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TELECOM SOLUTIONS

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Manufactured in
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50+ Countries | 1M+ Customers

Warm Welcome

Matrix SETU VFX

Multi-port VoIP-FXO-FXS Gateways

SETU VFX

Presentation Overview

- Introduction
- Interfaces
- Variants
- Target Customers
- Applications
- Key Features

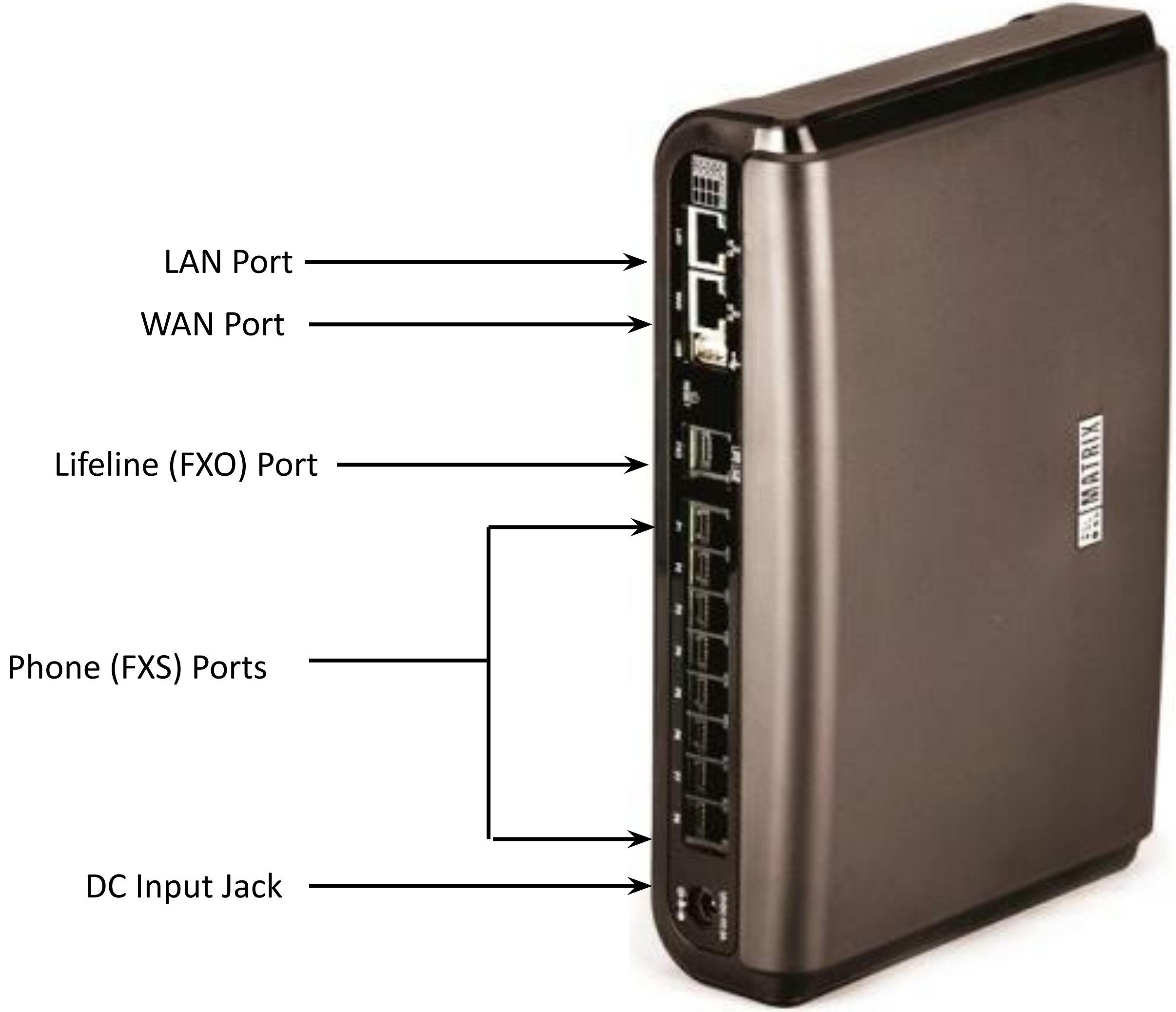
Introduction

SETU VFX
Multiport VoIP-FXO-FXS Gateway

- SETU VFX links VoIP networks to traditional telephony and vice-versa
- It can be interfaced with PBX or GSM FCTs having FXO Port
- Eliminates the need to upgrade an existing PBX for making IP calls

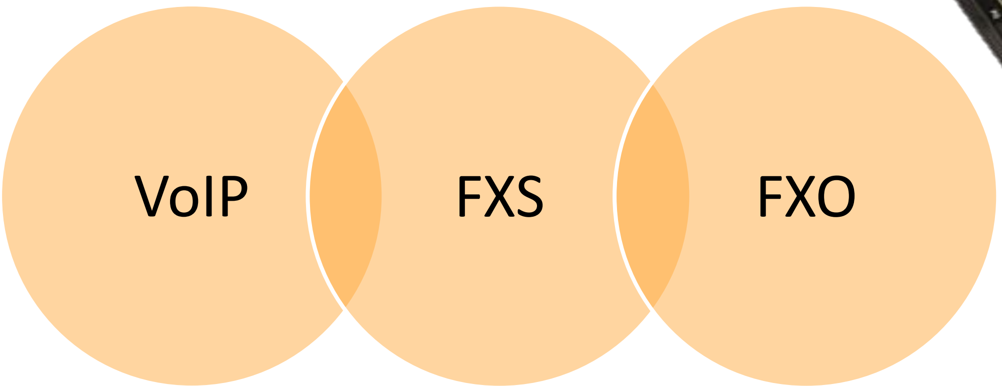
Multi-port VoIP-FXO-FXS Gateway

Interfaces



Multi-port VoIP-FXO-FXS Gateway

Product Variants



SETU VFX880
8 VoIP Channels
8 FXO

SETU VFX808
8 VoIP Channels
8 FXS

SETU VFX 404
4 VoIP Channels
4 FXS

SETU VFX 440
4 VoIP Channels
4 FXO

Target Customers

Corporate Branch Offices

Organizations with Field Staff

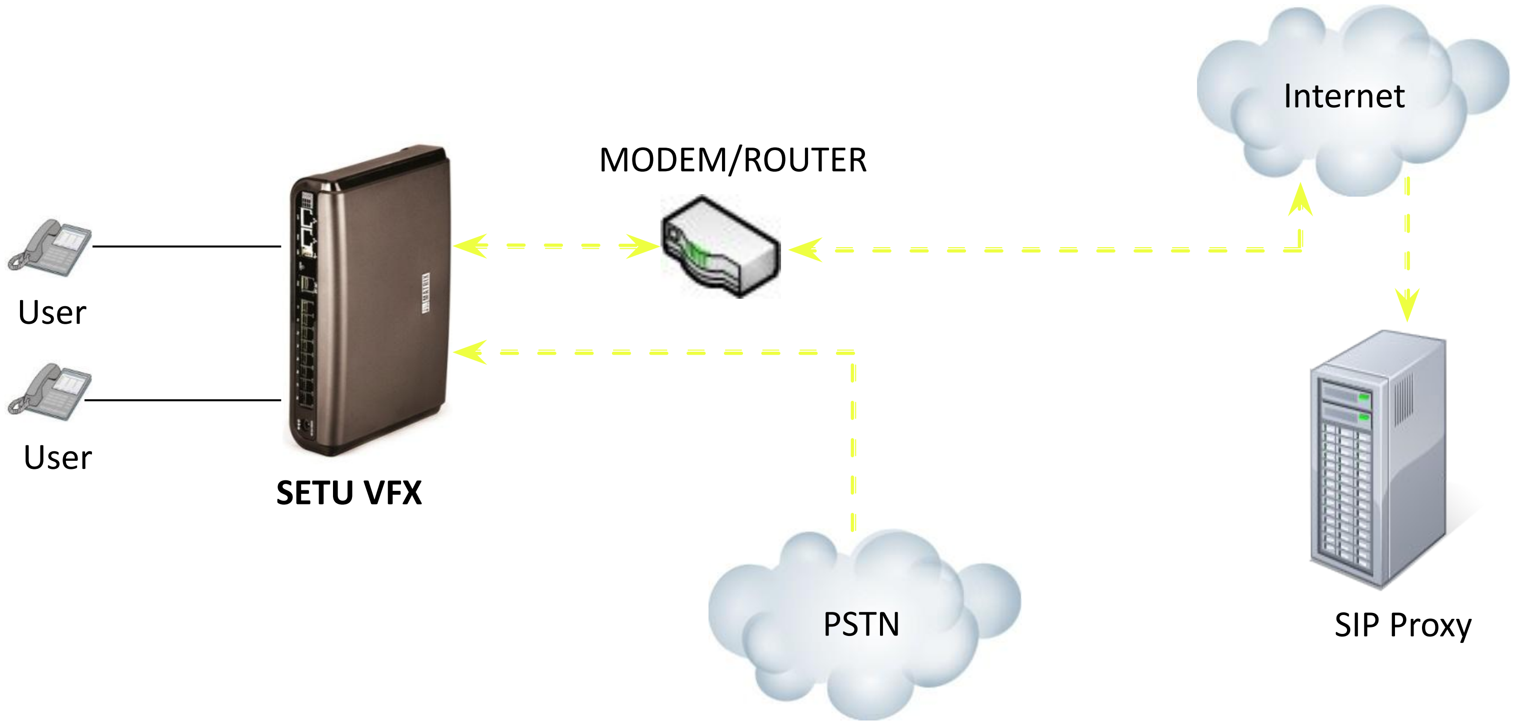
Remote Project Sites

Call Centers

Applications

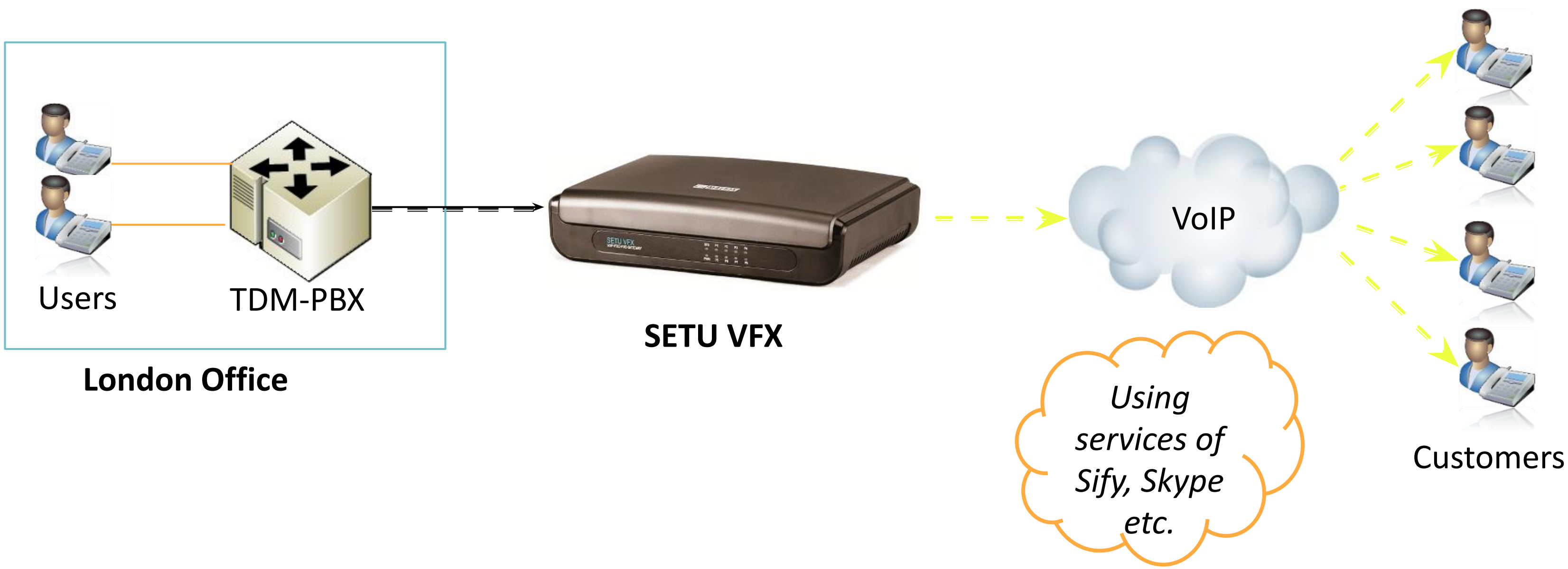
Stand-alone Application

Built-in PBX Functionality



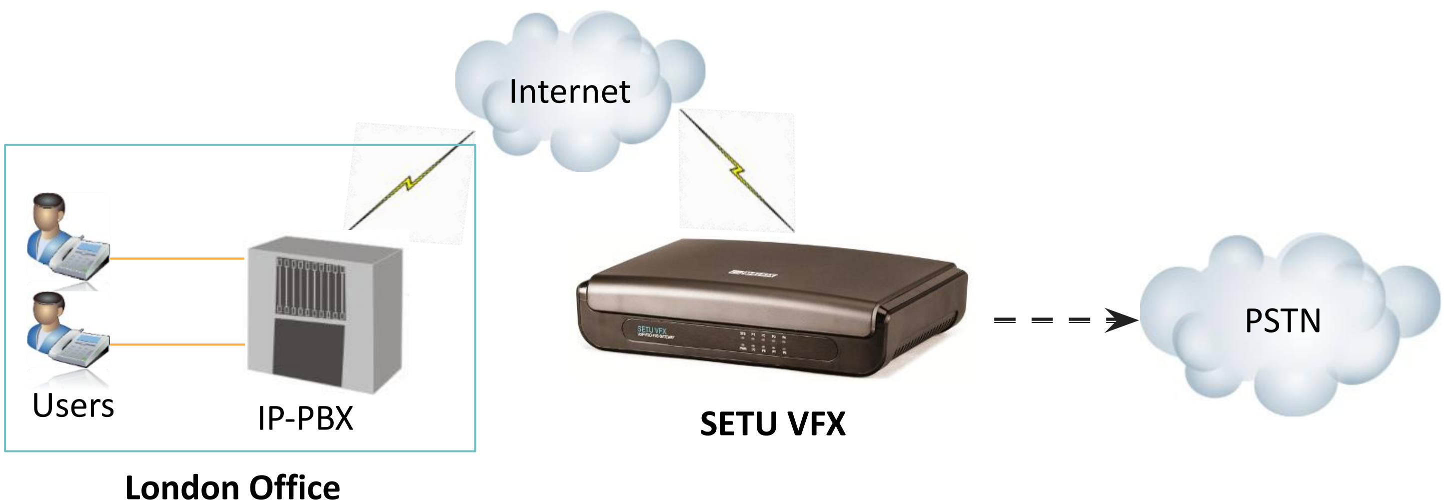
VoIP Gateway for Legacy TDM-PBX

- Use legacy TDM infrastructure to make cost effective VoIP calls
- Maintain existing dialing habits
- Increase network connectivity options



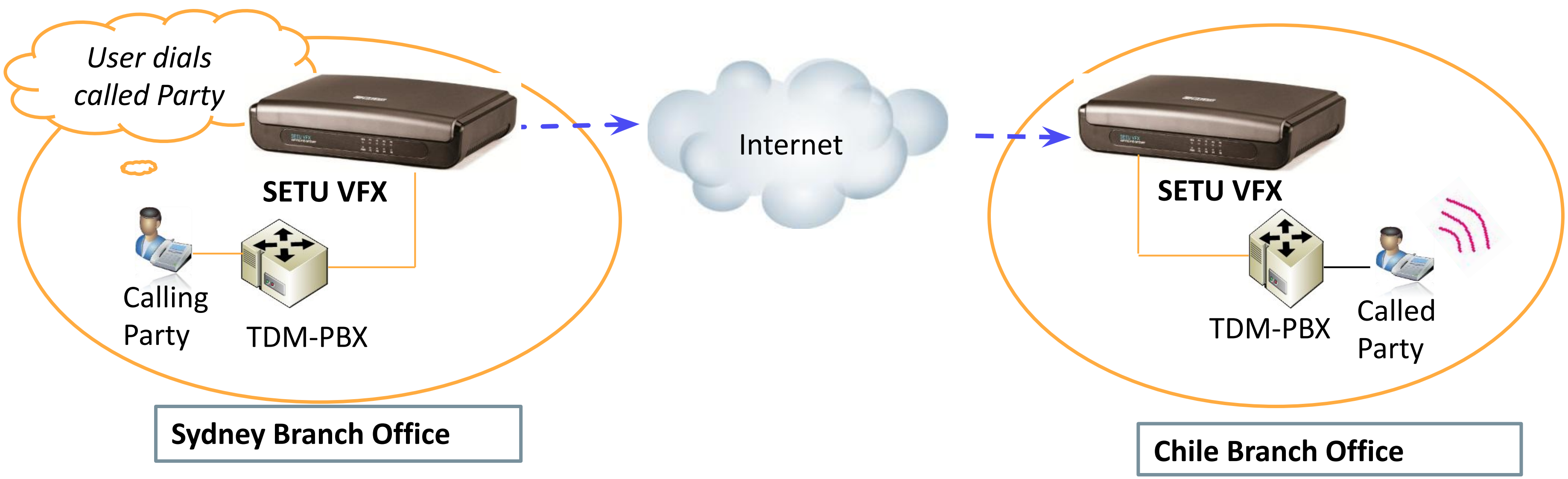
VoIP Gateway for IP-PBX

- Access PSTN from VoIP infrastructure
- Maintain existing dialing habits
- Deployable with any SIP based IP-PBX, Softswitches and Hosted Services



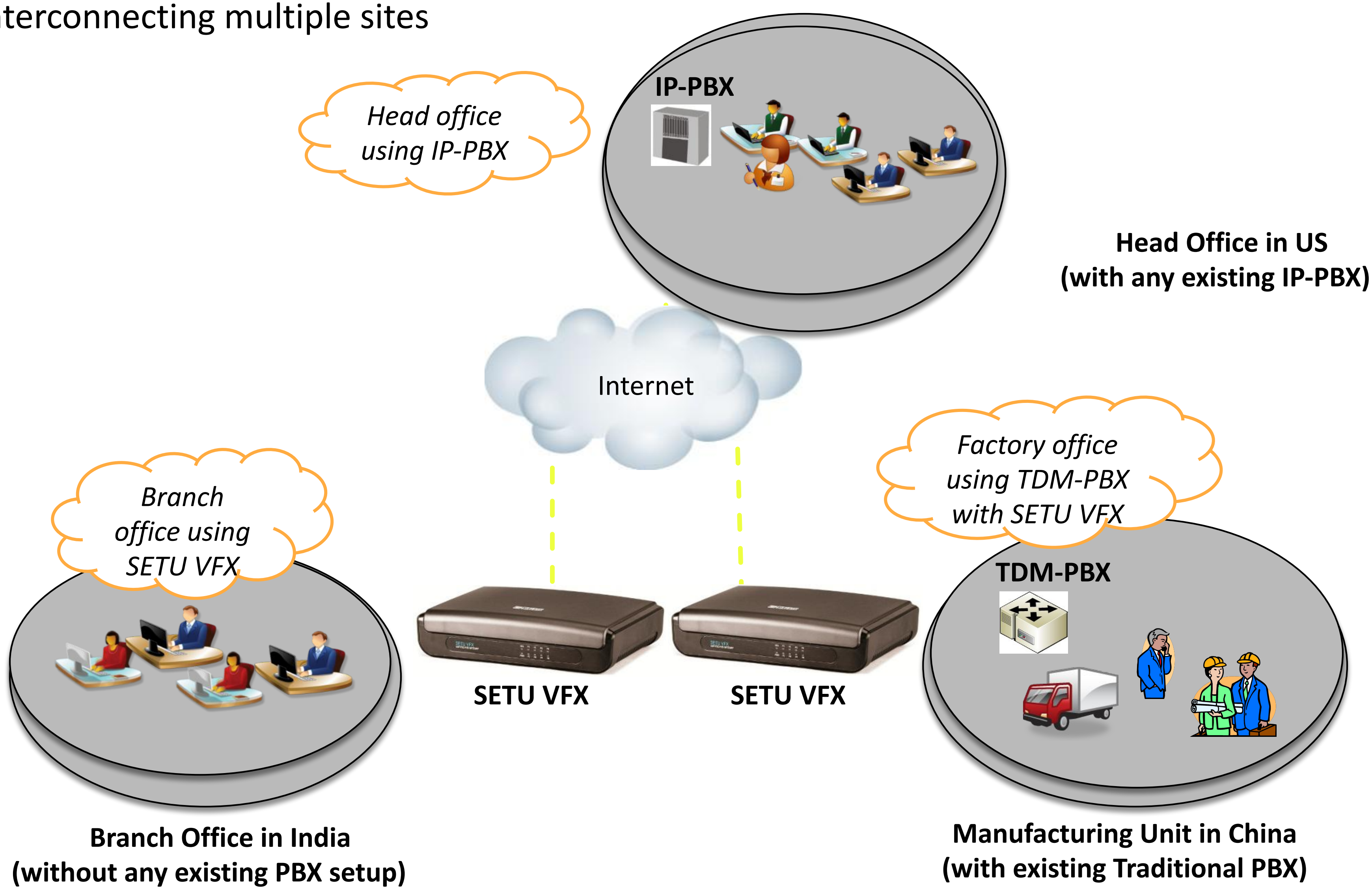
Peer-to-Peer Calls

- Connect remote branch offices over VoIP
- Make free VoIP calls between branch offices
- Increase workforce collaboration



Multi-Site Connectivity

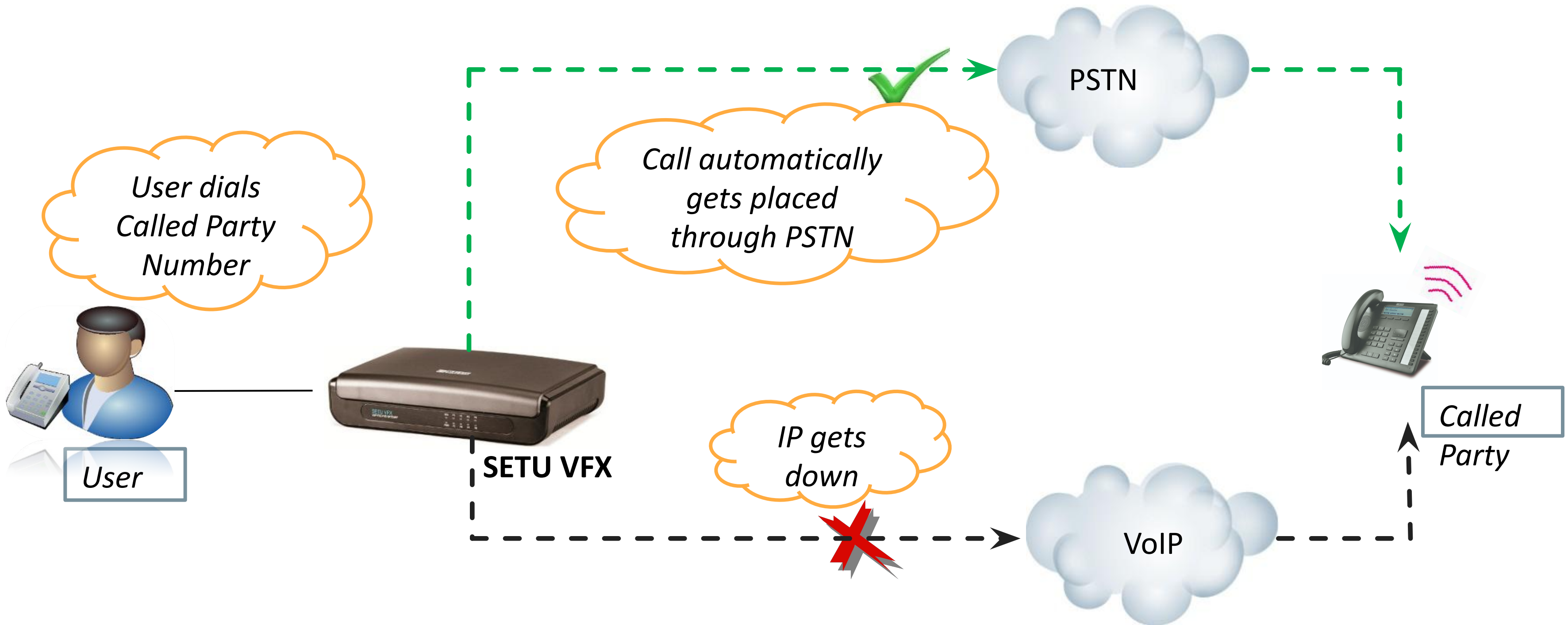
Interconnecting multiple sites



Key Features

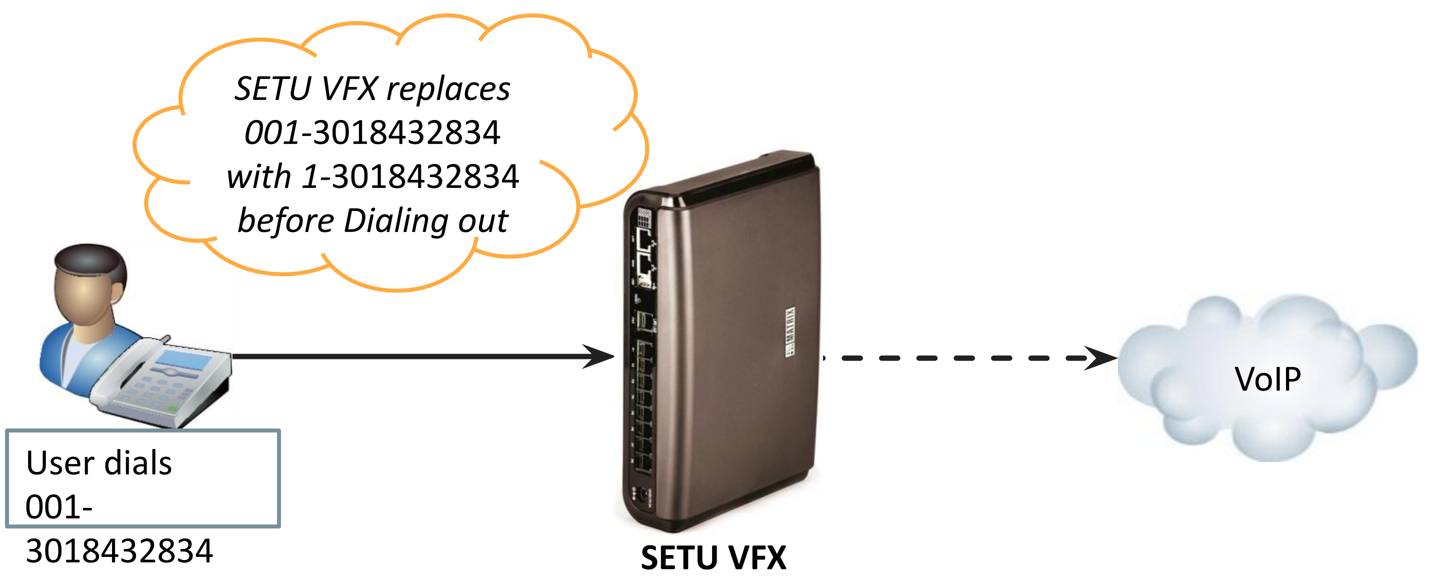
Auto PSTN Fallback

- Continuous network connectivity
- Reduces dependency on a single network
- More reliable communication



Automatic Number Translation

- Translates complete number or a part of a dialed number to match with the numbering plan of the destination network
- Automatic Number Translation (ANT) is supported on SIP
  - ✔ User dials 001-xxxx to reach a number in 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 Progress Tones and Rings

- Different Tones can be configured to indicate the progress of a Call Activity
  - ✔ Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Feature Tone and Confirmation Tone
- 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

Caller Line Identification and Presentation

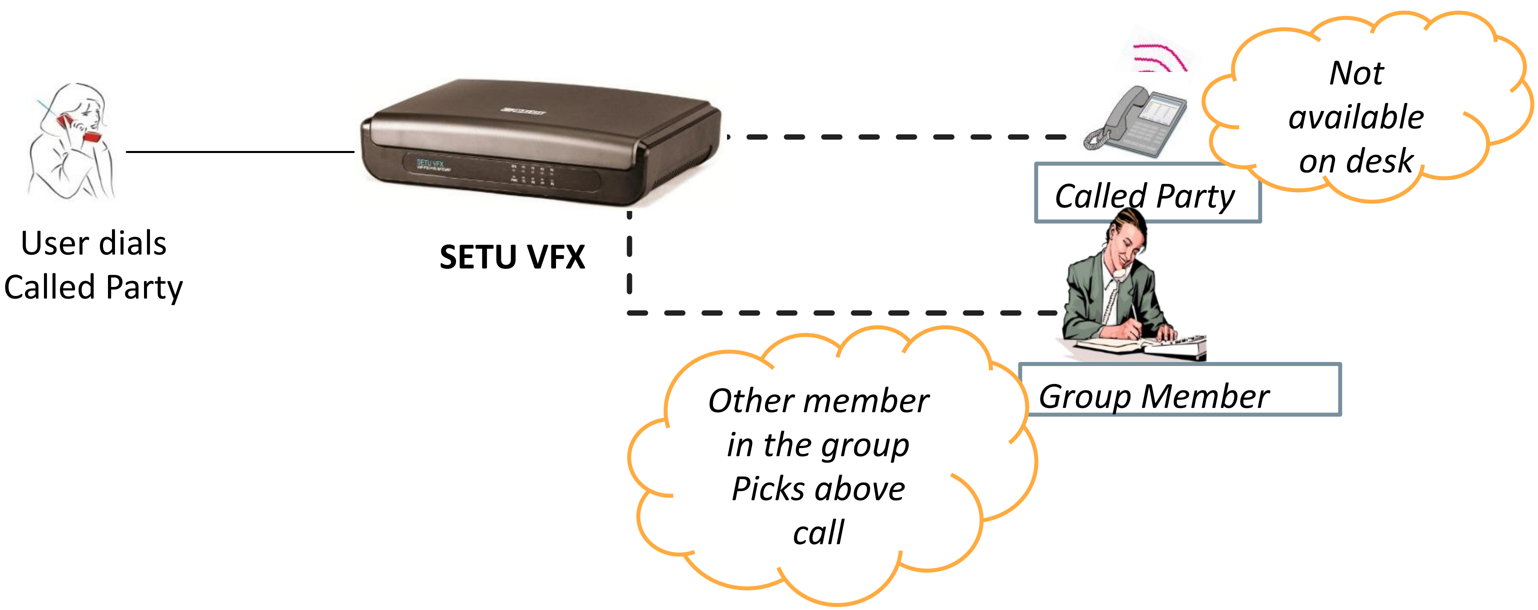
- Caller ID Types - DTMF Detection, FSK ITU-T and FSK Bellcore
- Caller line presentation for external and internal numbers
- FXS Port can be configured for CLIP protocol as per use

Emergency Number Dialing

- SETU VFX Supports a pass-through port for dialing emergency numbers
- About feature
  - ✔ Acts as a redundancy line in case of internet/power failure
  - ✔ Can be connected to PSTN/GSM FCT for making calls
  - ✔ It is meant for INCOMING calls only

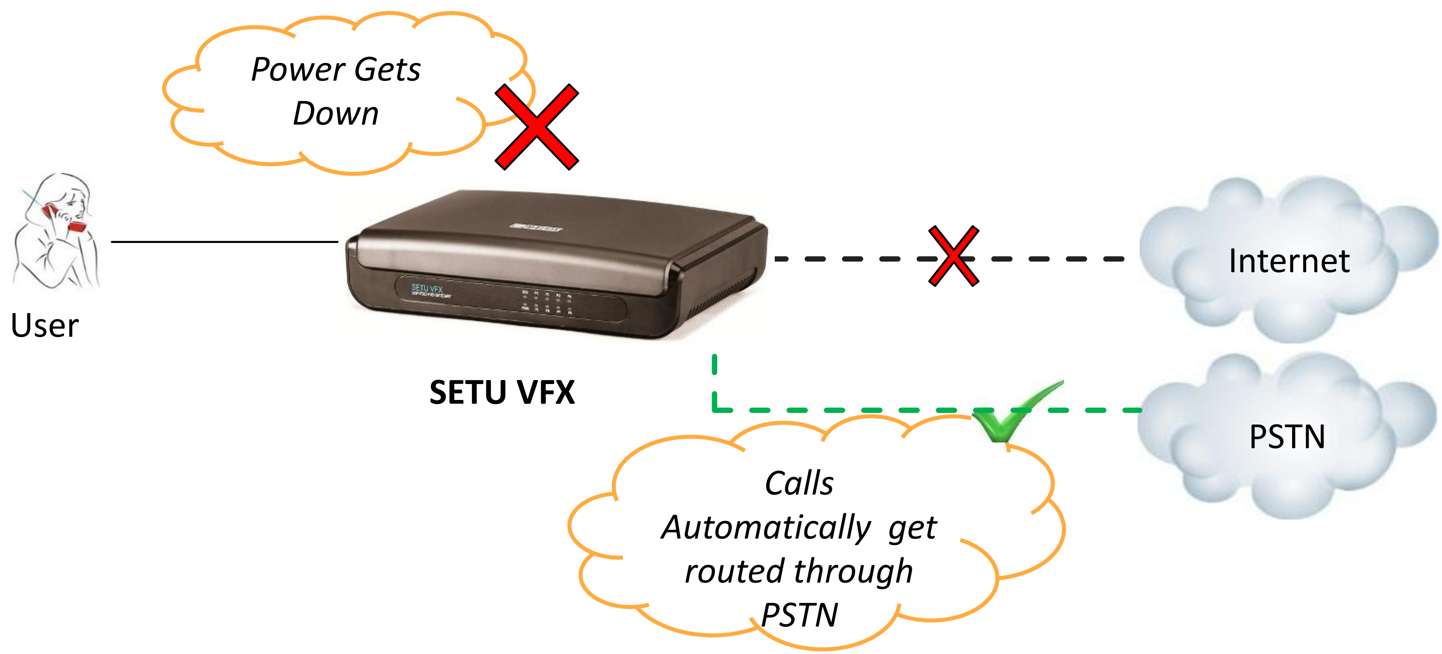
Group Call Pickup

- Flexibility to answer a call from any extension
- Save time and effort to pick a call
- Increase call response rate



Life line Support

- Maintain business continuity even in power failure
- Reliable network connectivity
- Increase productivity

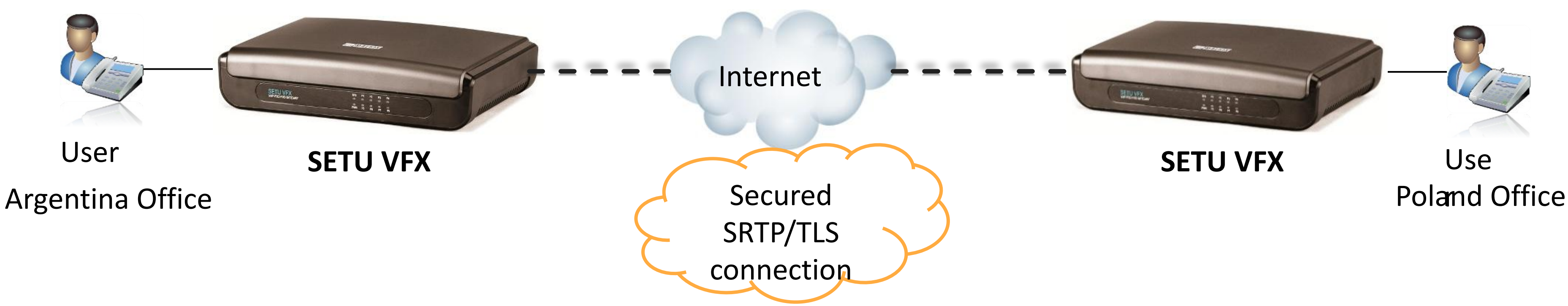


Message Wait Indication

- Quick and easy retrieval of voice messages
- Voice message indication by:
  - ✔ Message wait LED indication on phone
  - ✔ Change in dial tone (shuttered dial tone)
  - ✔ Voice message before dial tone

SIP Security over TLS/SRTP

- Communicate securely over VoIP
- Share business secrets with partners over VoIP without worrying for security
- Place important calls using VoIP



TR-069 Auto-Configuration

- TR-069 is a Secure Remote Management Protocol
- It is used for Auto-configuration and Remote management of the device at customer premises from an Auto-Configuration Server
- It supports a variety of functionalities:
  - ✔ Dynamic Service Provisioning
  - ✔ Firmware Management
  - ✔ Status and Performance Monitoring
  - ✔ Remote Diagnostics

Feature List

- Allowed and Denied Numbers
- TR-069 Auto-Configuration
- Auto PSTN Fallback
- Auto Provisioning
- Automatic Number Translation
- Call Detail Records (CDR)
- Call Duration Control
- Call Progress Tone and Rings
- Direct Dial-In (DDI) Routing
- Dynamic DNS
- Digest Authentication
- Emergency Number Dialing
- DHCP Server and Client
- Hotline
- Message Wait Indication
- NAT and STUN
- Port Forwarding and DMZ
- Programmable Call Progress Tones
- PSTN Life Line Support
- Peer-to-Peer Calling
- PIN Authentication
- Return Call to Original Caller (RCOC)
- SIP Security over TLS/SRTP
- SNMP Monitoring
- Web based Management

Matrix GATEWAY RANGE OF PRODUCTS

PRODUCT	DESCRIPTION
SETU ATA	VoIP Adaptor with GSM, FXO and FXS Ports
SETU VGFX	Multi-port SIP based VoIP to GSM/3G-FXO-FXS Gateway
SETU VGB	Multi-port SIP based VoIP to ISDN BRI Gateway
SETU VTEP	SIP based VoIP to T1/E1 PRI Gateway
SETU VFXTH	Medium-Density Multi-port SIP based VoIP to FXO-FXS Gateway
SIMADO GFX	Multi-port GSM to FXS Gateway
SIMADO GBR	Multi-port GSM to BRI Gateway

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