Warm Welcome

SAPEX

The All-in-One Embedded IP-PBX Server

Introduction

- Pure IP-PBX
- Embedded Server Platform
- Registrar, Proxy, Presence and Voice Mail Servers : All-Integrated
- Open-Standard SIP Protocol Support

System Models

SAPEX SDM

-With Single DSP Module (SDM)

SAPEX DDM

-With Dual DSP Module (DDM)

Server Specifications

- Open-Architecture, Supports SIP v2 Protocol
- Embedded Registrar, Proxy, Voice Mail and Presence Servers
- Back-to-Back User Agent (B2BUA)
- Registration of Multiple SIP Trunks
- Seamless User Connectivity (NAT and STUN Support)
- Dynamic DNS (DDNS) Support
- Presence and Instant Messaging

Open Architecture

- SAPEX Employs Open-Standard SIP Protocol to Establish Calls Over IP Network
- Full Interoperability With Any Third-party SIP Equipment Such as IP Phones, Softphones, Gateways and Proxy Servers

Open Architecture

- SIP is a Open-Standard Signaling Protocol for Establishing Communication Session over Internet
  - ✓ Voice, Video or Instant Messaging Sessions
- SIP Architecture
  - ✓ SIP Servers and SIP Endpoints (User Agents/Terminals)
- Varied Options of Communication Terminals
  - ✓ PC, PDA, Cell Phone With SIP Client, IP Phone, Softphones

SIP Servers Types

Registrar Server	Authenticates and Registers When User Comes Online
Proxy Server	Processes Session Requests and Responses of a SIP Call
Redirect Server	Redirects Calls to an Active SIP User Terminal
Presence Server	Stores and Distribute Users' Presence Information

The All-in-One IP-PBX Server

Embedded Registrar, Proxy, Presence and Voice Mail Server

Back-to-Back User Agent (B2BUA)

- Basic Call Services like Call Forwarding, Call Transfer Necessitates Call Management and Tracking for Entire Call Duration
- SIP Server With B2BUA Becomes an Active Participant in a SIP Call, Enabling Many Advanced Services in Addition to Basic Telephony Services
  - ✓ Applications Such as Billing Require Call State Monitoring
- Facilitates Centralized Call Management

Multiple SIP Trunk Support

- SAPEX Supports Registration With Multiple SIP Trunks
- Registration With a Maximum of 10 SIP Servers is Supported
- While Making an Outgoing Call, System Selects a SIP Trunk as per the Call Routing Algorithms

Seamless User Connectivity

- A SIP User Located Over Public/Private Network Can be Registered Easily With SAPEX
- A Remote User’s Connectivity is Maintained Even When Behind a NAT or Firewall
- The Embedded Dynamic DNS Client Ensures Round-the-Clock User Connectivity
- A User Can Have Multiple Contact Points (Terminals), Mapped to a Common User Identity
- A User Can be Called on Any/All of the Active Terminal at a Given Instant

Dynamic DNS (DDNS)

- Automates the Discovery and Registration of the Server's IP Address on the Public Network
- Fluctuating Dynamic IP of the Server is Mapped to a Unique Domain Name (DNS)
- Remote Administrator and IP-PBX Clients Can Connect to the Server Using the Non-Altering DNS
- Benefit:
  - ✓ Prevents Reconfiguring of Systems Every Time a Network Infrastructure Changes
  - ✓ Useful When Remotely Accessing the IP-PBX Connected to Ones Home or Office IP Connectivity, Usually Configured for a Dynamic IP

Presence and IM

- Before an Actual Conversation Begins, Presence Determines: :
  - ✓ The Availability of a User (Such As Online, Offline and Others)
  - ✓ His Willingness to Participate in a Communication Session (Busy, Available on Phone, Out of Office And Others)
  - ✓ His Preferred Mode of Communication (Call or Instant Messaging)
- The Presence Server Maintains and Distributes Presence Information of Users Registered With the IP-PBX
- Instant Messaging (IM) is a Popular Mode of Real-time Communication
- Knowing the Availability Status of Users, Reach a Right Contact, In Right Time and on the Right Terminal

Management Features

- Enhanced Administrative Ease
- World-Wide Portability of Extensions
- Busy Lamp Field (BLF)
- Call Detail Records (CDR)
- RADIUS Client
- Call Management

Enhanced Administrative Ease

- Route Calls Also Over IP Network, Besides Data
- Embedded Web Server and Web Based Programming GUI
- With a Few Clicks From the Intuitive GUI (Jeeves) Itself:
  - ✓ An Administrator Can Remotely Program and Configure the System
  - ✓ Administrator Can Monitor and Manage User Registration, Feature Access in Real-time
  - ✓ A New User Can be Added
  - ✓ User's Calling Rights Can be Defined
  - ✓ User Groups Can be Formed
  - ✓ Voice Mail Can be Allocated

Extension Portability

- In IP Telephony, a VoIP Call is Established via IP Network
- Instead of a Phone Number, An IP User is Identified by SIP URI
- An IP User is Free to Port his Extension Anywhere on the IP Network
- Users Can Establish Calls Retaining the Same Contact Credentials, though Registering From Varied Locations
- Unlike Traditional Telephony System, SAPEX Does Not Bind the User to a Fixed Location

Busy Lamp Field (BLF)

- A Busy Lamp Field is An Array of Extension Status Lamps
- An Extension Can Bear Different Statuses Such as:
  - ✓ Busy, Ringing or Idle
- If a User's Class of Service (Cos) is Provisioned, An Extension User Can Monitor the Status of Another Extension
- Based on Extension Status, the Operator Can Decide to Either Transfer a Call Directly or Else Pick Up a Call Incase Called Extension is Busy

Call Detail Records (CDR)

- Call Details are Stored in the System's CDR Buffer
- The Call Log is Stored for Different Types of Calls
  - ✓ Internal Calls Made Between System Users
  - ✓ External Calls Made Between System User and External User
- The IP-PBX Can be Configured to Send CDR Text-File as Email Attachments

Call Detail Records (CDR)

- CDR Report Details:
  - ✓ Calling/Called Number
  - ✓ SIP Trunk Used for External Calls
  - ✓ Date and Time of Call
  - ✓ Call Duration
  - ✓ Call Termination Cause

RADIUS Client

- SAPEX Logs the Details of Calls in CDR Files
- These CDR Files Contain Essential Information to Account a User for the Services Utilized by Him
- These CDR Files are Therefore Requiring a Safe and Longer Storage
- The Embedded RADIUS Client Facilitates Efficient Call Logging to a Remote Server/Database

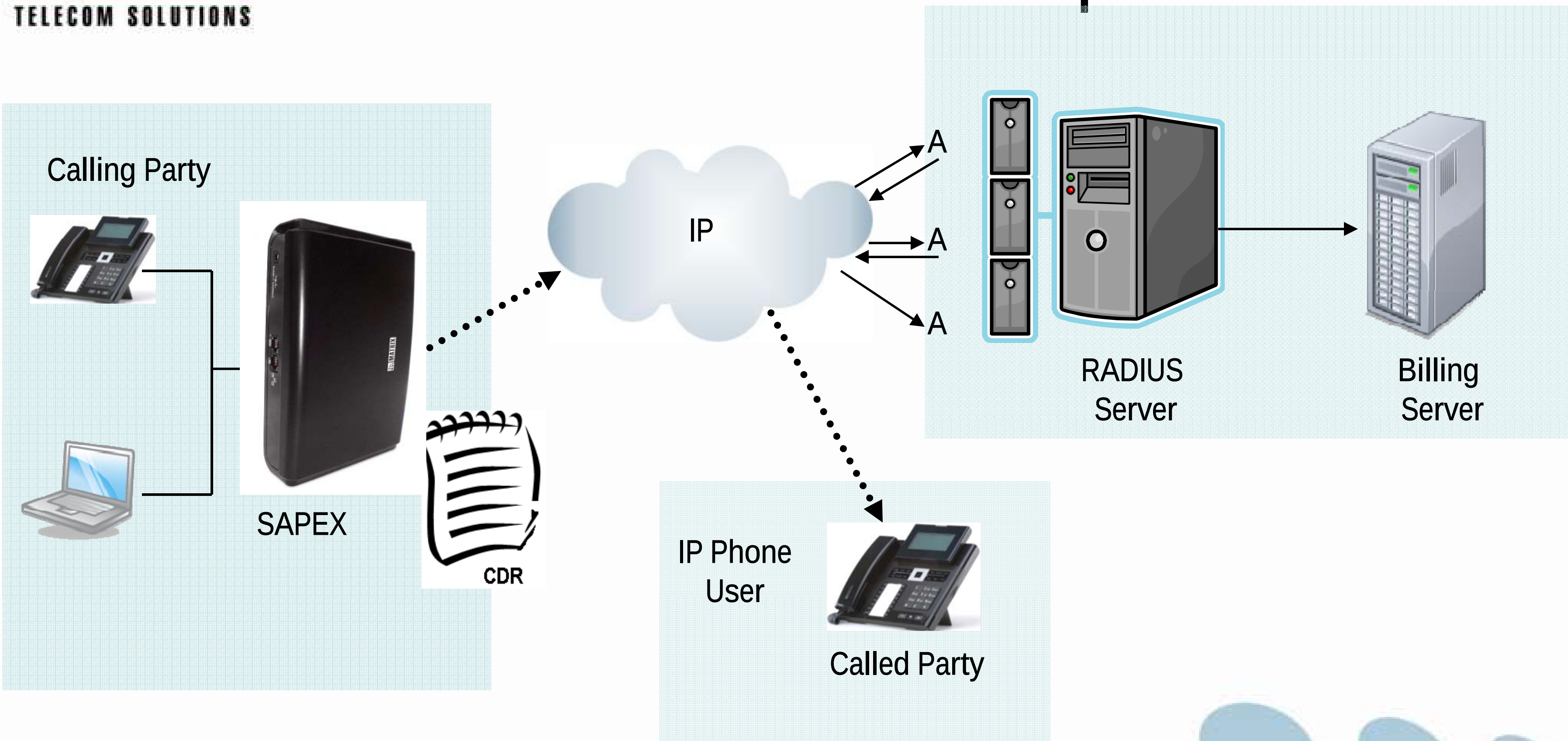
RADIUS Client

- RADIUS : Remote Authentication Dial In User Service
- Enables the IP-PBX to Send the CDR Files to a Centrally Managed Remote Server Called RADIUS Server/Database
- It Employs an Authentication, Authorization And Accounting Client-Server Protocol (AAA Client-Server Protocol)
- Further Integration With a Billing Server, Can Benefit the Service Providers in Accurate Billing

AAA Procedure

- Authentication
  - ✓ Validating a User
- Authorization
  - ✓ Defining Permissible Services for a User
- Accounting
  - ✓ Keep a Track of Resources Utilization by a User for the Purpose of Billing and Monitoring

RADIUS Client-Basic Operation



- Authentication
- Authorization
- Call Established
- Accounting

Call Management

- Incoming Call Management Features:
  - ✓ Anonymous Call Rejection
  - ✓ Auto Attendant
  - ✓ Caller-ID Based Routing
  - ✓ Do-Not-Disturb
  - ✓ DDI Routing
  - ✓ Time Tables
  - ✓ User Groups

Call Management

- Outgoing Call Management Features:
  - ✓ Allowed/Denied Number List
  - ✓ Automatic Number Translation
  - ✓ Dial Plans
  - ✓ Emergency Number Dialing
  - ✓ Peer-to-Peer Calls
  - ✓ Reverse DDI

Call Management

- Telephony Services:
  - ✓ Call Forward
  - ✓ Call Hold
  - ✓ Call Pickup (Selective and Group)
  - ✓ Call Park (Personal Orbit), Call Retrieve (Personal and Global Orbit)
  - ✓ Call Transfer (Blind and Attended)
  - ✓ CLIR
  - ✓ Conference

Anonymous Call Rejection

- An Incoming Call Without Caller Line Identification (CLI) Number is Termed as Anonymous Call
- Instead of Number, the term “Anonymous” is Displayed on the Screen
- Offers the Flexibility to Directly Reject Anonymous Calls or Route Such Calls to a Specific Extension

Auto Attendant

- The Auto Attendant Informs the Caller of the Way to Reach His Ultimate Destination
- Customized Welcome and Guiding Prompts as per Time of the Day, Music-on-Hold Can be Played to a Caller
- With the Automated Attendant, a Caller Can Find-his-way to:
  - ✓ Reach to a Desired Extension
  - ✓ Retrieve Information
  - ✓ Leave Back a Message in the Mail Box of the Called Extension

Voice Mail Features

- Configurable Voice Mail Server Size
- Individual Voice Mail for Each User
- Configurable Mailbox Size
- Customizable Greetings
- Voice Mail Notification by E-mail Using SMTP

Caller-ID Based Routing

- Calls Get Routed to Pre-defined Extensions as per CLI of Calling Party
- A Calling Party Number Table Can be Programmed
- The CLI of Calling Party is Tallied With the Table Entries
- Calls are then Routed to Defined Stations as per Routing Groups
- Depending on the Time, Call Can be Routed to Different Destinations
- For example:
  - ✓ Calls Important to Business May be Directed to Higher Authorities
  - ✓ Calls With Specific CLI May be Directed to Particular Departments
  - ✓ While Calls From Anonymous Numbers May be Directed to the Customer Support Teams

Do-Not-Disturb

- Do-Not -Disturb (DND) Feature Offers Users With the Flexibility of Not Receiving Calls for Particular Time Period
- Outgoing Calls Can be Made When DND is Enabled

DDI Routing

- A Call Landing on SIP Trunk Can be Directly Routed to an Extension as per the DDI Numbering
- The DDI Facility Should be Activated on the SIP Trunk by the SIP Service Provider
- Unlike Traditional Telephony Services, IP Telephony Does Not Bind a Number to its Geographical Location
- Calls are Placed Over Internet and Numbers are Mapped to IP Addresses, Which May be Anywhere on the Internet
- An IP Extension Can Always be Called Irrespective of Its Current Location

Time Table

- Route Incoming Calls as per 'Time of the Day' (Time Zone)
  - ✓ Defined Schedule for a Day is Called Time Table
  - ✓ A Timetable Divides Entire Day in to Different Time Zones
- 4 Such Timetables Can be Defined
- Provides Flexibility to Receive Ones Calls on Different Terminals as per the Time

User Groups

- Multiple Extensions Can be Clustered as One Group
- Defining User Groups, Calls Can be Distributed Between the Group Members
- The Group Members Can be Located at Different Geographic Locations
- On Reception of a Call, the Extensions Ring as per the Assigned Priorities
- Thus Ensuring No Call Remain Unanswered

Allowed and Denied Numbers

- Allow/Deny Dialing of Specific Numbers
- Avoids Misuse and Restricts Unproductive Calls
- 16 Such Allowed and Denied Number Lists Can be Programmed

Automatic Number Translation

- SAPEX Supports Registration of Multiple SIP Trunks
- These Trunks Can be Availed From Single or Multiple ITSPs
- While Placing a Call, a Caller is Not Conscious of the Routing Logics Defined and the SIP Account in Use
- SAPEX Itself Modifies the Dialed Number or Part Thereof, Such that it Matches With the Numbering Plan that Is Understood by the ITSP
- For Example:
  - ✓ If a User Dials 223344 to Call www.abc.com
  - ✓ SAPEX Adds the Appropriate Access Code "*777" Specified by the ITSP and Dials-Out the Number “*777223344” Instead of 223344

Dial Plans

- SAPEX Supports Multiple SIP Trunk Registrations
- Registration With Maximum of 10 SIP Servers is Supported
- Calling Rates Differ on the Basis of Area of Call, Service Provider, Call Time, etc
- A Dial Plan Allows a User to Place Call through the Most Cost-effective SIP Trunk, as Per a Defined Call Routing Logic
- Each User Can be Assigned Multiple Dial Plans
- The Dial Plans May be Same For All Users or May Differ Individually

Emergency Number Dialing

- Emergency Calls are Not Subjected to Outgoing Call Management Rules
- This Reduces Any Latency While Placing Emergency Calls
- SIP Trunk Can be Specified for Such Calls
- Four Such Numbers Can be Programmed

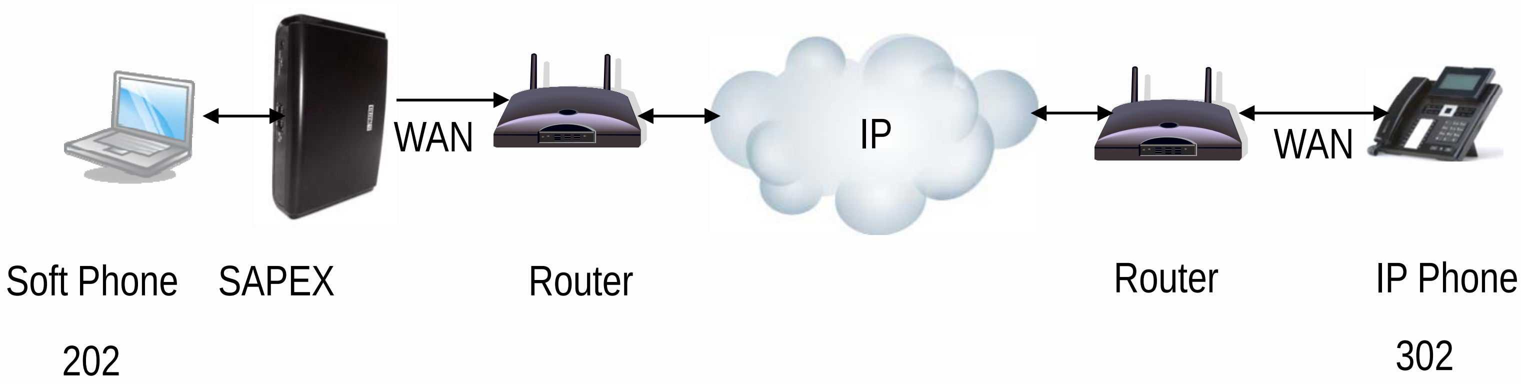
Peer-to-Peer Calling

- Calls Can be Placed Between Two SIP Devices, Without Going through a Proxy Server
- Fixed IP Addresses of Various SIP Devices Can be Programmed in a Peer-to-Peer Table
- 500 Such Entries Can be Programmed

Peer-to-Peer Calling Table

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

Peer-to-Peer Calling



- Shorter Extension Codes Can be Defined in Place of Long Number Strings/Addresses
- Call Gets Routed via the Public IP Network

Reverse DDI

- This Feature Activates Carting of DDI Numbers as Caller ID When a User Makes a Call via SIP Trunk
- When a Call is Received From the Called Party, the Call Can be Directly Routed to the DDI Number Which Placed the Call
- Eliminates the Hassle of Searching the DDI User Who Made the Call
- Saves Time and Enhances Productivity

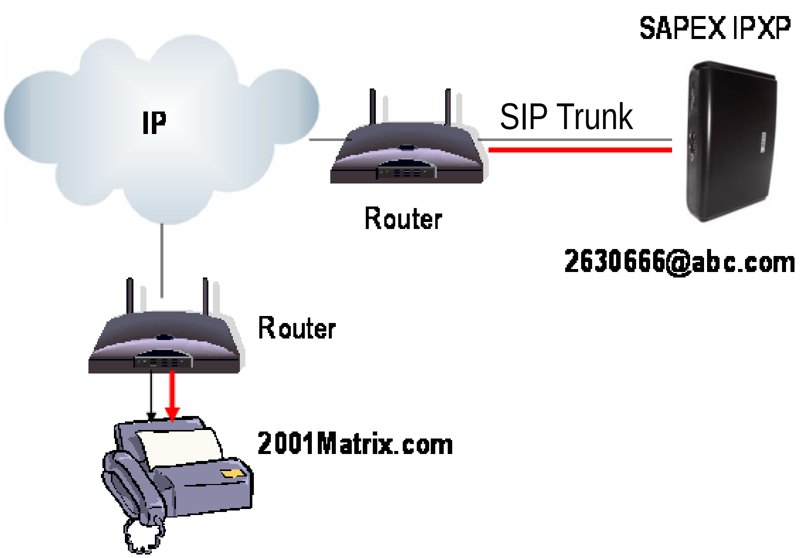
FAX Homing

- FAX Homing Allows a SIP Trunk to be Used for Both:

  Voice Calls and to Receive FAX

  ✓ System Detects FAX Tone on SIP Trunk Only When Call is Answered by the Auto Attendant

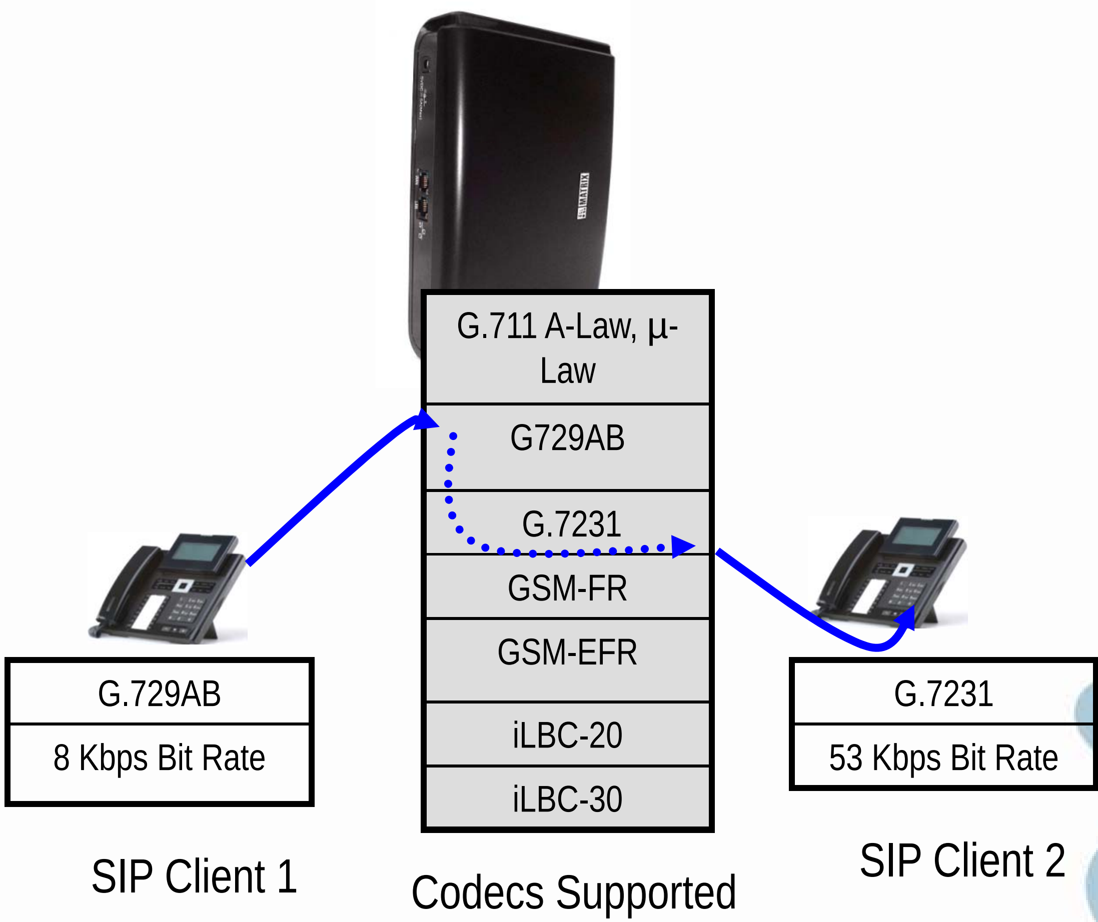
  ✓ When FAX Tone is Detected, System Routes Call to the Extension Where FAX Machine is Connected



Voice Transcoding

- There is Diversity Among Available SIP Endpoints (Terminals) and their Capabilities
- Audio Transcoding Helps to Establish Communication Between SIP Devices With Diverse Audio(Codec) Specifications
- Reduces the Ratio of Dropped Calls

Voice Transcoding



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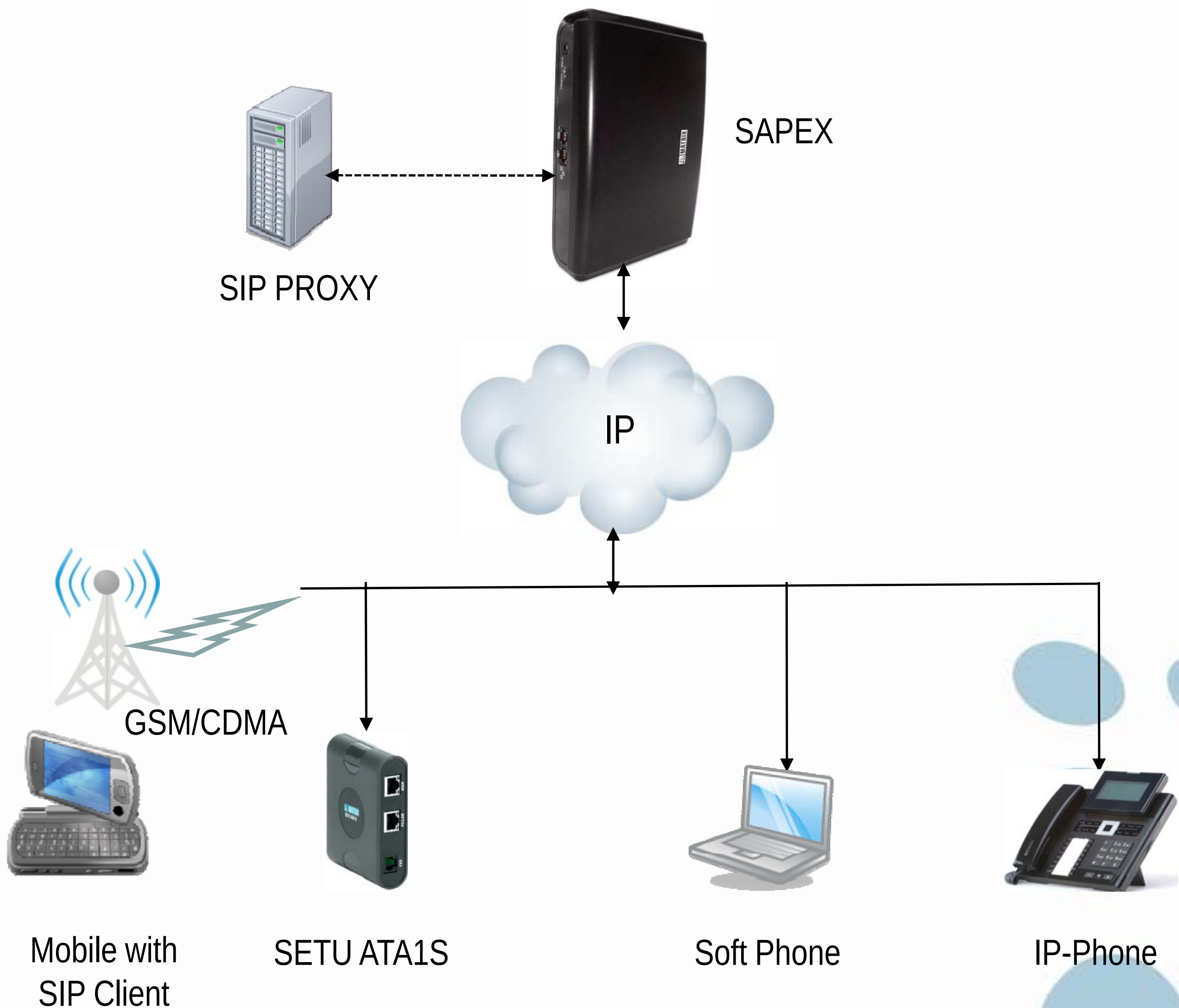
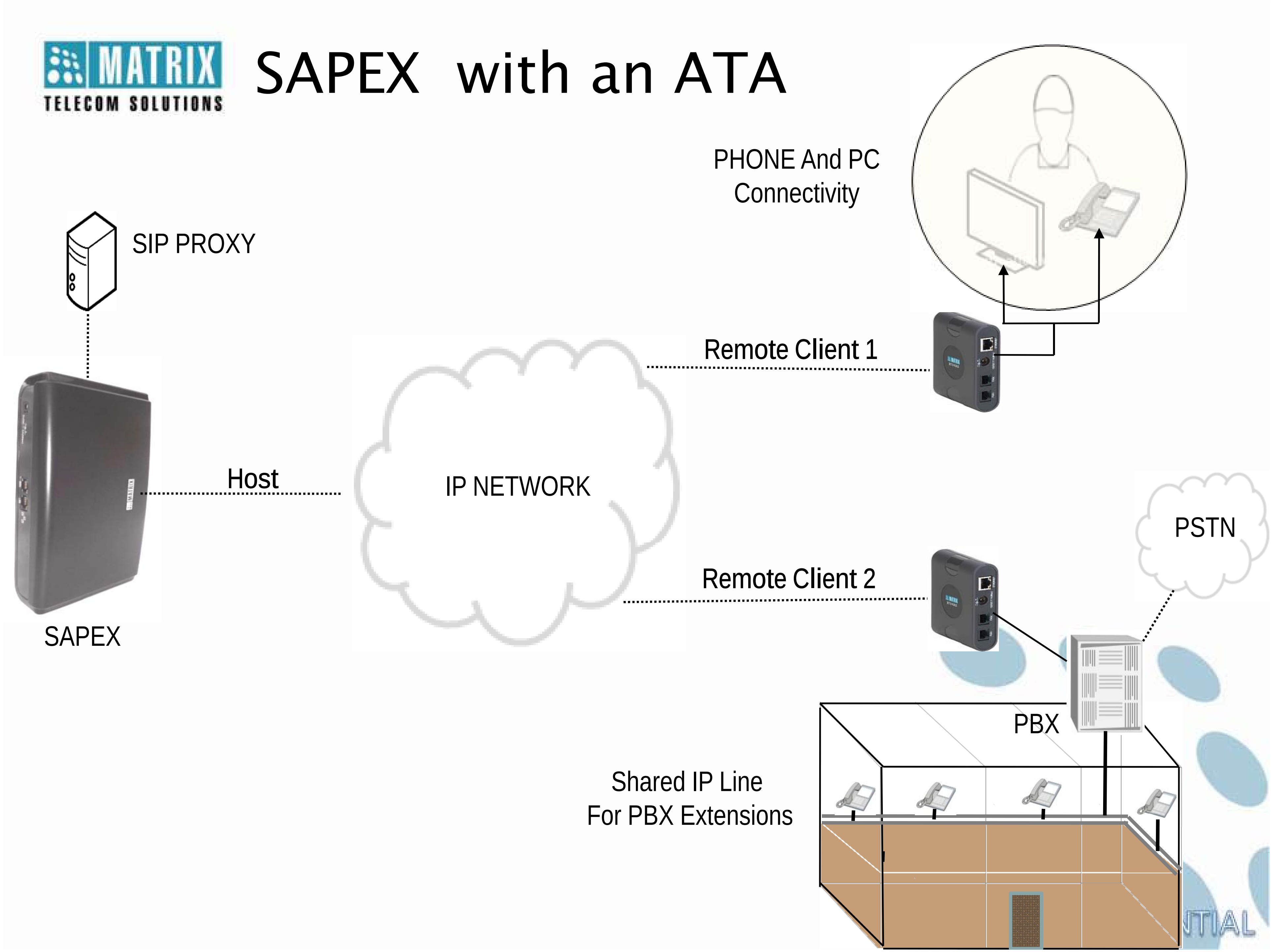
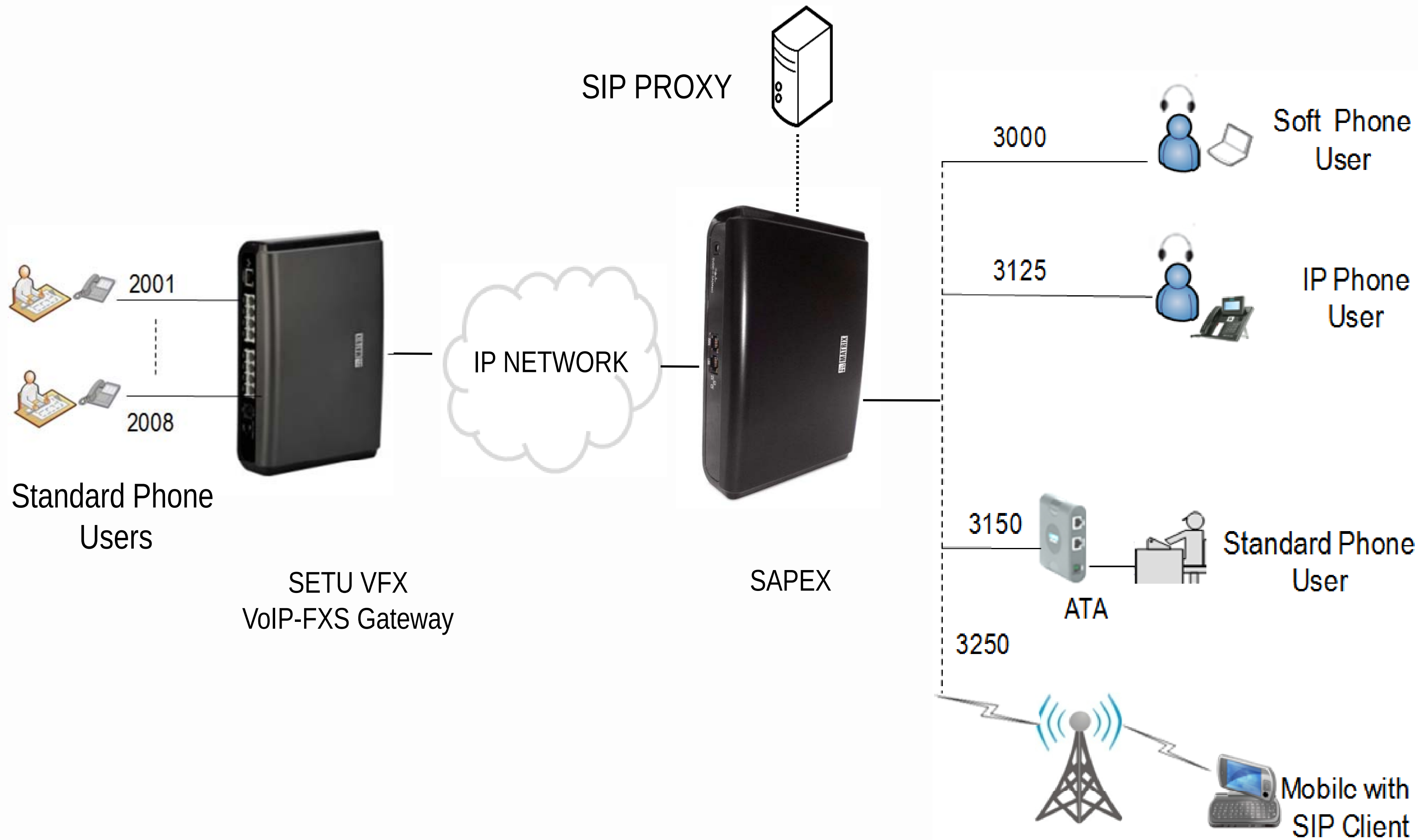
Applications

SAPEX Stand-Alone Application

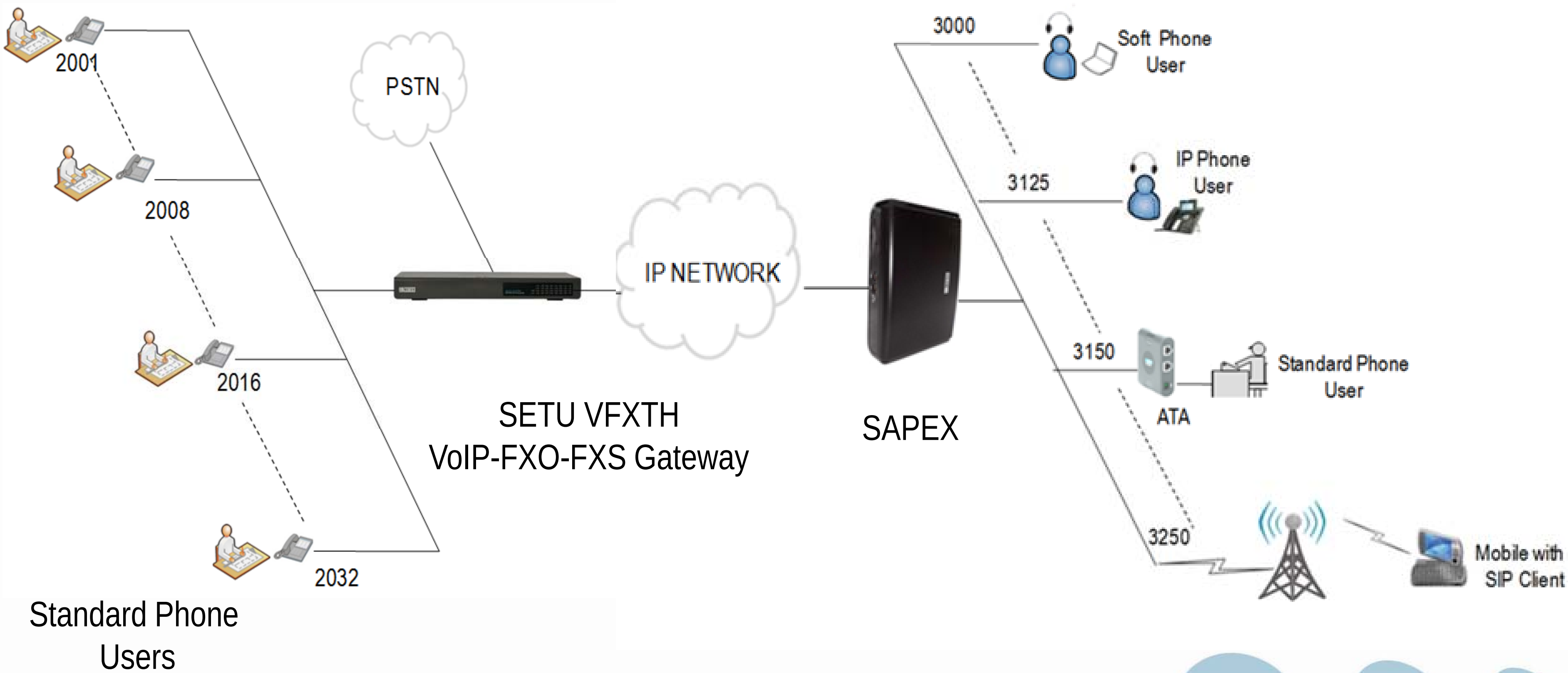



MATRIX
TELECOM SOLUTIONS
SAPEX with an ATA
PHONE And PC
Connectivity
SIP PROXY
Remote Client 1
Host
IP NETWORK
SAPEX
PSTN
Remote Client 2
PBX
Shared IP Line
For PBX Extensions

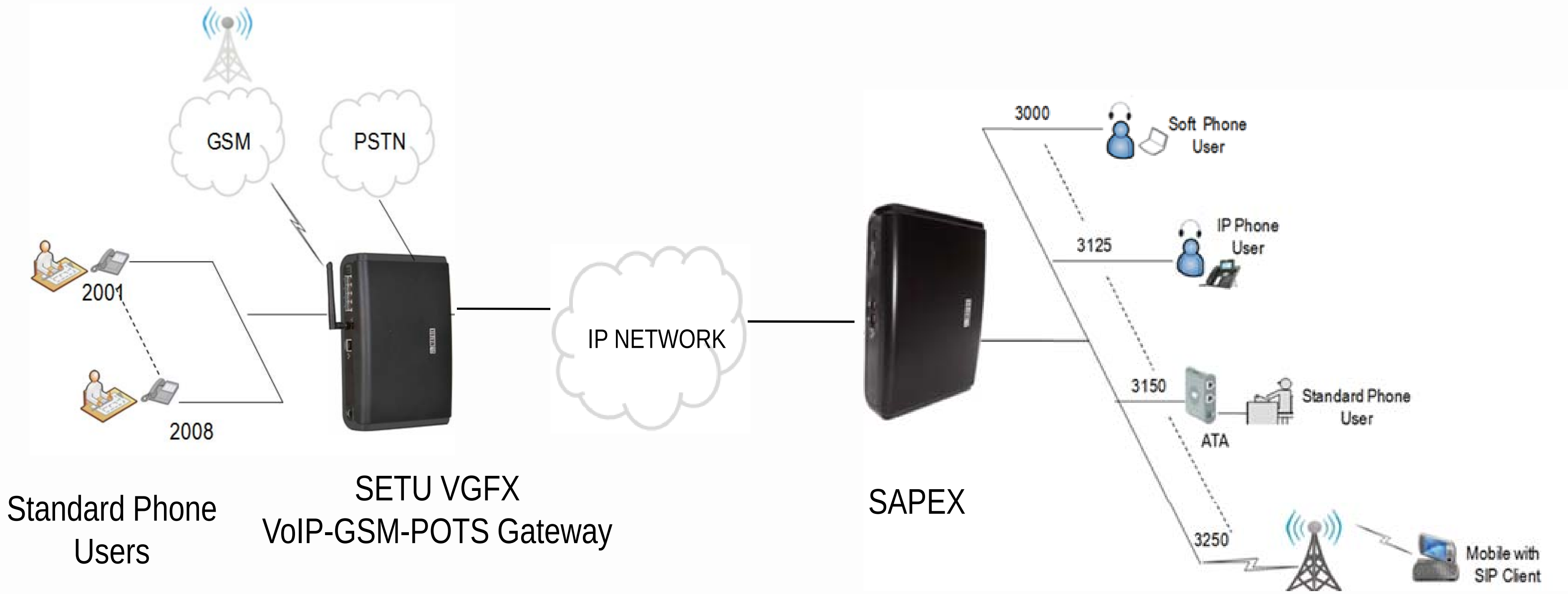
SAPEX with VoIP-FXS Gateway



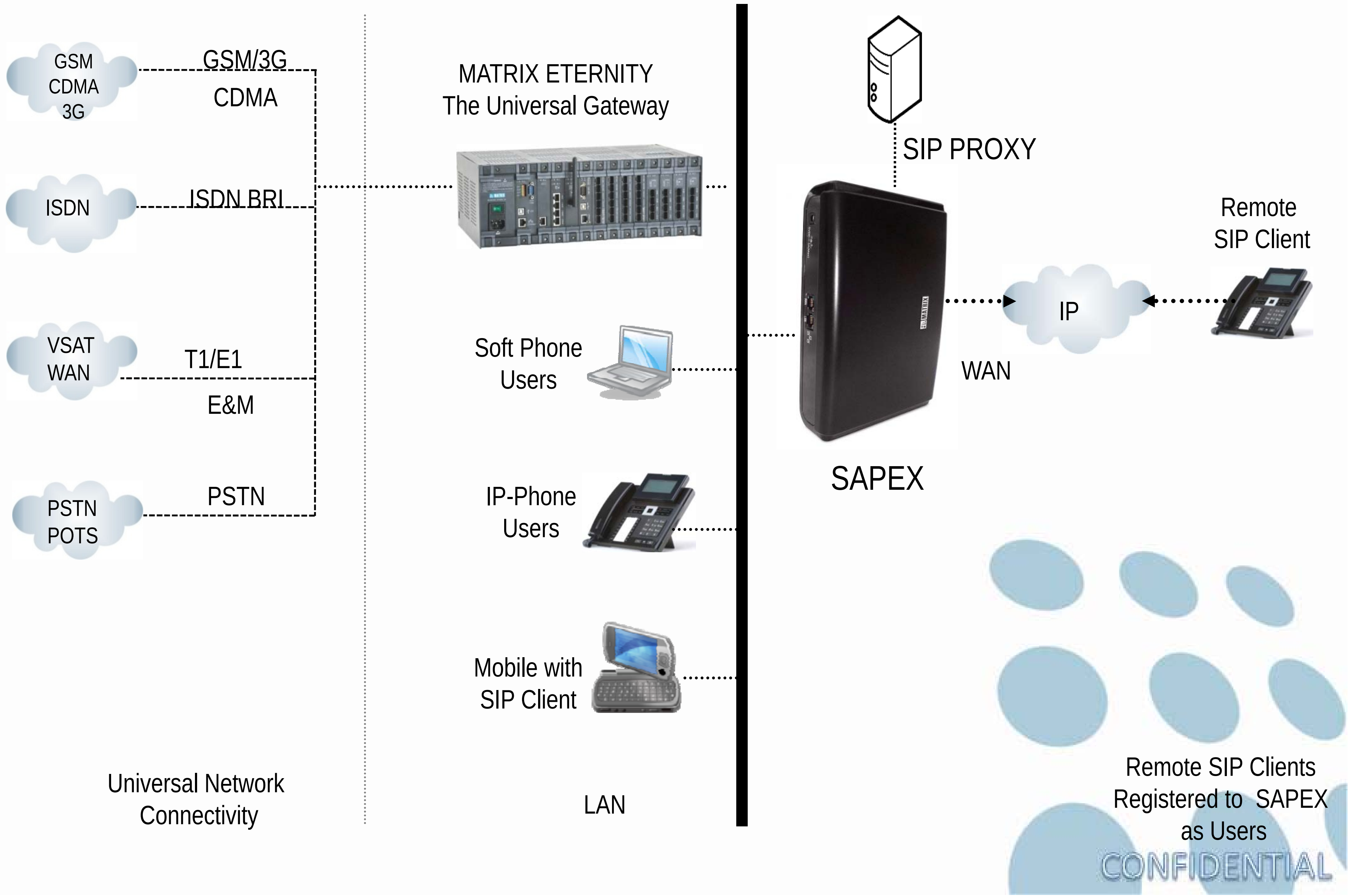
SAPEX with VoIP-FXO-FXS Gateway



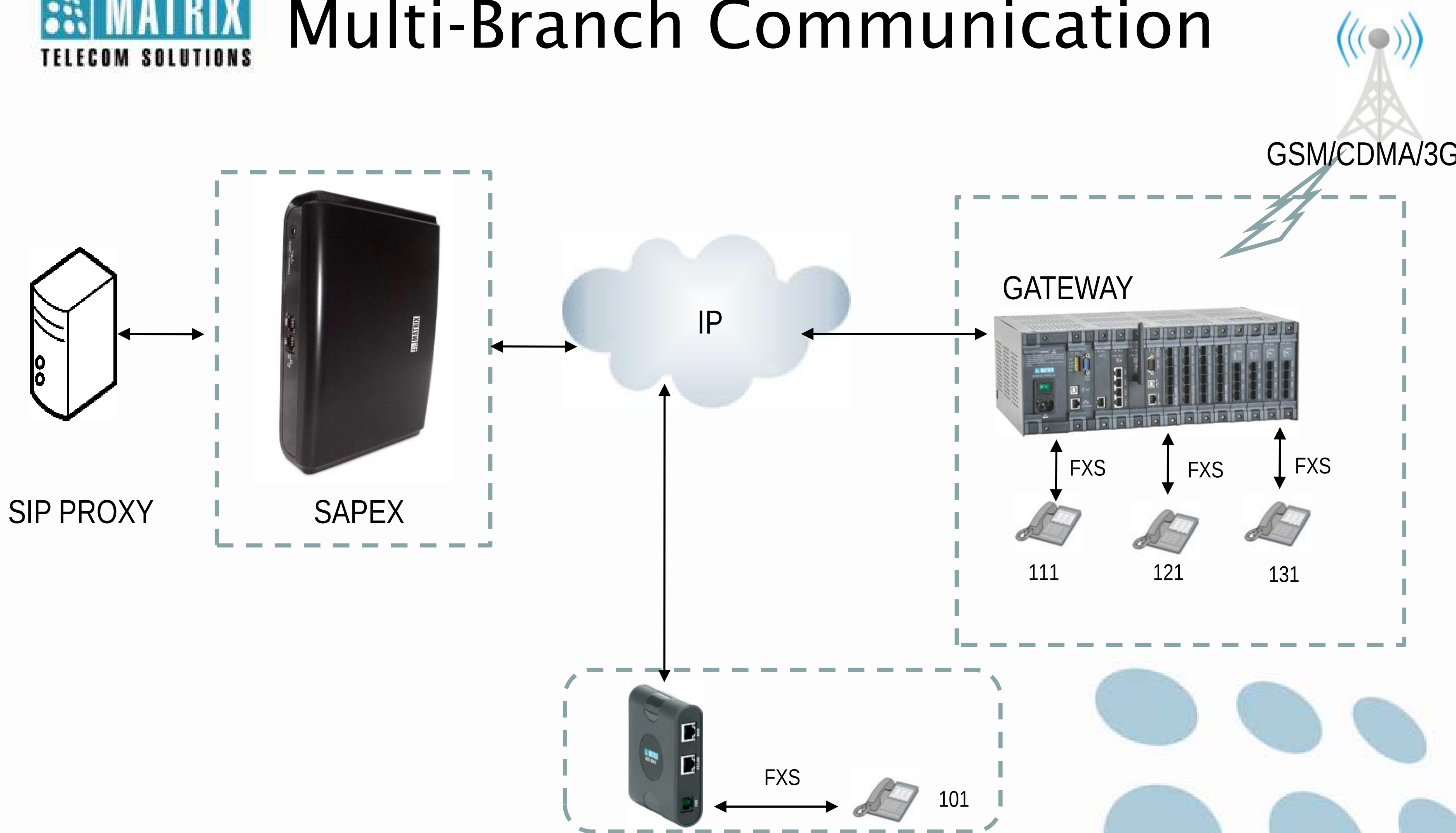
SAPEX with VoIP-GSM-POTS Gateway



SAPEX IP-PBX with Universal Gateway



Multi-Branch Communication



System Capacity

RESOURCE	SAPEX SDM	SAPEX DDM
DSP Modules	Single	Dual
Users	200	500
SIP Trunks	10	

Concurrent Calls (Transcoding)

CODEC	SAPEX SDM	SAPEX DDM
G.723L/H	15	30
G.729	16	31
GSM EFR	13	26
GSM FR	21	30
iLBC 20/30	13	26

Hardware Specifications

PORTS	
WAN Port	1(RJ45, 10/100/1000 Base T)
LAN Port	1(RJ45, 10/100/1000 Base T)
USB Port	1 (Internal)
DC Jack	1 (DC Power Jack)
LED Indications	1 for Power Status and 1 for SIP Trunk Status
Power Supply	External Adaptor 5V DC / 70A
Power Consumption	20 W (Maximum)
Dimensions (WxHxD)	230mmX55mmX163mm (9.06”X2.17”X6.42”)
Installation	Wall Mount and Table-Top

System Ports

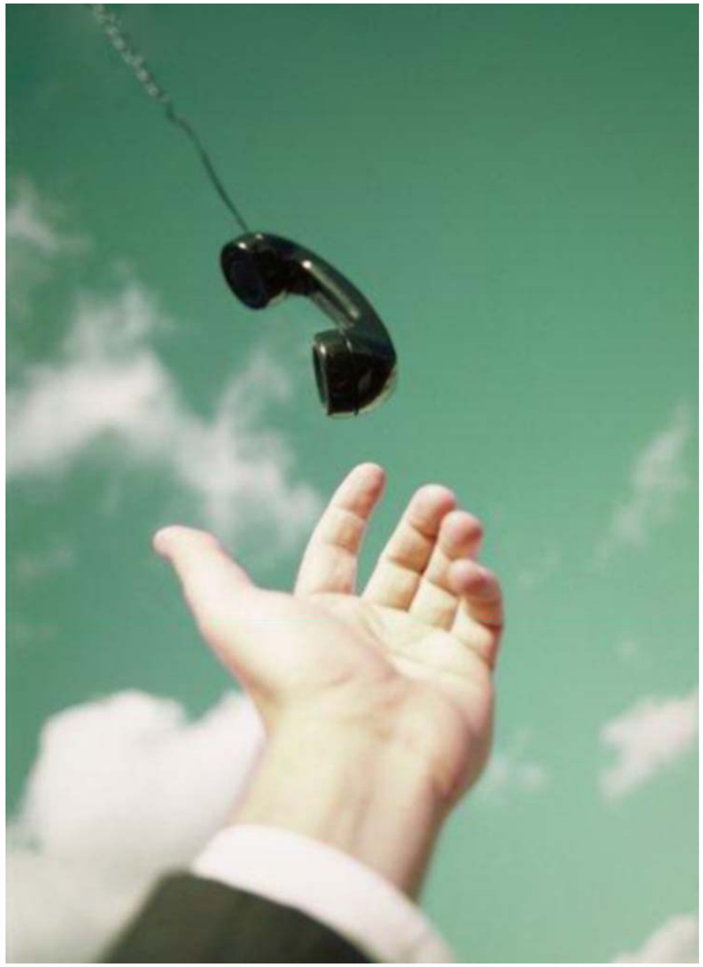
Power Adaptor

WAN Port

LAN Port

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

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For Further Information Please Contact:

Email ID: mt.pbx@matrixcomsec.com

Mobile: +91 9662544401

Visit us at www.MatrixComSec.com