving Requirement of the Country
- DST can be Forwarded or Set Backward According to Day-Month wise or Date-Month wise as Per the 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 VFX Searches for a Matching Number in the Programmed Number's list
  - ✓ A Number is then Dialed as Per the 'Best-Fit' Logic
  - ✓ 10 Numbers can be Programmed & Mapped along with Preferred SIP Account

Dial Plan

Index	Number	SIP Account
01	2001	SIP1
02	2002	SIP2
:	:	:
10	2010	SIP9

Echo Cancellation

- A Technique for Isolation and Filtering of Unwanted Signals Caused by Echoes
- Feature Requirement
  - ✓ Echoes are Caused during Conversion of 4-Wire to 2-Wire by Telephony Hybrid
  - ✓ Echo Cancellation Device puts a Signal on the Return Transmission Path which is Equal and Opposite of the Echo Signal
  - ✓ Echo Cancellation Devices (Digital Filters) are used Depending on the Length of Echo
- Feature Benefit
  - ✓ Provides Better Voice Quality

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Emergency Number Dialing

- SETU VFX Supports a Pass-through Port for Dialing Emergency Numbers
- Feature Requirement
  - ✓ Acts as a Redundancy Line in case of Internet/Power Failure
  - ✓ Can be connected to PSTN/GSM FCT for Making Certain Calls
  - ✓ It is Not a ‘True’ FXO Port
  - ✓ It is meant for INCOMING Calls Only

Password Protection

- SETU VFX Provides Password Facility to Ensure System Security
- Feature Benefit
  - ✓ It Allows User to Set Password of their Choice
  - ✓ Prevents Un-authorized use and Tampering with System Settings

Peer-to-Peer

- SETU VFX 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 Calling

Index	Number	IP Address
01	2001	192.168.1.10
02	2002	192.168.1.125
:	:	:
500	2010	192.168.1.145

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

Supplementary Services

- Call Forward on Busy
- Call Forward on No Reply
- Call Forward Unconditionally
- Call Hold
- Caller ID
- Call Transfer-Blind
- Call Transfer-Attended
- Call Conference (3 Party)
- Do Not Disturb (DND)

Call Forward

- Allows user to Forward Calls from One SIP Number to another SIP Number (SIP1 to SIP2 or Vice-Versa)
- Various Options for Call Forwarding:
  - ✓ Call Forward : Un-conditional
  - ✓ Call Forward : When Busy
  - ✓ Call Forward : When No Reply
- Timer (1 to 99 Sec) can be Set for Call Forwarding
- On expiry of timer, the System Initiates Call Forwarding

Call Hold

- Flexibility to Receive Second Incoming Call while Answering the First Call
- Flexibility to Make Outgoing Call while Answering the First Call
- Basic Operation
  - ✓ User can Put First Party on Hold and Attend Second Call
  - ✓ On Completion of the Second Call, user is connected to the First Party
  - ✓ Allows user to Speak to Two Parties Alternatively
  - ✓ The Person On-Hold hears Music, if Provided by the Operator

Call Transfer

- An ITSP Dependant Feature
- Two Types of Call Transfer are Offered
  - ✓ Attended Transfer: Transfer of Call after Consultation with Third Party
  - ✓ Blind Transfer: Transfer of Call without Prior Consultation of the Third Party

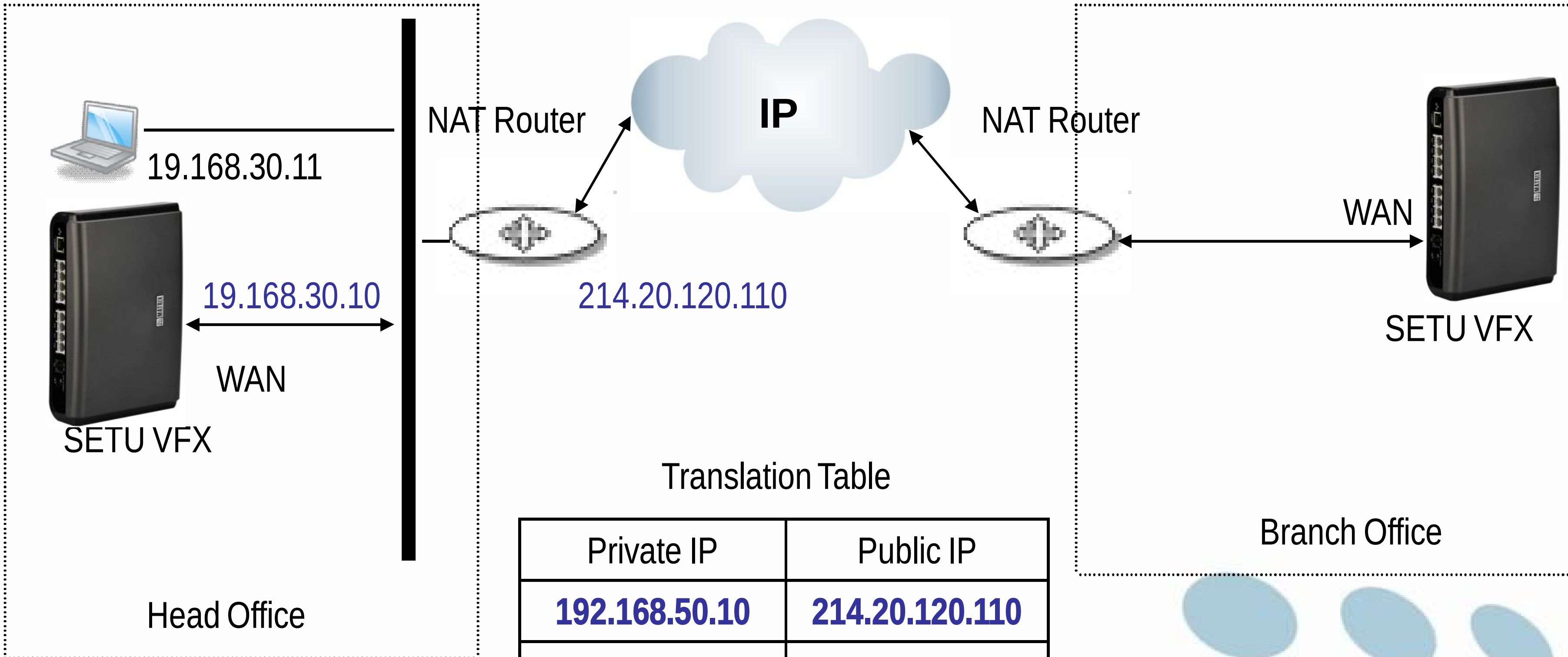
DO Not Disturb (DND)

- Provides User a Flexibility to Stop Receiving Incoming calls
- DND is a Programmable Feature for SIP Account
- DND can not be Set for FXS Ports

NAT Support

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

NAT Router-Basic Operation



Translation Table

Private IP	Public IP
192.168.50.10	**214.20.120.110**
192.168.50.11	214.20.120.111

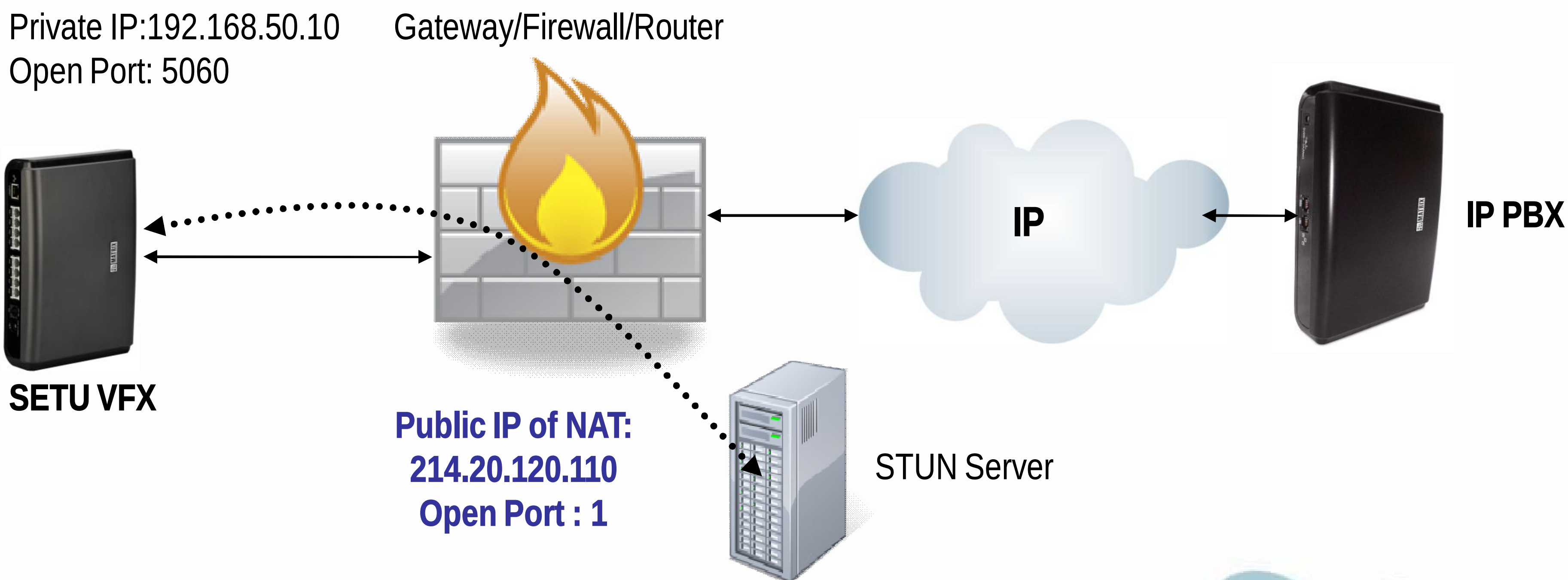
Private IP Translated to Public IP

Response Delivered to Device which Started the Connection

STUN Support

- STUN : Simple Traversal of User Datagram Protocol (UDP) through Network Address Translators (NAT's)
- A Client-Server Protocol
- STUN Allows SETU VFX to work behind a Symmetrical NAT and Establish a VoIP Call
- 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 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 Back into the Network
- SETU VFX can Now Communicate the Public IP and Port details While Attempting to Communicate With the Remote IP PBX

Voice Activity Detection

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

Web Based Programming

- SETU VFX - Jeeves is a Web based Software Tool
- Intuitive, User Friendly GUI based Programming Tool
- It is Pre-Loaded in the SETU VFX
- Feature Benefit
  - ✓ Allows User to Configure WAN, SIP, FXS ports, FXO Port, Dial Plan, Peer-to-Peer setting of VFX
  - ✓ User can Save/Upload the Changes Made
  - ✓ Jeeves-Being Password Protected, Prevents VFX from Unauthorized use or Tampering of Settings

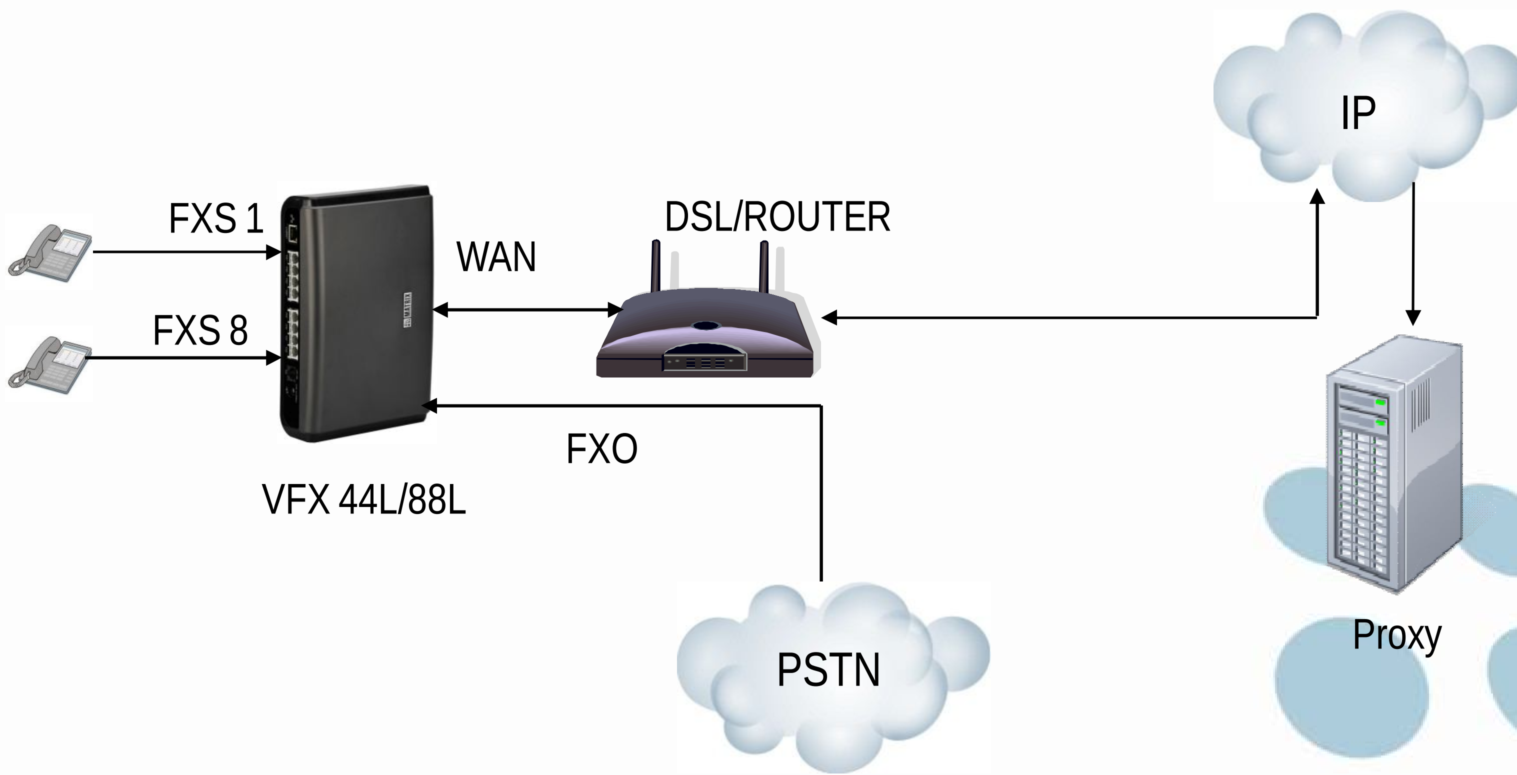
Target Customers

Target Customers

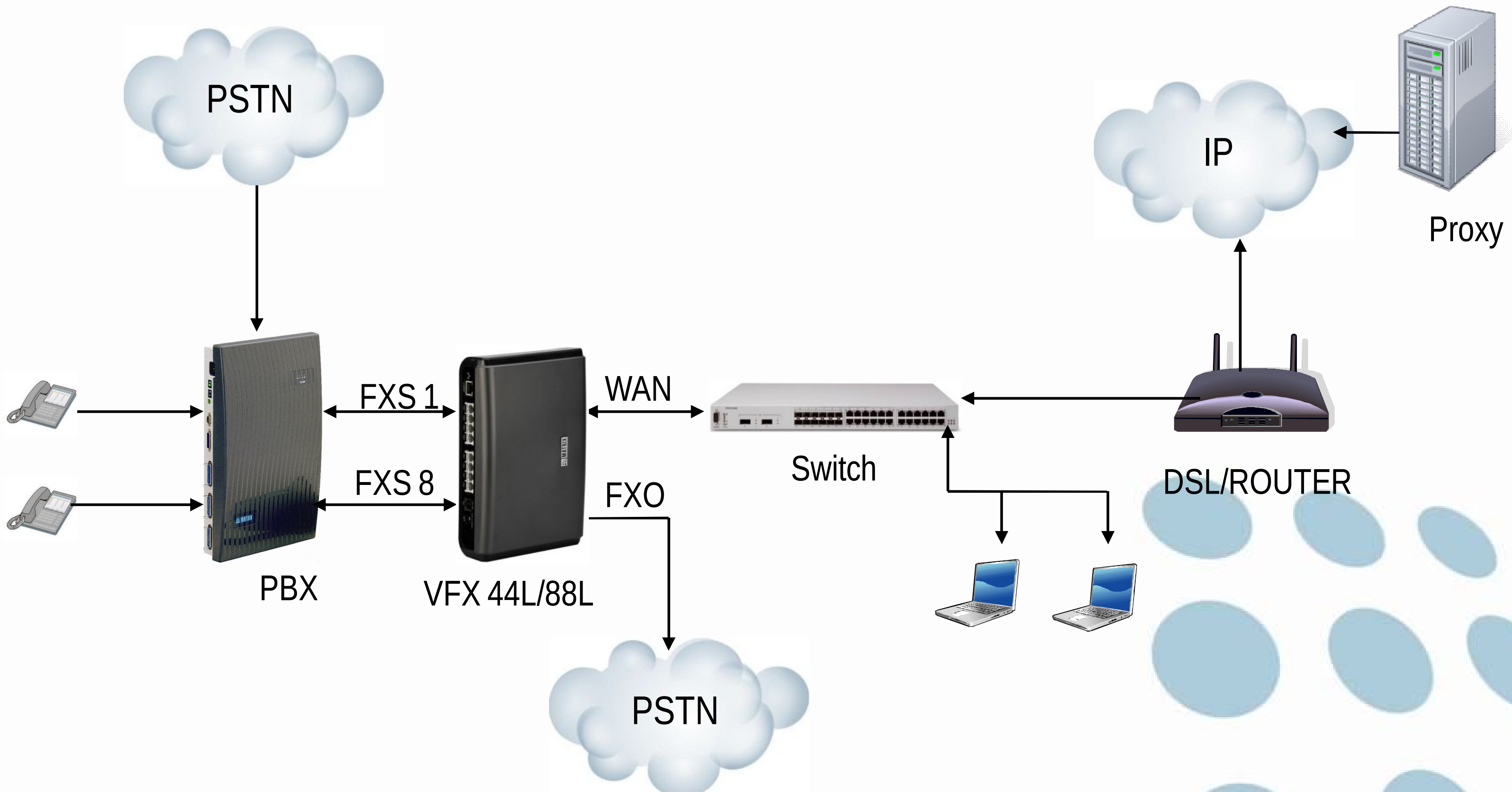
- Home Users
- Offices
- Multi-Site Scenarios
- Businesses

Matrix SETU VFX – Applications

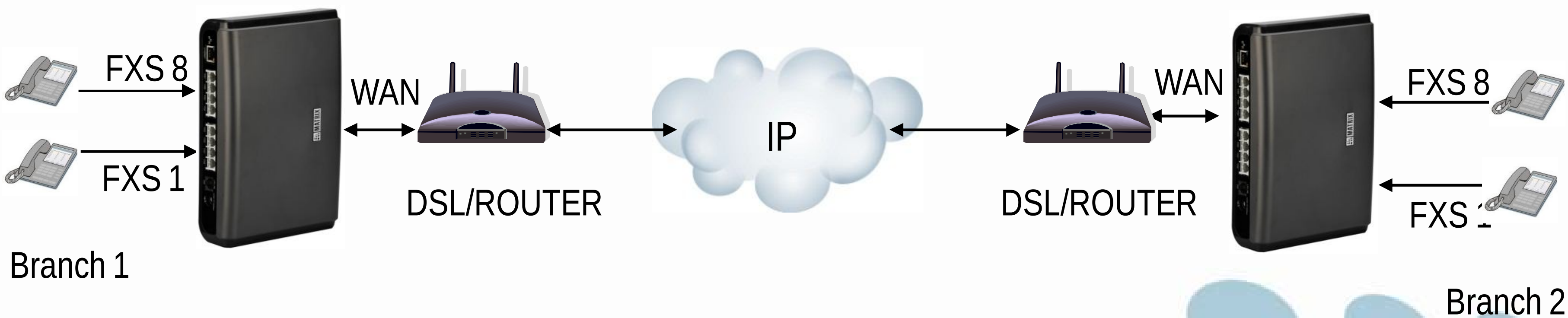
SETU VFX44L/88L: SOHO Application



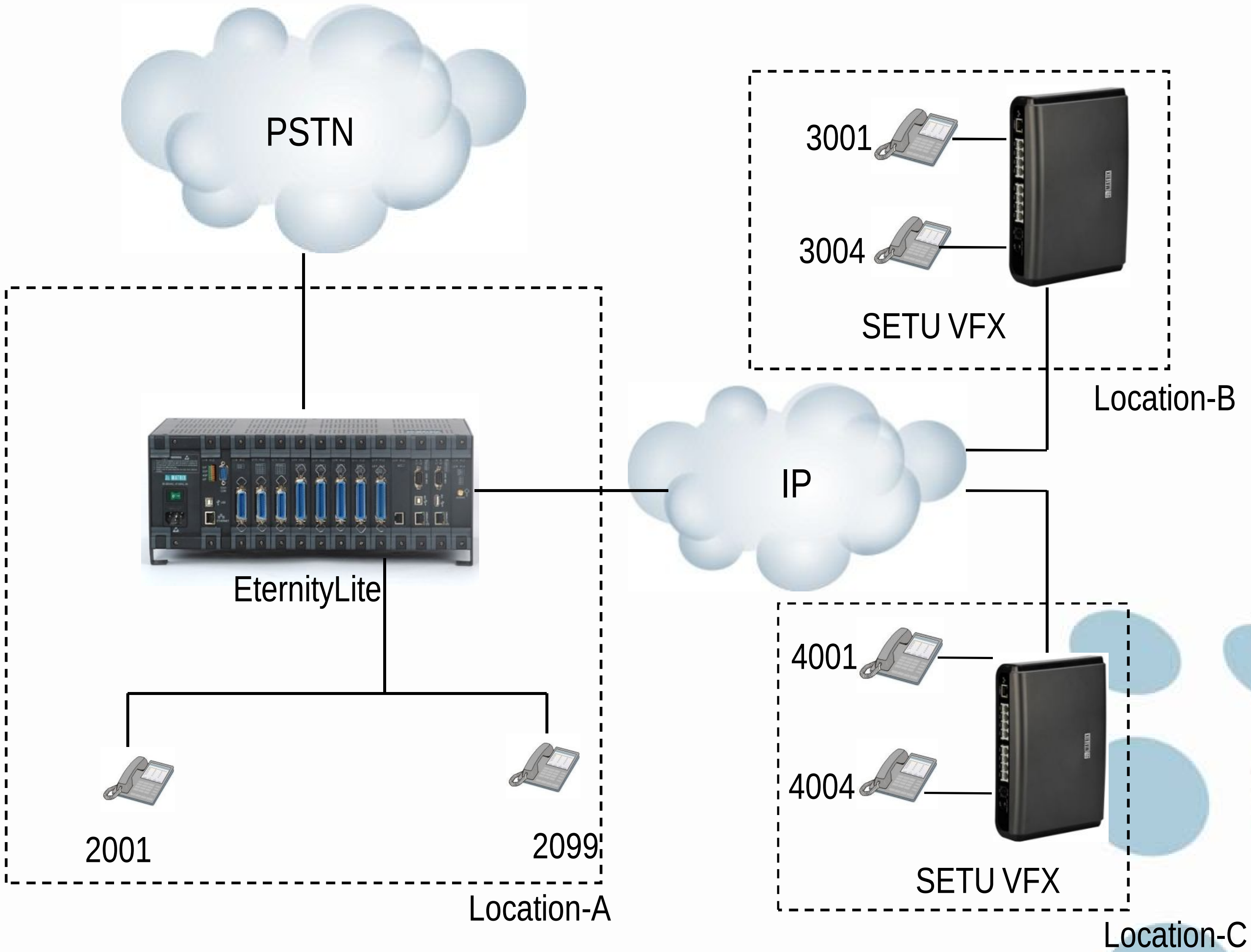
SETU VFX44L/88L: Business Application



SETU VFX44L/88L: Peer-to-Peer

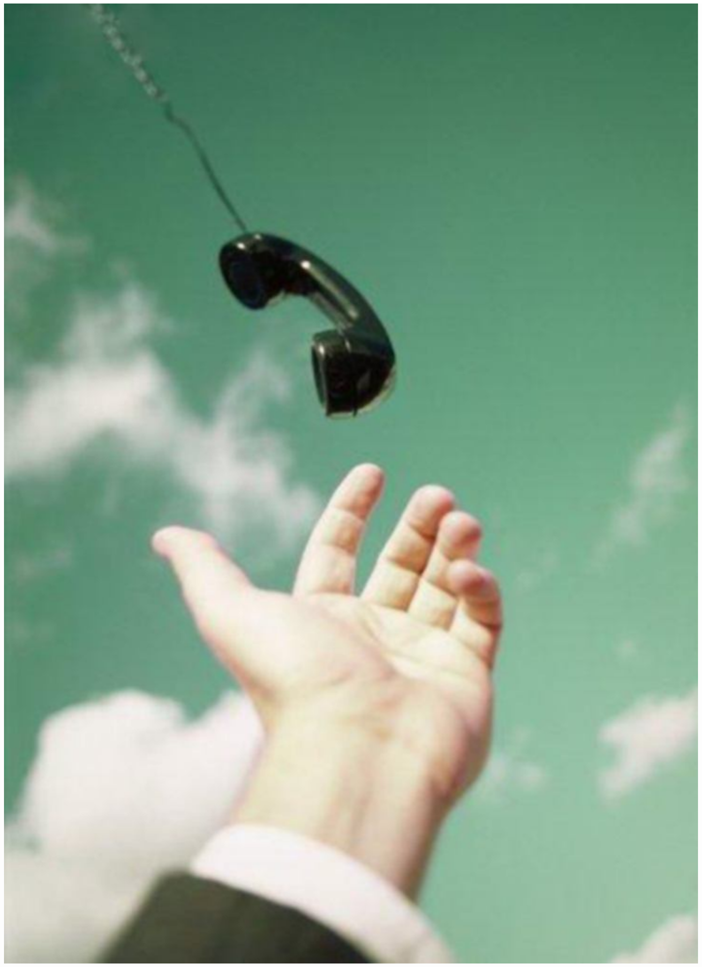


Eternity GE and VFX in Peer-to-Peer



Matrix VoIP Product Range

VYOM CCX	High-Density Universal Gateway
SAPEX	All-in-One Embedded IP-PBX Server
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 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



Thank You

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