

The Gateway to VoIP World





CONTENTS

DINSTAR





About us



COMPANY PROFILE

DINSTAR



10⁺

Shenzhen Dinstar Co., Ltd. was founded in 2011.

200⁺

DINSTAR has focused on developing, manufacturing and marketing of next generation communication and networking products. We have more than 200 employees.

18⁺

Our specialists come from top-rank global telecommunication enterprises, who have dedicated themselves to VoIP, unified communication and wireless mobile network for more than 18 years.

1st

No. 1 market share of VoIP Gateways & SBCs in China



We're a Listed Company

DINSTAR

Stock Code: 870319

NEEQ, New Over-The-Counter Market
China Growth Enterprise Board(GEB)
Second board





MILESTONES

Core Team Established

Key engineers from Huawei established a team to develop VoIP ISP solution, aiming to be the leader of NGN industry



2002



2011

DINSTAR Established

Focus on R&D, manufacturing, sales and service of IP communication devices including GSM VoIP gateway, trunk gateway and FXS/FXO VoIP gateway



A Listed Company

Dinstar successfully listed on China's New OTC Market on 21st, December;
Became one of the first 30 members of
Huawei eLTE alliance

2016



New Office

Floor 18, Building 7A, Vanke Cloud City Phase 1, Xingke 1st Street, Xili Sub-district, Nanshan District, Shenzhen, P.R. China 518000

2020



OUR TEAM

18 Years in VoIP Industry

DINSTAR



R&D Team

Software & Hardware Development



Sales Team

Marketing & Sales



Support Team

Technical Support



Supply Chain

Factory



Awards & Certificates

AWARDS & CERTIFICATES

- ❖ National Hi-tech Enterprise
 - ❖ ISO9001
 - ❖ Invention Patents & Software Copyrights
 - ❖ Product of the Year by Internet Telephony
 - ❖ Best VoIP Gateways, Editors Choice by CTI Forum
 - ❖ Certified by Huawei, Broadsoft and Elastix
 - ❖ CE, FCC & CCC





Partners & Customers



OUR CUSTOMERS

DINSTAR

Turnover
>\$20 million



Customers
>120,000



Countries
>100



Lines
>10,000,000





OUR PARTNERS

Our partners in China

DINSTAR



China Mobile

China
unicom

HUAWEI

ZTE

Tencent

360
www.360.cn

Alibaba.com

JD

NETEASE
www.163.com



STATE GRID

SHANDONG
AIRLINES

SOOCHOW LIFE

NISSAN

vatti

Cestbon

meten

vivo

EXPRESS
顺丰速运

TCL





OUR CUSTOMERS

Projects of Telecom Operators

DINSTAR



Digicel

mts



Claro



vivo



movistar

optcl

МЕГАФОН



OUR PARTNERS

Our partners around the world





COMPATIBILITIES

DINSTAR

Dinstar products are compatible with leading platforms





Professional

18-year
Development and manufacturing
experience of VoIP industry

Broad Compatibility

Compatible with mainstream
IP PBX and IMS platform vendors

One-stop Shop

Full series VoIP hardware including
VoIP Gateways, IP Phones, IP PBXs
and SBCs

Service

Fast-respond technical services
including installation, configuration
and trouble-shooting etc.



Benefits of VoIP

DINSTAR

Cost-saving



Clear Voice Quality



Easy to Install, Manage
and Maintain



Benefits of VoIP



Scalability



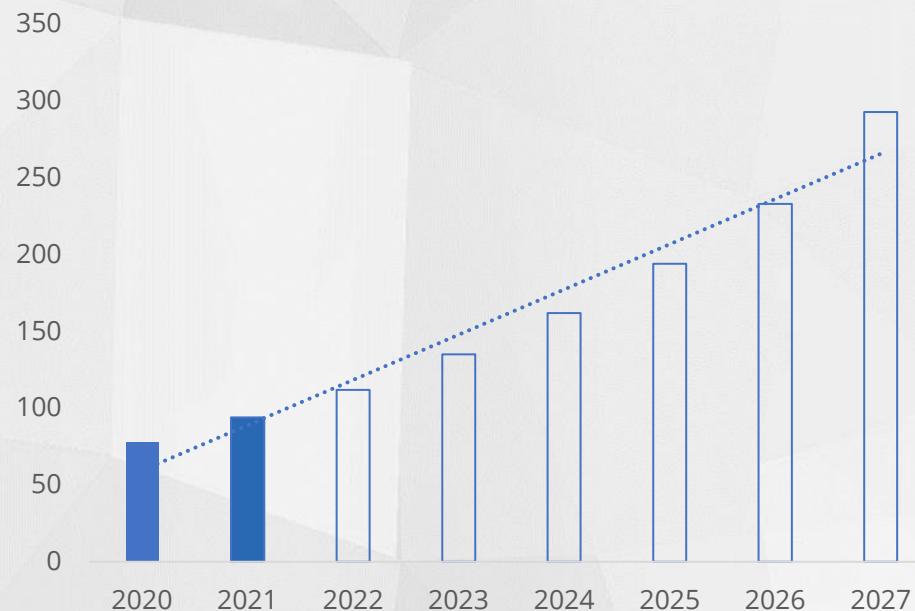
Abundance of Features



Mobility & Geographical Flexibility
& Working from home

Global Unified Communications Market

The global unified communications market size is expected to reach \$293 billion by 2027, rising at compound annual growth rate (CAGR) of 19.6% from 2021 to 2027.



2021-2027 CAGR



Applications



Telecom Operator



Service Provider



System Integrator



Contractor



SME/Enterprise



Call Center

By Verticals



BFSI



Education



Healthcare



Government



Emergency Service



Logistics & Transportation



IT & Telecom



Public Sector & Utility



Travel & Hospitality



Consumer Goods & Retail



Products



OUR PRODUCTS

IP Communication Solutions



SME IP PBX



Enterprise IP Phone



Analog VoIP Gateway



Digital VoIP Gateway



Session Border Controller



GSM VoIP Gateway



SIP Door Phone

IP PBX



Desktop-size

UC120/UC200



Enterprise IP PBX

UC350



Customizable IP PBX

UC1500/2500



Software IP PBX

UC8000

- 60/500 SIP Extensions
- 15/30 Concurrent Calls
- Recording
- Voicemail
- LTE Failover

- 500 to 1,000 SIP Extensions
- 60 to 200 Concurrent Calls
- Up to 32 FXS/FXO
- Up to 4 E1/T1
- Redundant Power Supplies

- X86 Architecture
- Open API: Perfect Integration
- Modular Design
- FXS/FXO: up to 32/80 ports
- E1/T1: up to 16/32 ports

- Up to 20,000 SIP Extensions
- Up to 4,000 Concurrent Calls
- Easy to Scale
- On-premise or Cloud
- High Availability



UC120

2 FXO, 1 LTE(optional)

Up to 60 SIP Extensions

Up to 15 concurrent calls

Wi-Fi Hotpot

4G LTE as internet connection

4G LTE network failover as business continuity

IP PBX features

SME / Branch Office / Temporary Office / Remote Area



60
SIP Extensions



15
Concurrent Calls



Data Voice



IVR



Wi-Fi



Recording



Voicemail





FXS/FXO



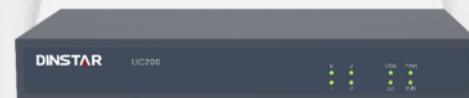
500 SIP Extensions
60 Concurrent Calls



Recording

UC200

2 FXS, 2 FXO, 1 WAN, 1 LAN



SIP Video



Voicemail

T.38



VPN



Remote
Management

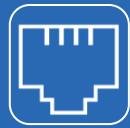
UC350 IP PBX



LCD



Gigabit
Ethernet Port



Up to 4
E1/T1



Up to 32
FXS/FXO



Redundant
Power Supplies



Voice
Recording



VPN



1000 SIP Extensions
200 Concurrent Calls



Customizable IP PBX UC1500/UC2500

DINSTAR

Open Hardware Platform

X86 Architecture
CPU Selectable
Open API: Perfect Integration
Install Your Own PBX Software

High Availability

Redundant Power Supplies
Redundant MCU
Flexible Routing
PSTN Fallback

Modular Design

FXS/FXO: up to 80
E1/T1: up to 32
GSM/LTE: up to 32
Hot Swappable

Verticals Applications



Mining



Utility



Public Security



Healthcare



Education



Government



Banking & Financial Service

UC1500/UC2500



New Software IP PBX UC8000

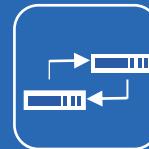
Designed for Your Own Hardware or Virtual Server



20,000 SIP Sessions



4,000
Concurrent Calls



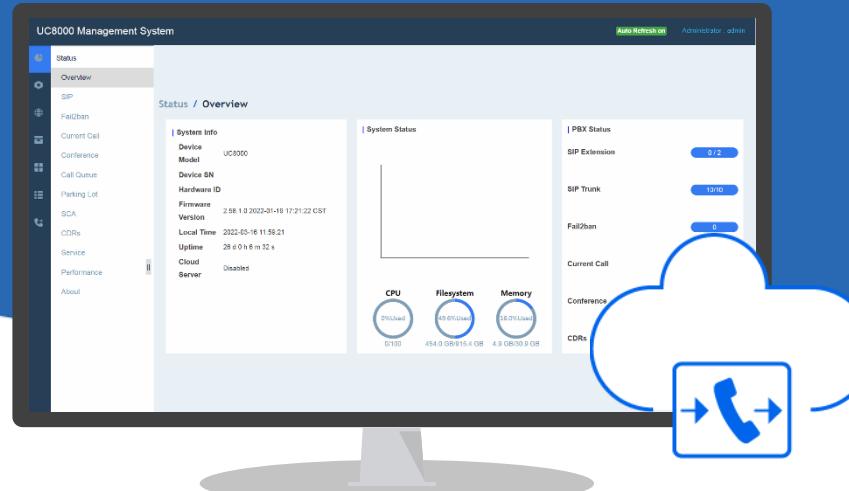
High Availability
Active/Standby



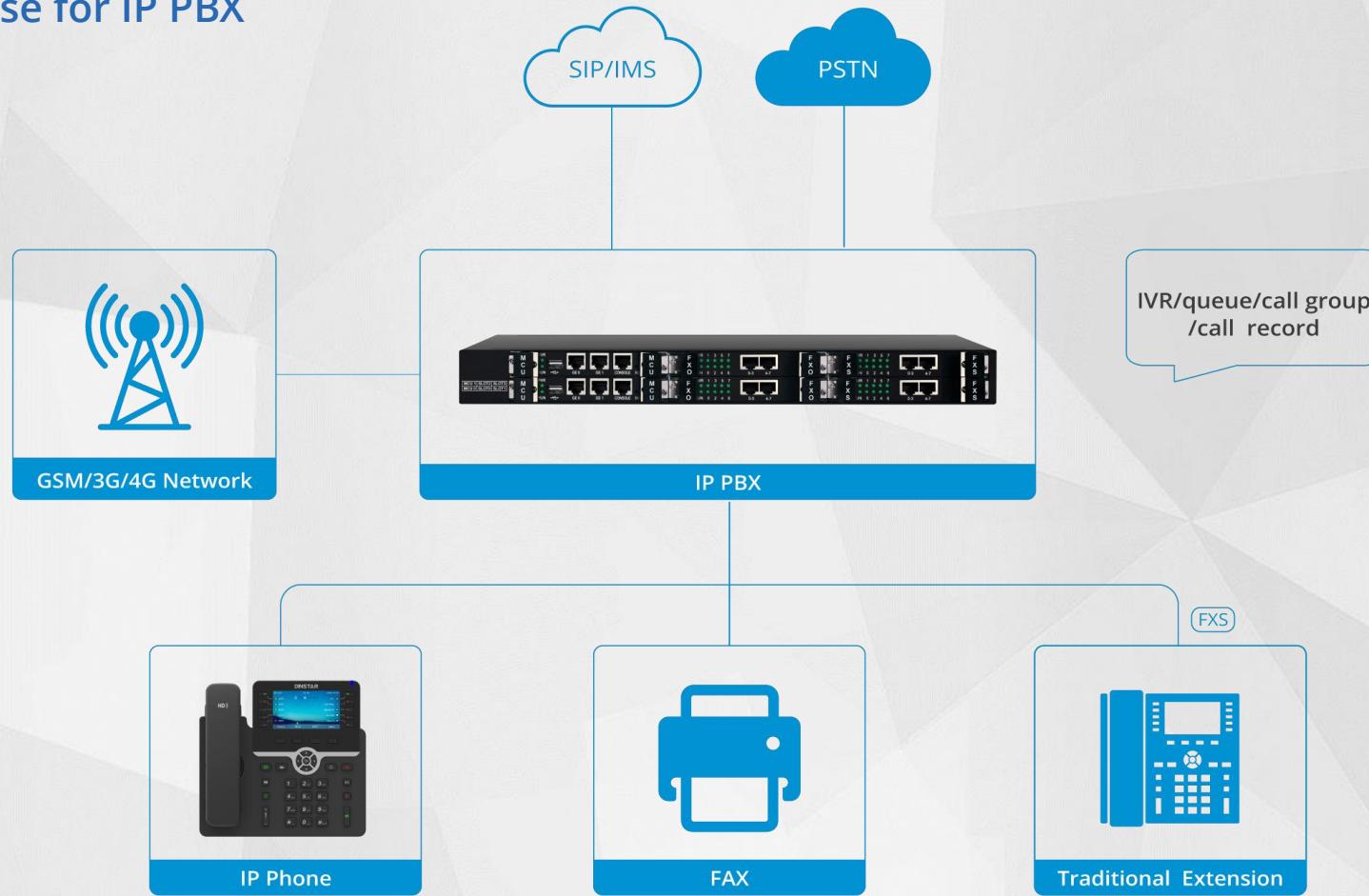
Scalable



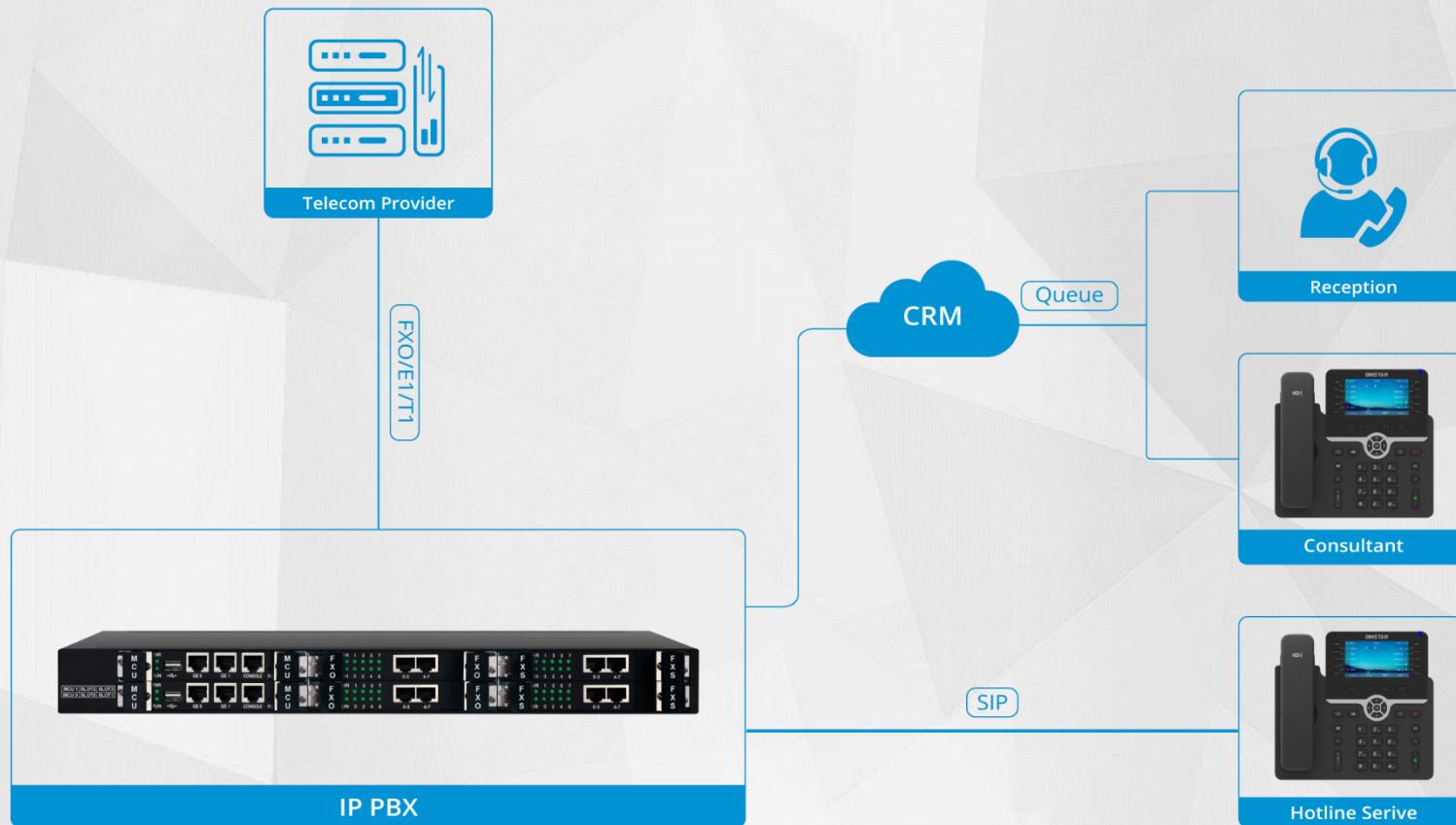
Easy
Deployment



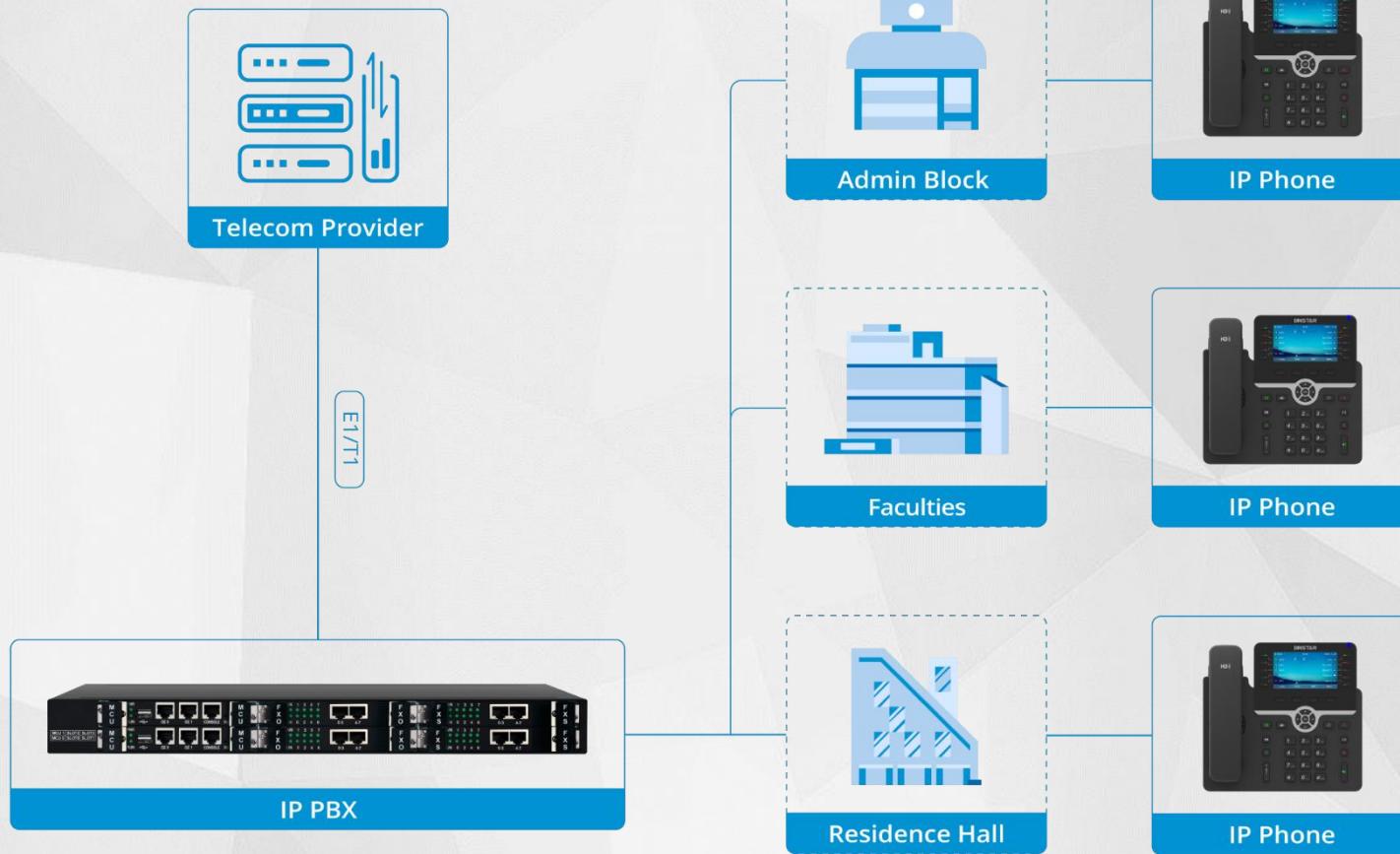
Use case for IP PBX



Support System for Banking/Insurance Companies



Education

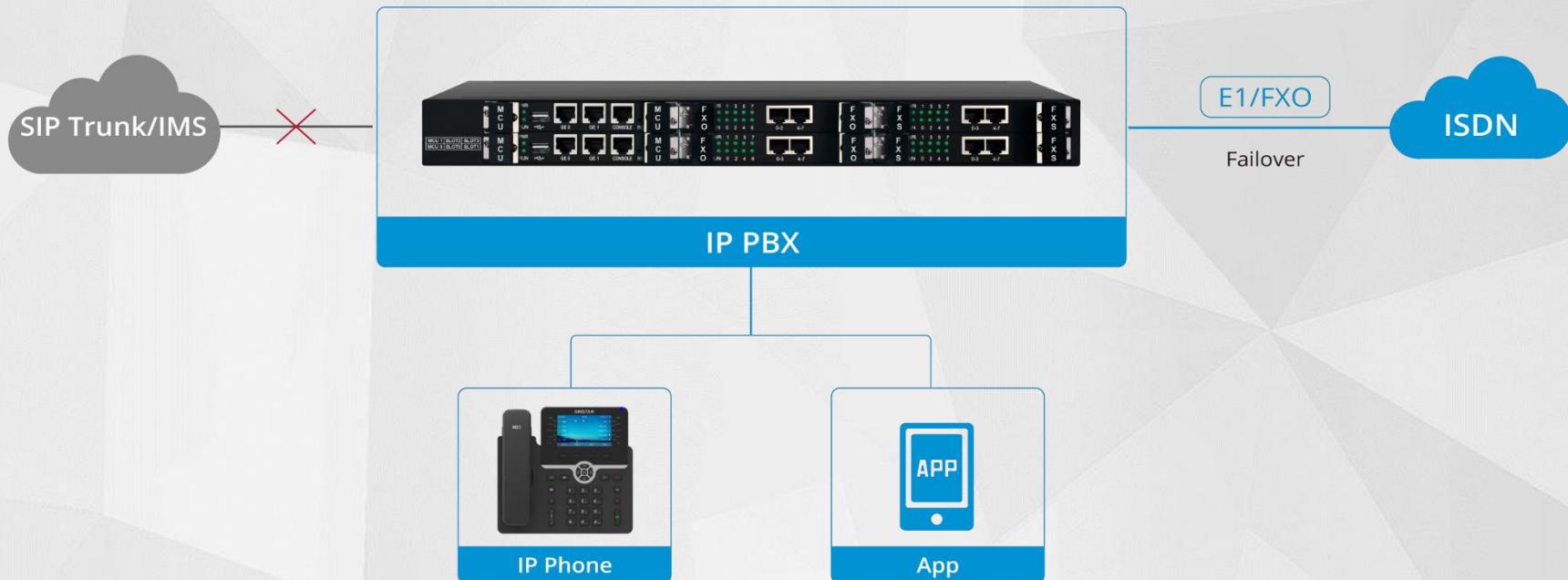




Failover Trunk



Failover Trunk



IP Phone
for Enterprise



High-end IP Phone



C66G/GP

20 SIP Lines

50 Line Keys

6-way Conference

USB2.0*1

EHS



C64G/GP

16 SIP Lines

40 Line Keys

6-way Conference

USB2.0*1

EHS

Business IP Phone



C63G/GP

6 SIP Lines

30 Line Keys

5-way Conference

EHS



C62G/GP

6 SIP Lines

2 Line Keys

5-way Conference

EHS

Entry Level IP Phone



C61S/SP

2 SIP Lines

2 Line Keys

5-way Conference

EHS



C60S/SP

2 SIP Lines

2 Line Keys

5-way Conference

EHS

High-end Business SIP Phone

C66G/C66GP



HD Voice



4.3" Graphical LCD



PoE



20 SIP Lines



6-way Conference



50 Line Keys



Wi-Fi & Bluetooth
Dongle



High-end Business SIP Phone

C64G/C64GP



HD Voice



3.5" Graphical LCD



PoE



16 SIP Lines



6-way Conference



40 Line Keys



SIP Phone for SMEs

C63G/C63GP



HD Voice



2.8" Graphical LCD



PoE



6 SIP Lines



5-way Conference



30 Line Keys





Entry Level IP Phone

C62G/C62GP



HD Voice



2.4" Graphical LCD



PoE



6 SIP Lines



5-way Conference



2 Line Keys





Entry Level IP Phone

C61S/C61SP



HD Voice



2.3" Graphical LCD



PoE



2 SIP Lines



5-way Conference



2 Line Keys





Entry Level IP Phone



HD Voice



2.3" Graphical LCD



PoE



2 SIP Lines



5-way Conference



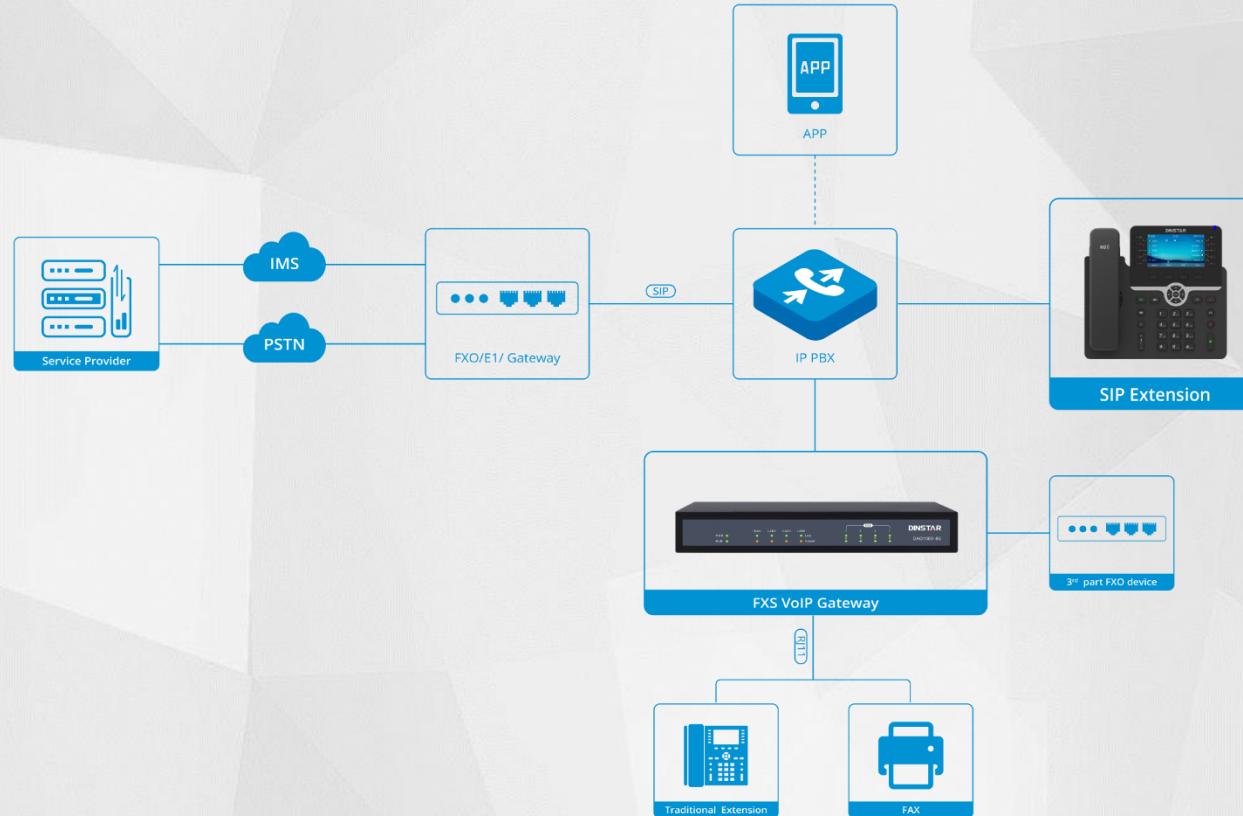
2 Line Keys



Model	C60U/C60UP	C60S/C60SP	C60L/C60LP
Ethernet Port	2	2	1
Backlight	✓	X	X

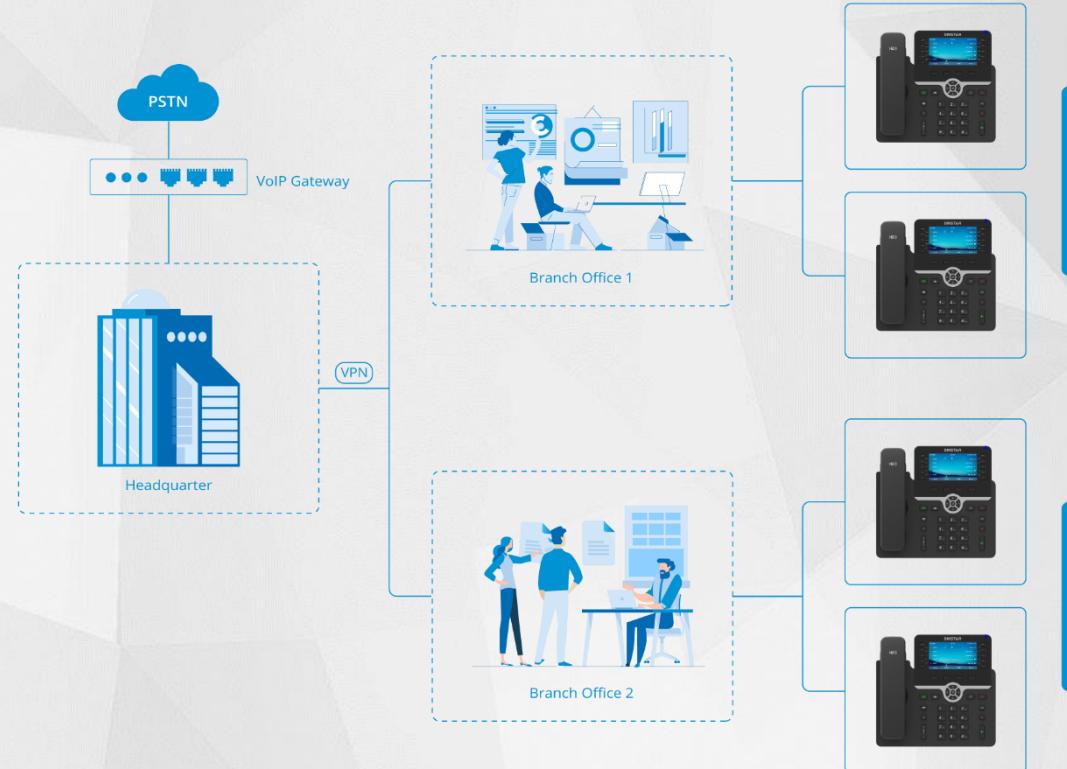
Use Case for IP Phone

SIP phones can be used for any SIP based IPPBX , paging system and UC solutions. In general purpose, SIP phones must work with IPPBX through registering SIP accounts. They can work with DINSTAR IPPBX or other's brands with IP/SIP.



Headquarter and Branch Office Connection by VPN

Headquarter IPPBX can be connected by VPN or SIP trunk. The diagram below uses VPN connecting with different locations and branch offices. SIP phones can be easily setup by using VPN feature in SIP phones or by VPN provided by headquarter. There is no extra cost charged.



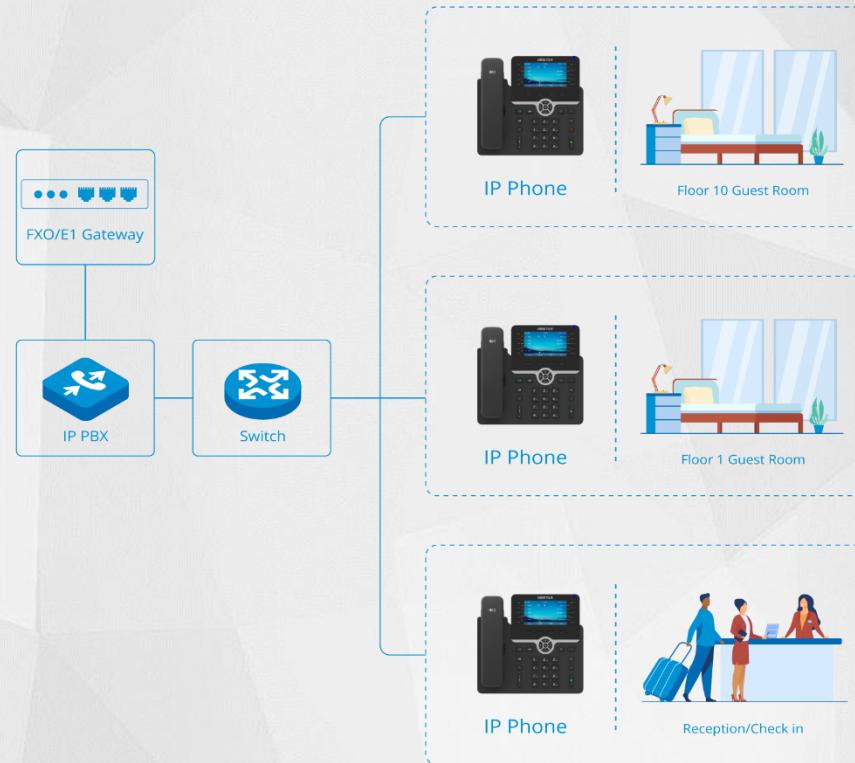
Cloud IPPBX/UC Service and Hosting IPPBX Using IP Phones

DINSTAR IP Phone supports kinds of cloud based IPPBX/UC or hosting service solution.



Hotel System with Guest Rooms

With the Hotel IPPBX, each guest room can setup a sip extension for the room with room number. Check-in also can be used to call each room. guests can make any call through IPPBX. In the hotel IPPBX, the sip can be used for internal calls, wakeup call service and paging if it is necessary. The incoming calls can be handled by receptions or IPPBX IVR with hotel PMS system.



IP Based Intercom/Paging System

The Intercom/paging system can use SIP phone as a terminal to play voice broadcasting.



Analog
VoIP Gateway



Analog VoIP Gateway

Elastix and Broadsoft Certified

DINSTAR



DAG1000/DAG2000

1/2/4/8/16/24/32 Ports FXS
2/4/8/16 Ports FXO
4/8 Ports FXS/FXO Hybrid



DAG2500

48/64/72 Ports FXS
50 PIN Telco connectors



DAG3000

32 FXS/FXO
128 Ports FXS
128 Ports FXO



SIP



IMS



IPv6



Auto
provisioning



TR069



3-way
Conference

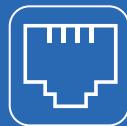


T.38 Fax



TLS / SRTP

GE&SFP FXS VoIP Gateway



4 FXS



Gigabit Ethernet



HD Voice



IPv6

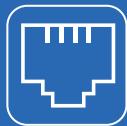


SIP TLS / SRTP

DAG1000-4S(GE)



High-end Analog VoIP Gateway



32 FXS /
FXO



Modular
Design



Gigabit
Ethernet



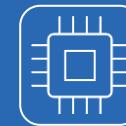
HD
Voice



IPv6



SIP TLS / SRTP



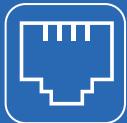
Powerful
CPU

DAG3000-32S/O



High-density Analog VoIP Gateway

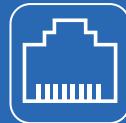
DAG3000-128S/O



128
Ports



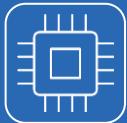
Modular
Design



Gigabit
Ethernet



Redundant Power
Supplies



Powerful
CPU



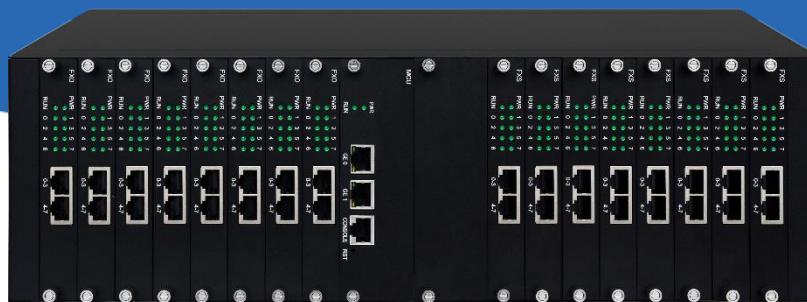
HD
Voice



IPv6



SIP TLS / SRTP



Analog VoIP Gateway

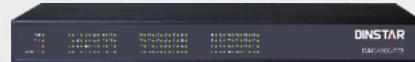
2022 Q3



DAG1000-2S2O/DAG1000-20

2 Ports FXO

2 FXS, 2 FXO Hybrid



DAG2500-96S

96 Ports FXS

6 User Boards

1U Size



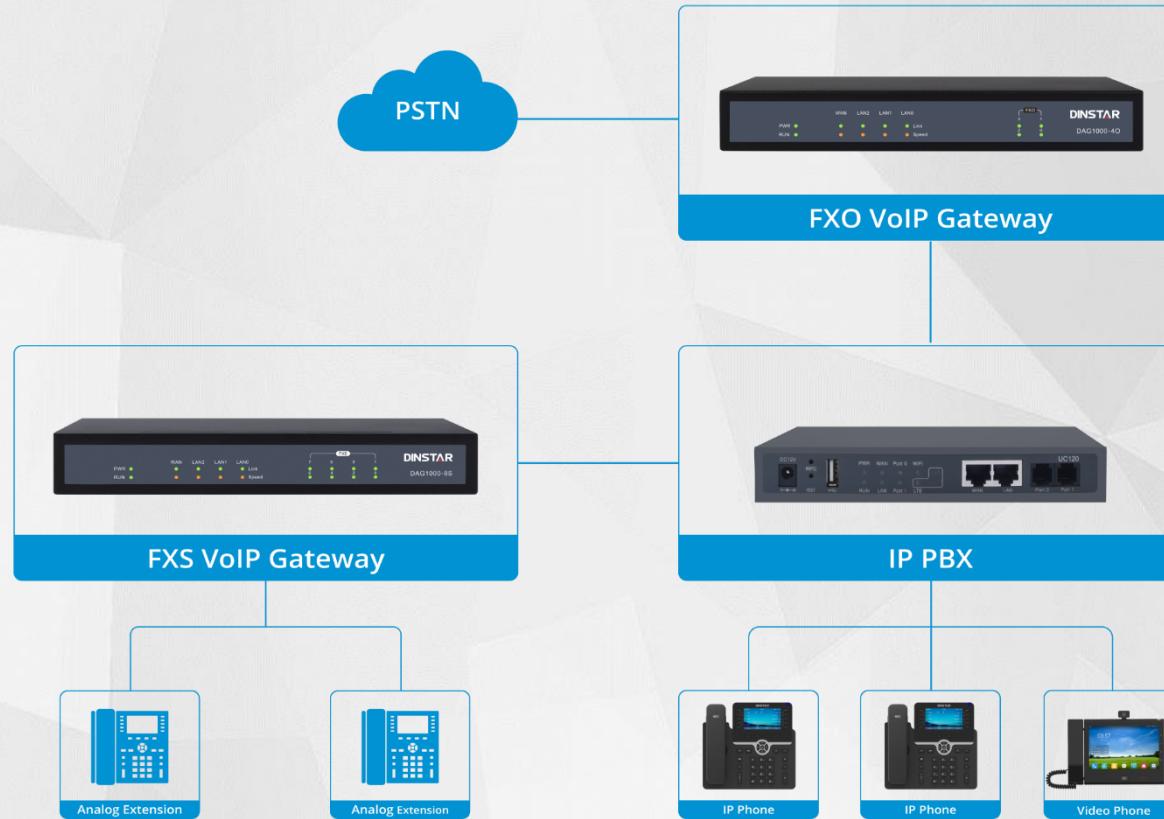
DAG3000-320S

320 Ports FXS

10 User Boards

4U Size

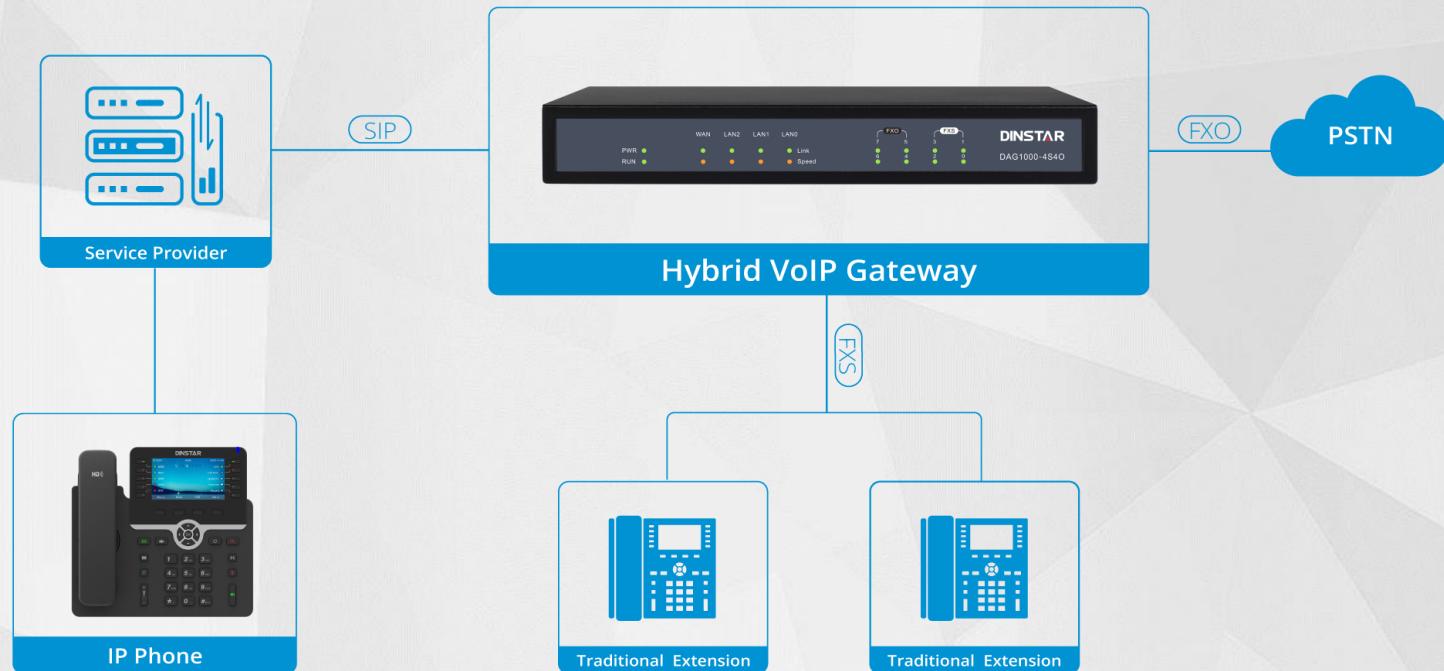
Use Case for Analog VoIP Gateway



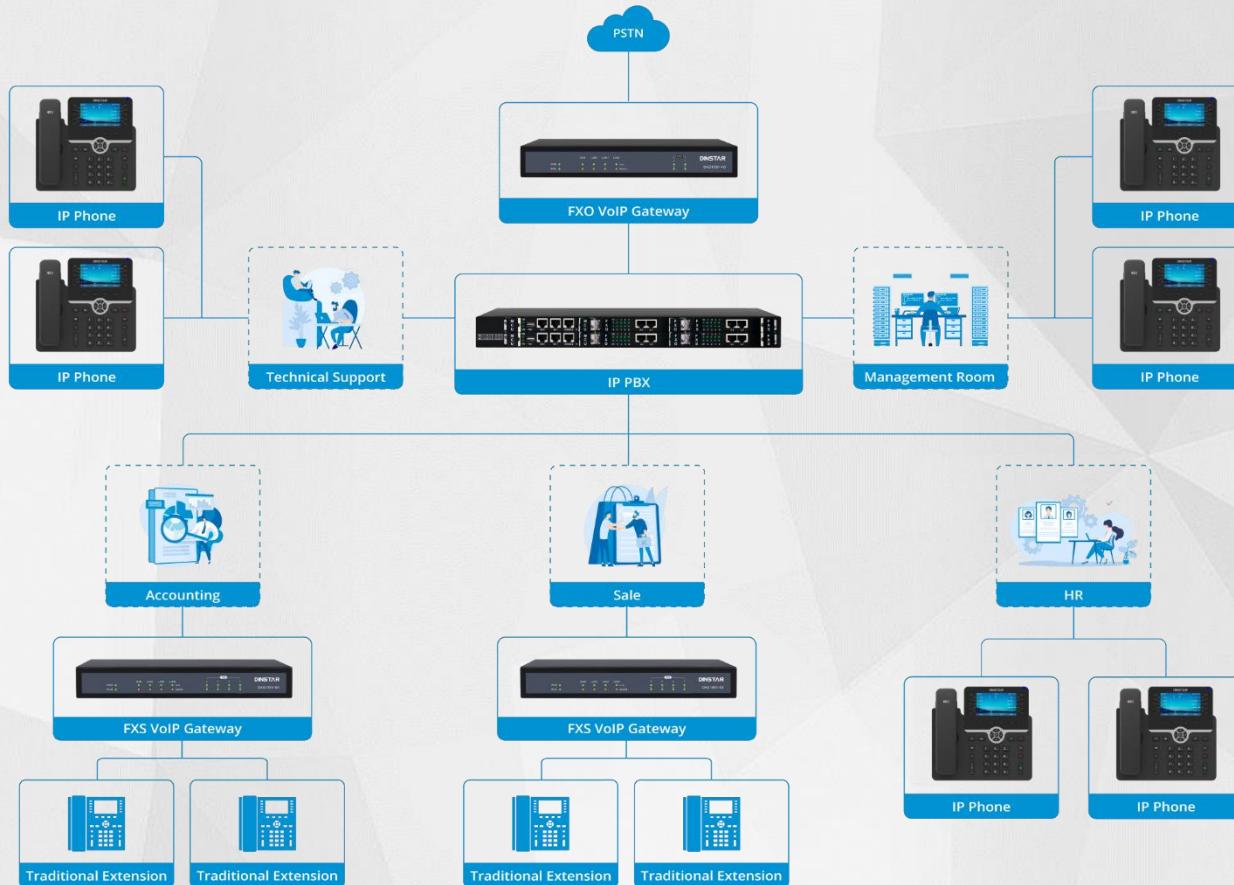
Use Case for FXO VoIP Gateway



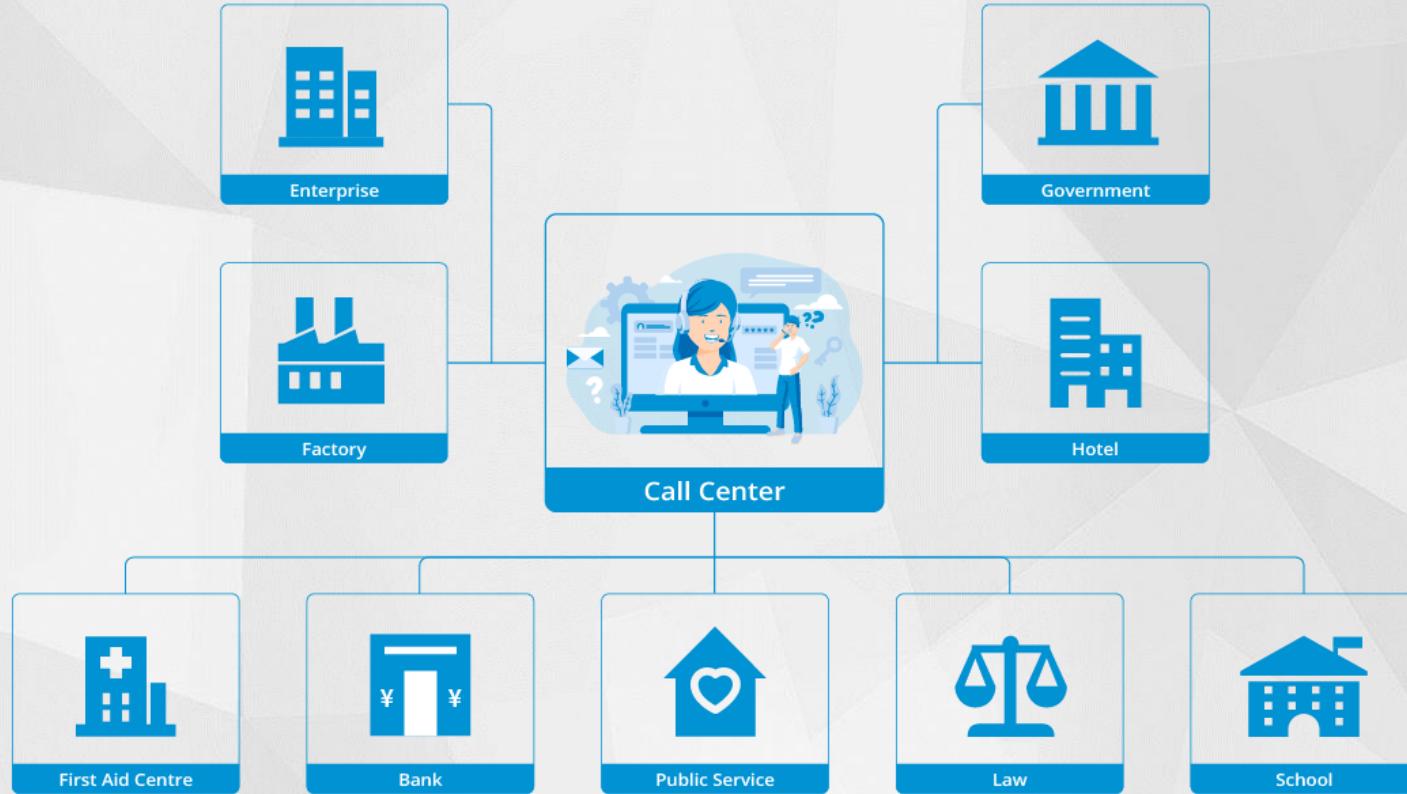
Use Case for Hybrid VoIP Gateway



Typical Scenario in the Enterprise Workplace



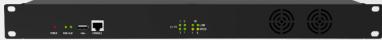
Call Center in Many Sectors



Digital
VoIP Gateway

Digital VoIP Gateway





MTGG200/MTG1000
1/2 * E1/T1
Up to 60 Concurrent Calls



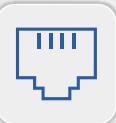
MTG2000/MTG2000B
4/8/12/16/20 * E1/T1
Up to 600 Concurrent Calls



MTG3000
16/32/48/63 E1 Ports
Up to 1890 Concurrent Calls



MTG5000/MTG5000B
64 * E1/T1
Up to 1920 Concurrent Calls



E1/T1



SIP/SIP-T/PRI
SS7/R2 MFC



Echo Cancellation
up to 128ms



Web Interface,
SNMP, Cloud NMS



G.711, G.729, G.723,
iLBC, AMR



T.38 Fax

Enterprise Media Gateway

MTG200

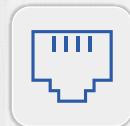


Desktop-sized

MTG1000



1U Rack-mountable



1/2 * E1/T1



SIP/SIP-T/PRI
SS7/R2 MFC



Echo
Cancellation
up to 128ms



Web Interface,
SNMP,
Cloud NMS



Up to 60
Concurrent
Calls



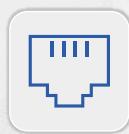
G.711,
G.729, G.723,
iLBC, AMR



T.38 Fax

High-end MTG2000/MTG2000B

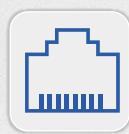
Rack-Mountable, HA



4/8/12/16/20
E1/T1



Modular
Design



Dual Gigabit
Ethernet Ports



ISDN PRI, SS7,
SIP UDP/TCP



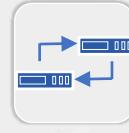
Up to 600
Concurrent Calls



Redundant
Power Supplies



T.38 Fax



1+1 Mainboard
Redundancy
(MTG2000B)



MTG3000 STM-1 Media Gateway

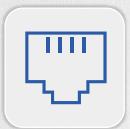
SDH 155M, 16/32/48/63* E1



Modular
Design



1890
Concurrent Calls



16/32/48/63
E1 Ports



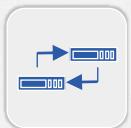
Redundancy
Uplink GE



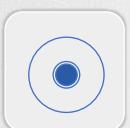
ISDN PRI, SS7,
SIP UDP/TCP



Redundant
Power Supplies



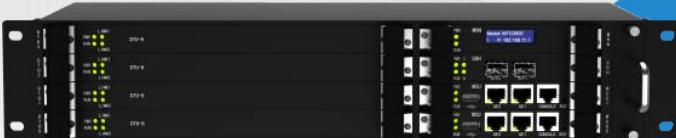
1+1 Mainboard
Redundancy



STM-1



T.38 Fax



MTG5000/MTG5000B Electrical Port Media Gateway

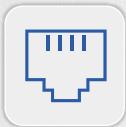
RJ48C, 64* E1



Modular
Design



1920
Concurrent Calls



64
E1 Ports



Redundancy
Uplink GE



ISDN PRI, SS7,
SIP UDP/TCP



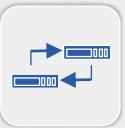
Redundant
Power Supplies



T.38 Fax



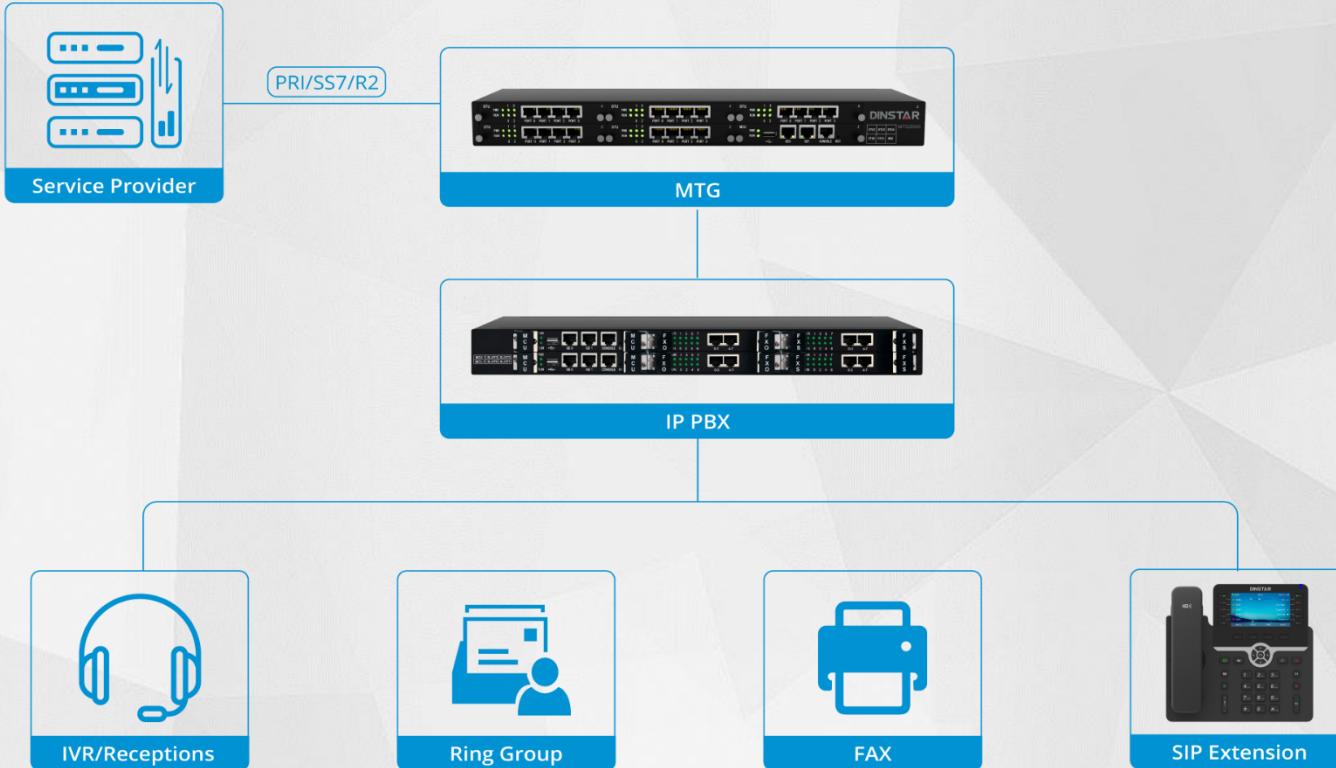
RJ48C



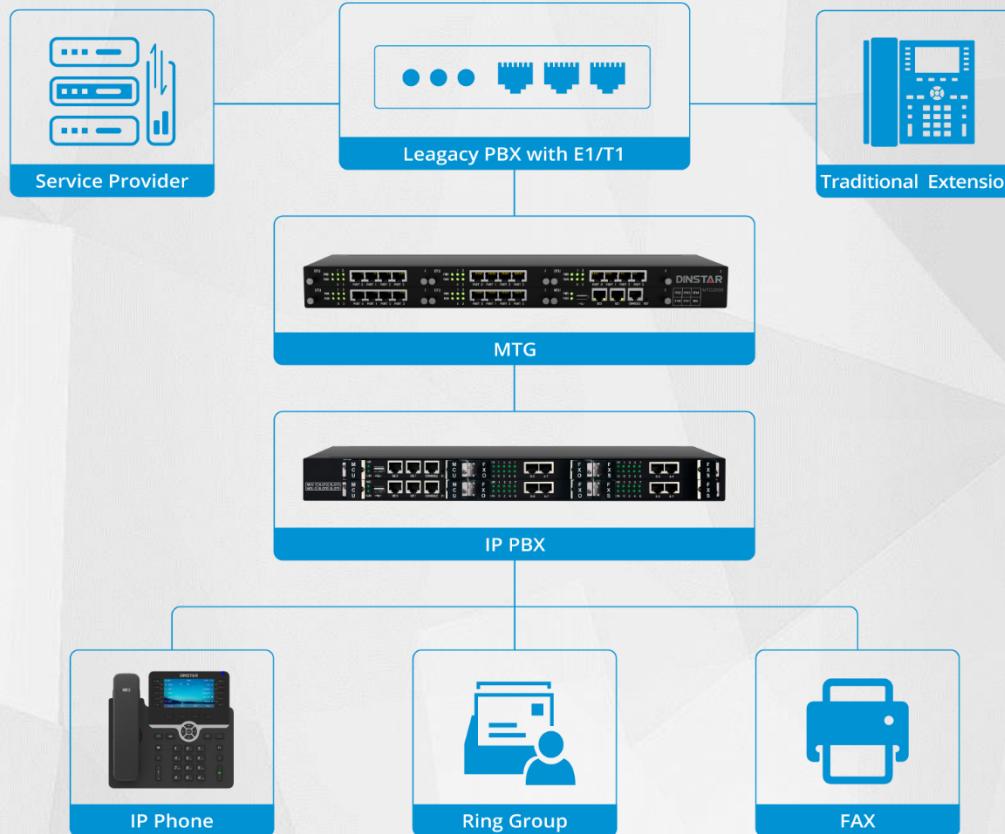
1+1 Mainboard
Redundancy
(MTG5000B)



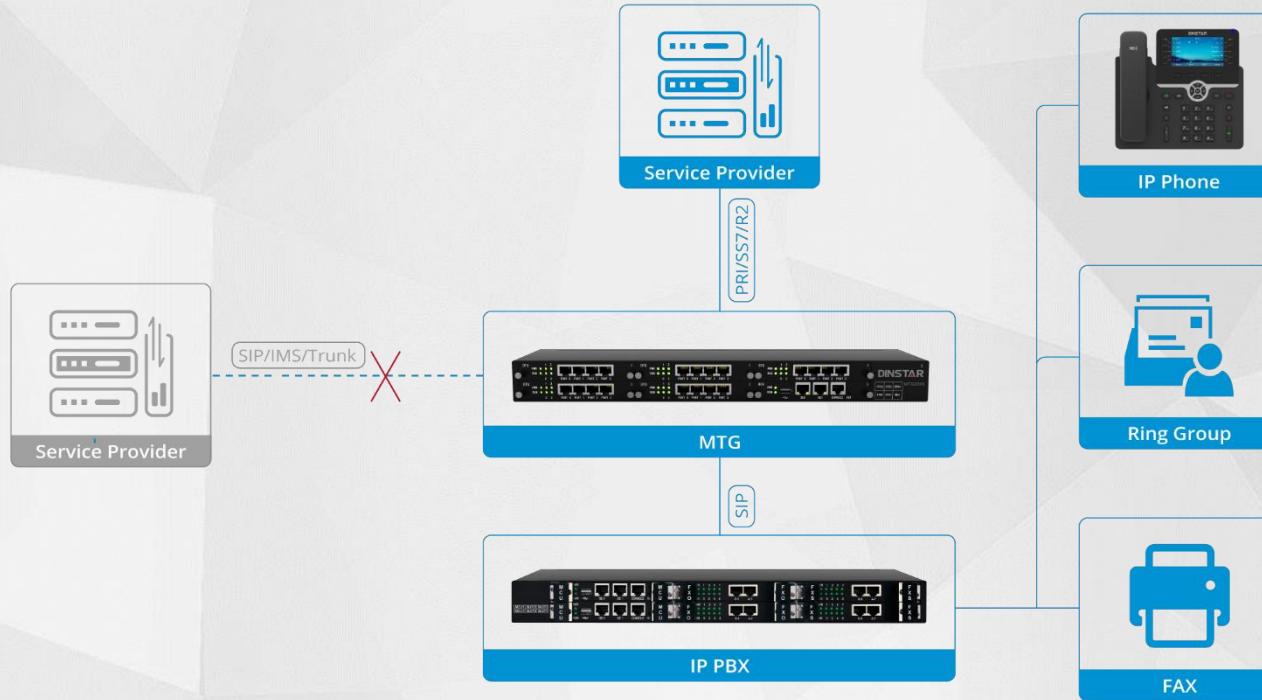
Typical Scenarios for Digital VoIP Gateways



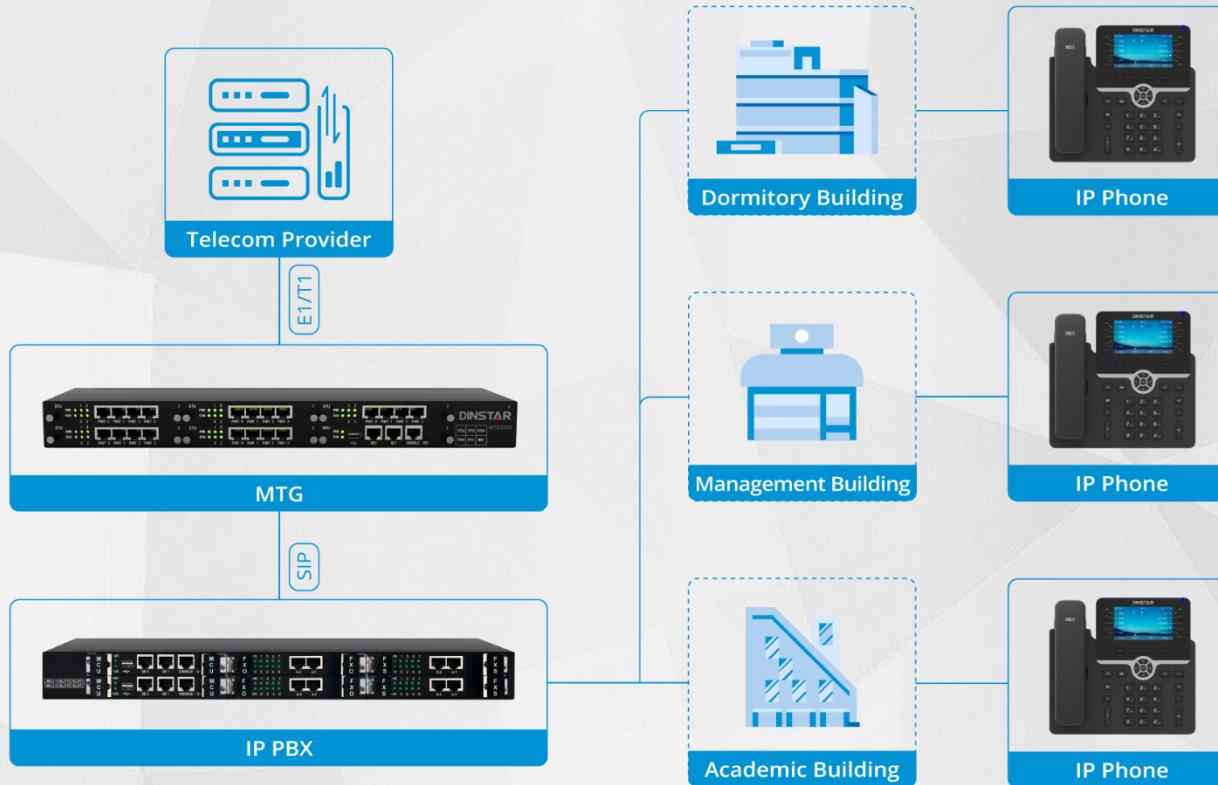
Connecting with Legacy E1/T1 Device or PBX/ Phone System



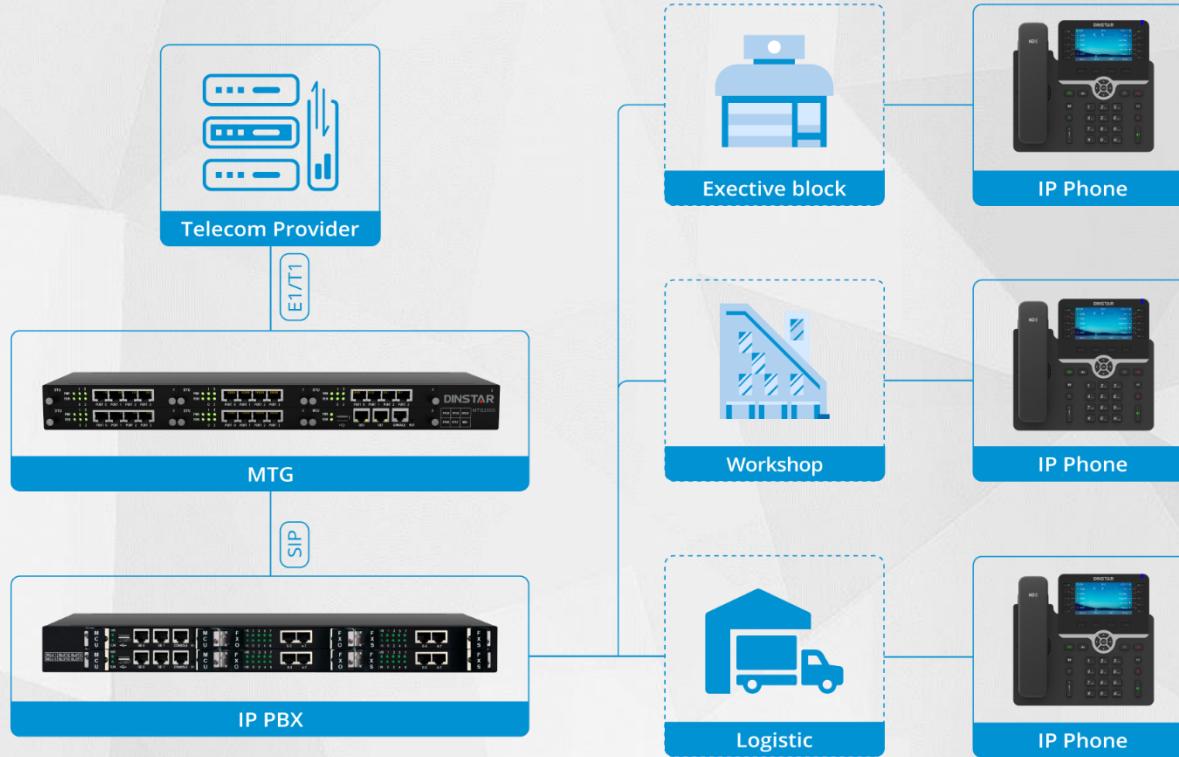
SIP/IMS Fail-over with E1/T1



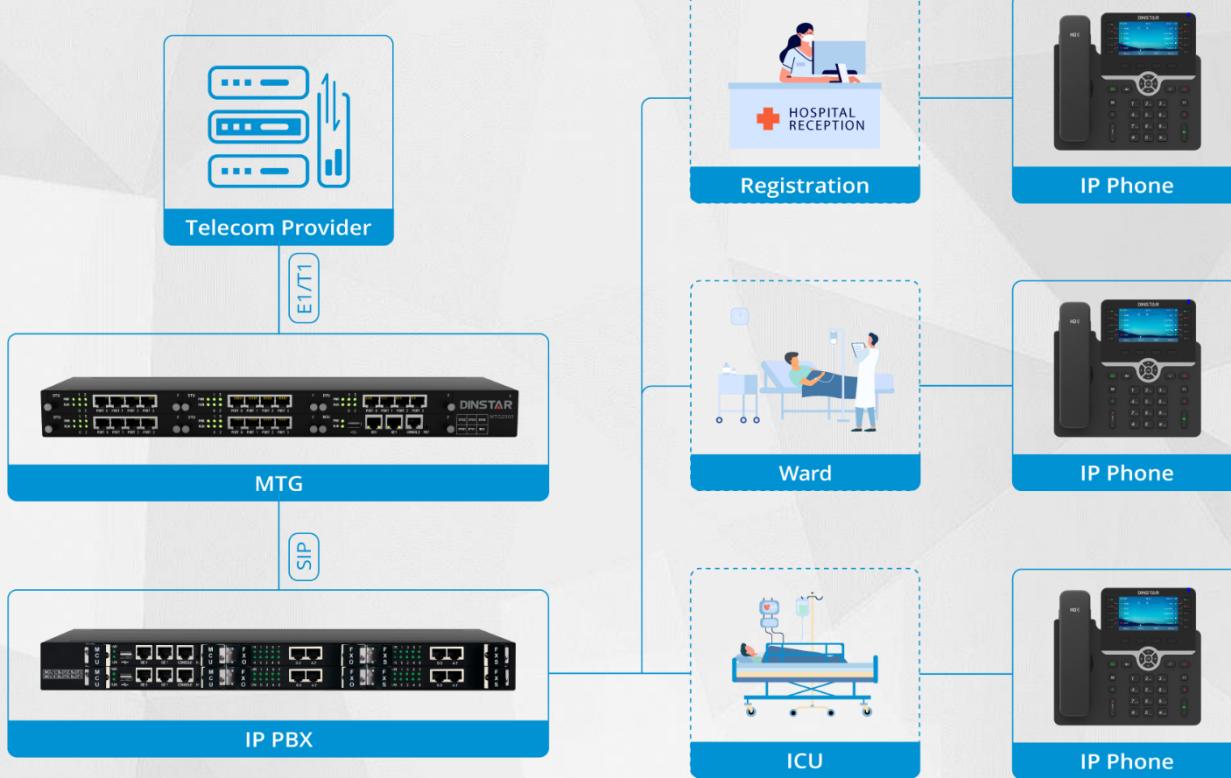
School/University Campus Phone System



Connect with a Phone System for Manufacturers



Work with Phone Systems in Hospitals



Session
Border
Controller



Session Border Controller

DINSTAR



Security

- SIP Firewall
- SIP TLS/SRTP
- Anti DDoS Attack
- Access Control



Resiliency

- High Availability
- SIP Trunk Load Balancing
- Alternative Routing



WebRTC

- WebRTC Gateway
- Reverse HTTP Proxy

- Topology Hiding
- Endpoint Authentication
- SIP Intrusion Prevention
- SIP Malformed Packet Protection



Session Control

- Traffic Control
- Registration / Call Rate Limiting
- Flexible Routing
- QoS



Interoperability

- SIP Header Manipulation
- SIP Message Manipulation
- B2BUA
- IPv4 – IPv6
- Transcoding





Session Border Controller

DINSTAR



SBC300

SMB SBC

- 5 to 50 SIP sessions
- Up to 50 transcoding calls
- 25 calls per second at maximum
- Maximum SIP registrations: 1000
- 20 Registration per second



SBC1000

Enterprise SBC

- 50 to 500 SIP sessions
- Up to 200 transcoding calls
- 25 calls per second at maximum
- Maximum SIP registrations: 5000
- 25 Registration per second



SBC3000

Carrier SBC

- 500 to 2000 SIP sessions
- Up to 1200 transcoding calls
- 200 calls per second at maximum
- Maximum SIP registrations: 10000
- 200 Registration per second



2022 Unified Communications Product of the Year Award

DINSTAR

INTERNET TELEPHONY MAGAZINE

TMC, a global, integrated media company helping clients build communities in print, in person and online, has named our Session border Controller as a 2022 Unified Communications Product of the Year Award winner.

“It gives me great pleasure to honor DINSTAR as a 2022 recipient of TMC’s Unified Communications Product of the Year Award for their innovative solution, Session border Controller,” said Rich Tehrani, CEO, TMC. *“Our judges were very impressed with the ingenuity and excellence displayed by DINSTAR in their groundbreaking work on Session border Controller.”*



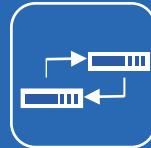
New Software SBC8000



10,000 SIP Sessions



5,000
Transcoding Sessions



High Availability



WebRTC



IPv6 / IPv4

The screenshot displays the DINSTAR SBC software interface. The main window is divided into several sections:

- System Status:** Shows various network and system metrics.
- Calls Statistics:** Displays current calls (0), peak calls (500), and registered users (2).
- MCU Status:** Shows CPU and Memory usage.
- Device Info:** A grid of icons representing different ports and connections.
- General:** Device model (SBC8000 X-8F), device name (SBC8000 X-8F), software version (2.0.1.0), and other device details.
- Graphs:** Call statistics graphs showing active and forwarded calls over time.

SBC3000 V2.0



4000 SIP Sessions
1500 Transcoding Sessions



8 Gigabit
Ethernet Ports



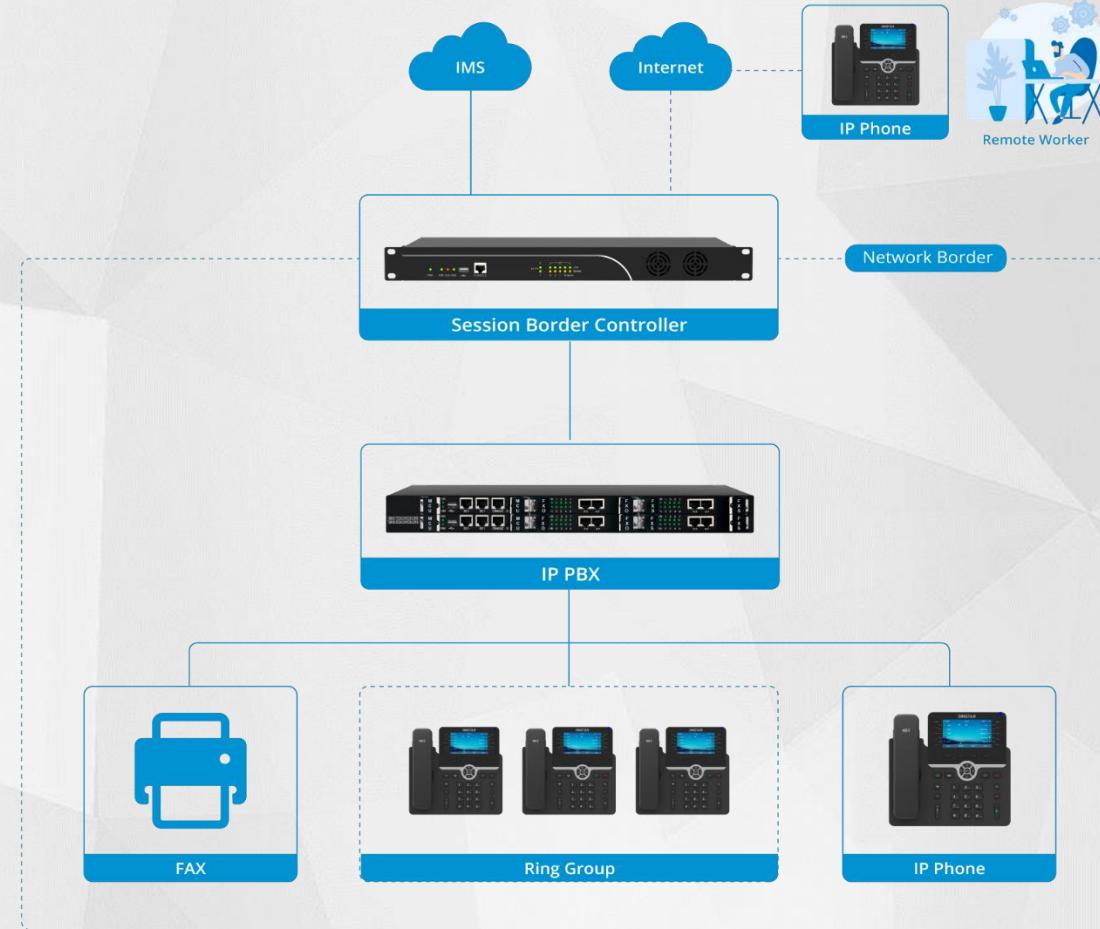
SIP TLS / SRTP



WebRTC



Use Case for SBC



Peer and Access Connection by Using SBC

Peer Connection



Wholesale VoIP Provider Data Center

Retail VoIP Service Provider Data Center

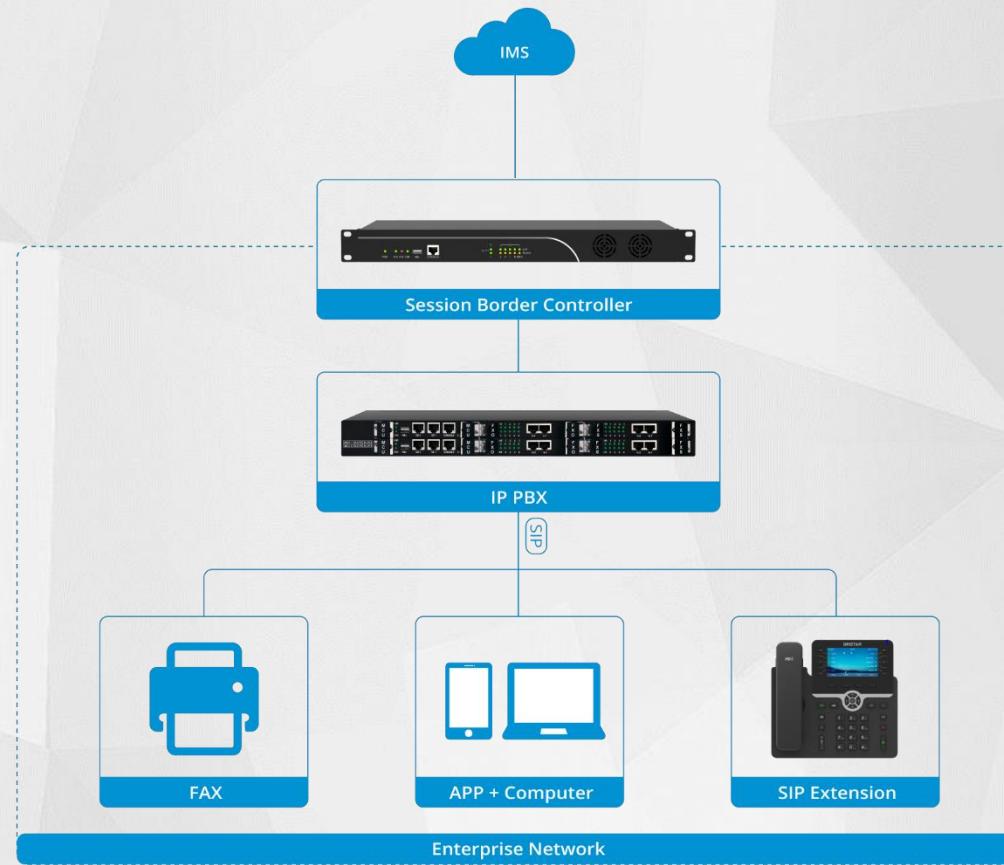
Access Connection



SMB Enterprise

Retail VoIP Service Provider Data Center

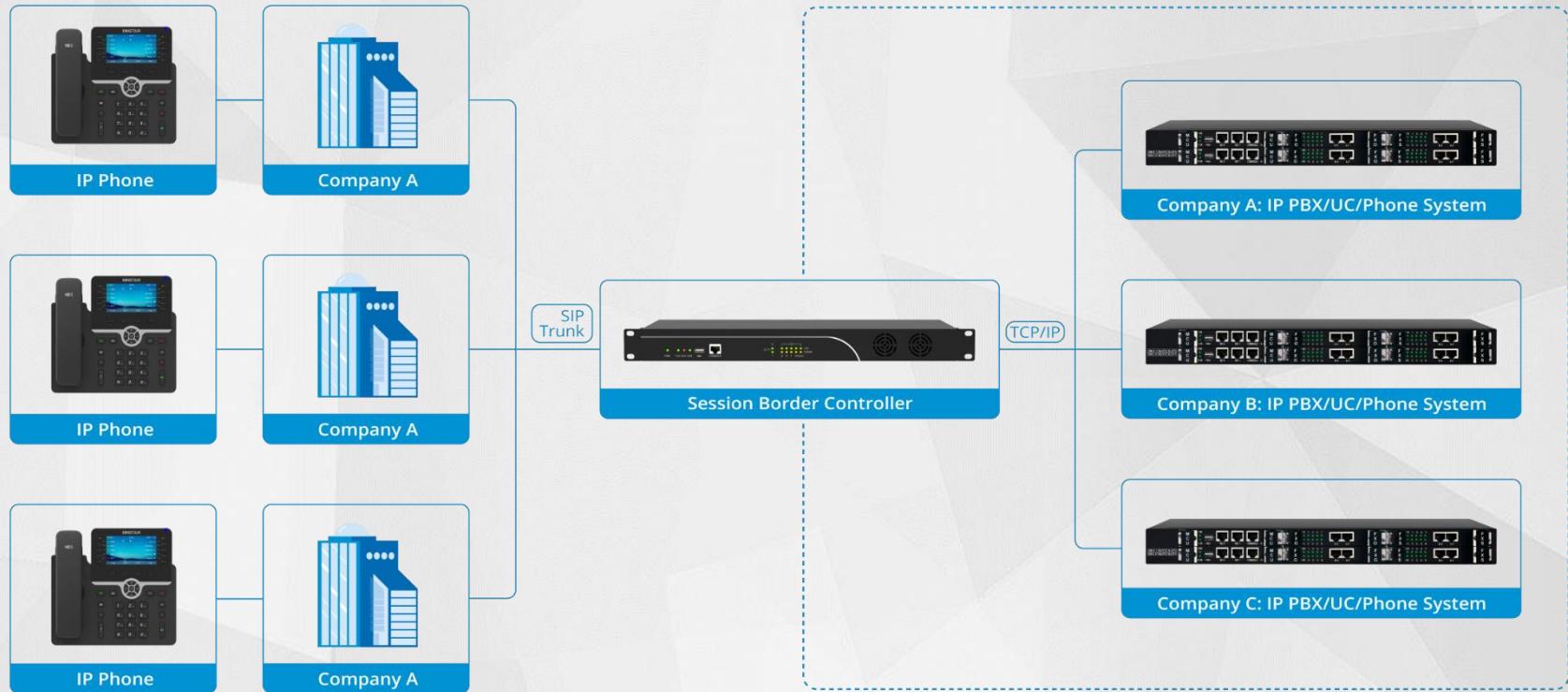
Enterprise IPPBX/UC Connects SIP Trunk/IMS through SBC for the Security Purpose



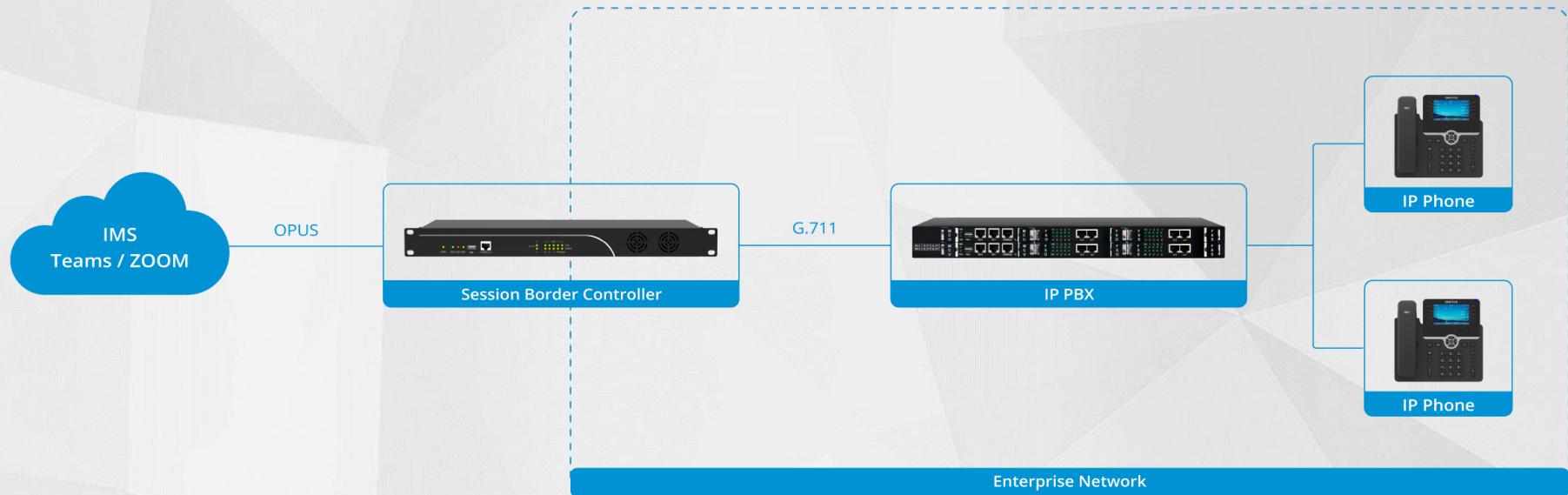
Remote Offices and Workers Connect Enterprise IPPBX through SBC



Hosting IPPBX/UC Service through SBC



SIP Trunk with Codecs Transcoding



GSM
VoIP Gateway

GSM/4G VoIP Gateway



UC2000-VE/UC2000-VF

- 4/8/16 Channels GSM/LTE
- 10/100Mbps



UC2000-VG/UC2000-VH

- 32/64 Ports GSM/LTE
- Gigabit Ethernet
- LCD Display



Multi-SIM VoIP Gateway

- 8/16 /32 GSM/LTE Channels
- 4 SIM Slots Per Channel, SIM Rotation



VoLTE



GSM



HD audio



SMS



Remote SIM Management



OpenVPN



SMS API

VoLTE



UC2000-VE
4/8 Channels GSM/LTE

Open API for SMS/USSD



SMS Receiving/Sending



GSM/LTE



Remote SIM Management
(SIMBank & SIMCloud)



Next Generation GSM VoIP Gateway

DINSTAR

UC2000-VF-16G/T

16 Ports GSM/LTE



VoLTE



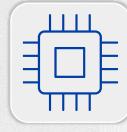
GSM



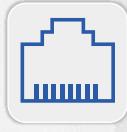
HD audio



SMS



Powerful CPU



Gigabit Ethernet



OpenVPN



Next Generation GSM VoIP Gateway

DINSTAR

UC2000-VG-32G/T

32 Ports GSM/LTE



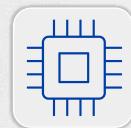
LCD



VoLTE



GSM



Powerful CPU



SMS



Gigabit Ethernet



HD audio



OpenVPN





Next Generation GSM VoIP Gateway

DINSTAR

UC2000-VH-64G

64 Ports GSM



LCD



GSM



Gigabit Ethernet



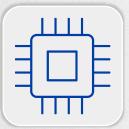
SMS



HD audio



OpenVPN



Powerful CPU





UC2000-VF / UC2000-VG



8/16/32
GSM/LTE Channels



4 SIM Slots/Channel,
SIM Rotation



SMS



VoLTE



Open API for
SMS/USSD





SIMCloud & SIMBank

Remote & Centralized SIMs Management

DINSTAR

SIMCloud

SIM Server Software

The screenshot shows the SIMCloud Management Software interface. On the left, a sidebar menu includes Configuration, Maintenance, Performance, Provision, Alarm & Log, Definition, License, Version, and User. The main area displays a tree view of system nodes under SYSTEM, SERVER, and DOMAIN. A network diagram on the right illustrates the architecture, featuring a SIM Cloud Management Server, SIM Cloud Centralized Database, and various network components like a Soft Switch, PSTN/PLMN, LAN 1, LAN 2, and SIMBANK Devices, all interconnected via a Global Forwarding Node.

SIMBank

64/128 SIM slots



Auto Recharge



Centralized Management



Scalable



SIM Allocation



Human Behavior



API

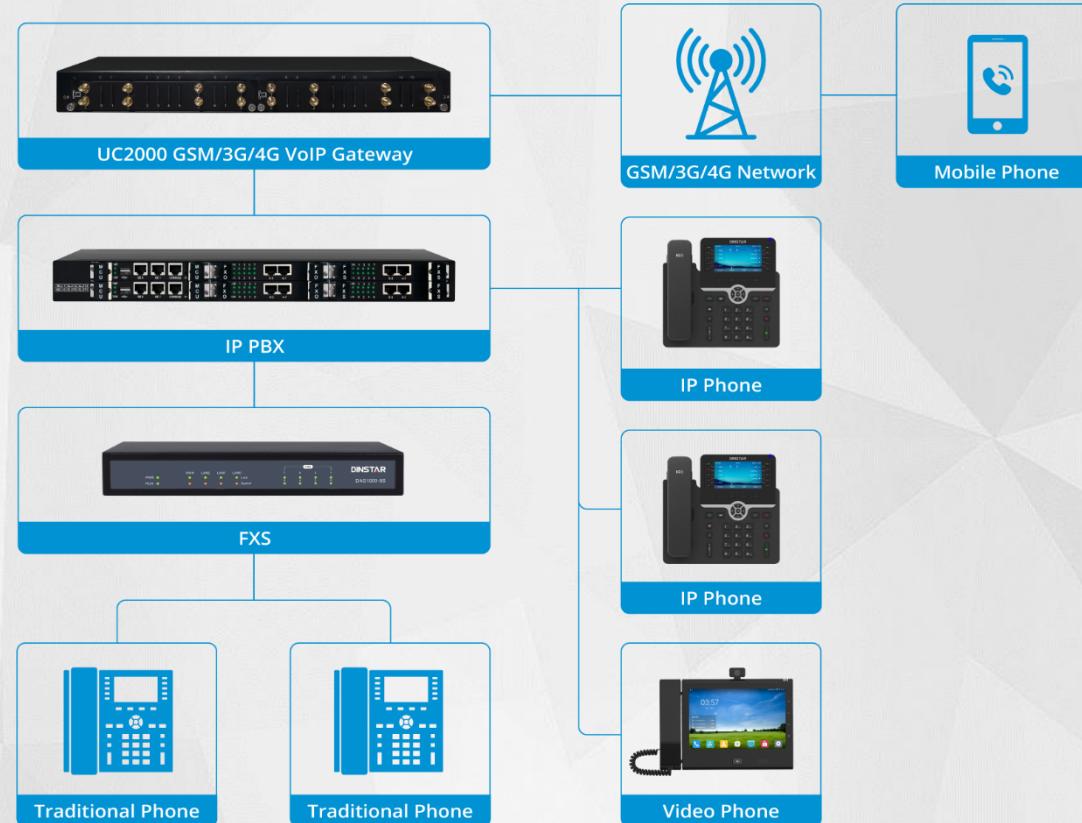


Security



Bandwidth Optimizer

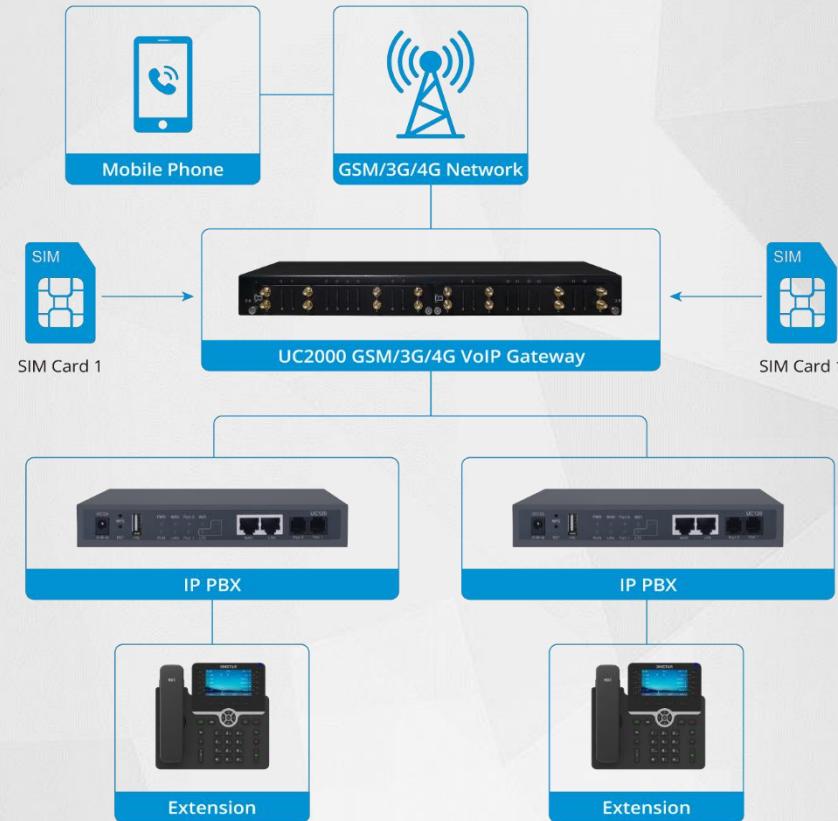
Mobile Connectivity with SME IP PBX or Phone System



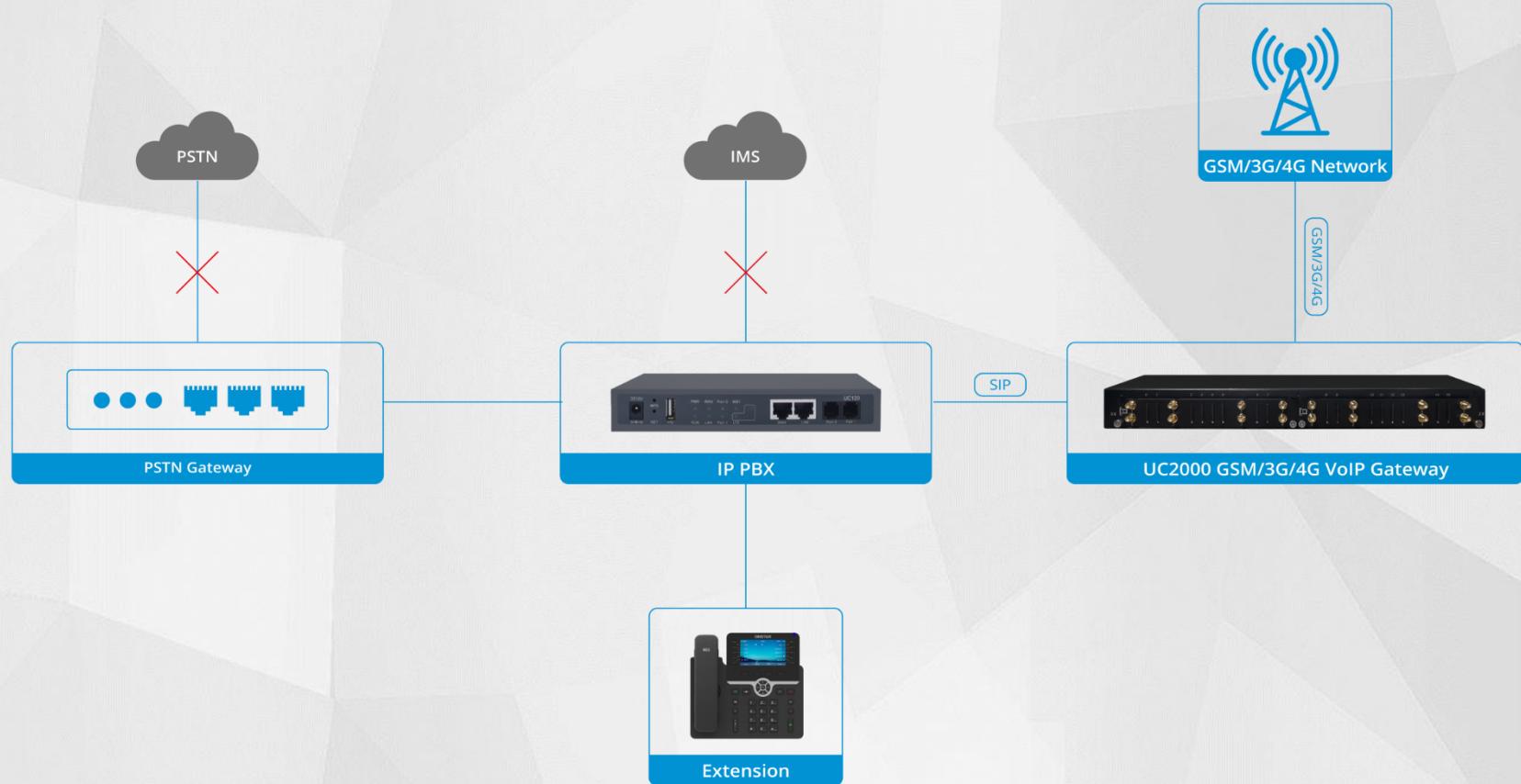
LCR Call with Different Service Provider



GSM/3G/4G Trunk for Multi-site Offices



Backup or Failover for PSTN/SIP Trunk



PSTN Replacement for Certain Areas



No PSTN Service and Fix Line



GSM/3G/4G Network

GSM/3G/4G



IP PBX



UC2000 GSM/3G/4G VoIP Gateway

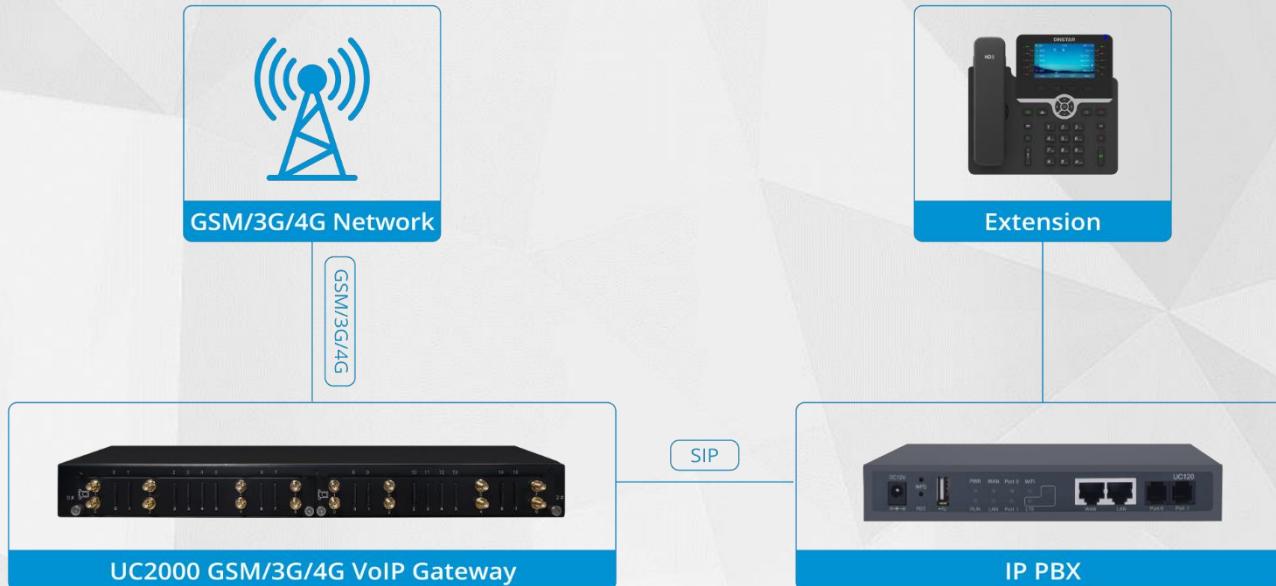
SIP



Extension

Hot desk / Co-working

No Physical Geographical Limitation



SMS at No Answer/Email to SMS



No Answer Call

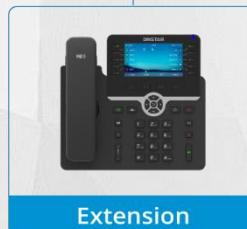


GSM/3G/4G Network

IP PBX→API: Gateways→SMS→Mobile Users



SIP



No Pick up the Call?

SIP Intercom



DP81

2 SIP Accounts
2 DSS Keys
DTMF
Default Auto Answer
Remote Control



DP82

2 SIP Accounts
2 DSS Keys
DTMF, IC Card Reader
Default Auto Answer
Remote Control



DP83

2 SIP Accounts
2 DSS Keys
DTMF
Default Auto Answer
Remote Control
HD Camera



DP85

2 SIP Accounts
2 DSS Keys
DTMF, IC Card Reader
Default Auto Answer
Remote Control
HD Camera



SIP Intercom

DP88

- 2 SIP Accounts
- 2.3" Graphic LCD Display
- Speed Dial
- Double DSS Keys
- Full Duplex
- Default Auto Answer
- Action URL/Active URI remote control
- DTMF , IC Card Reader, Password



HD Camera



PoE



2.3" LCD



IP65



Onvif



Indoor Panel



DP70

- Linux OS
- 7" TFT LCD Display
- SIP v2.0, RFC3261
- 800x480 resolution
- 10/100Mbps Ethernet
- POE
- Multi- Alarm Input



DP73

- Android 5.1 OS
- 10" TFT LCD Display
- SIP v2.0, RFC3261
- 1280x800 High Resolution
- 10/100Mbps Ethernet
- POE
- Multi- Alarm Input
- Bluetooth/WIFI

DPA Paging Gateway

- HD Video/Audio
- Dual SIP Lines, Dual SIP Servers
- SIP v2.0, RFC3261
- Support 3MP external HD camera, 1080p, 20fps/s above
- 2 ports RJ45 ports, one for Ethernet, one for HD camera
- Rich ports to connect alarm, amplifier, speaker, sensor, LED, MIC, etc.
- Self-diagnosis



HD Voice



PoE



2 SIP Lines

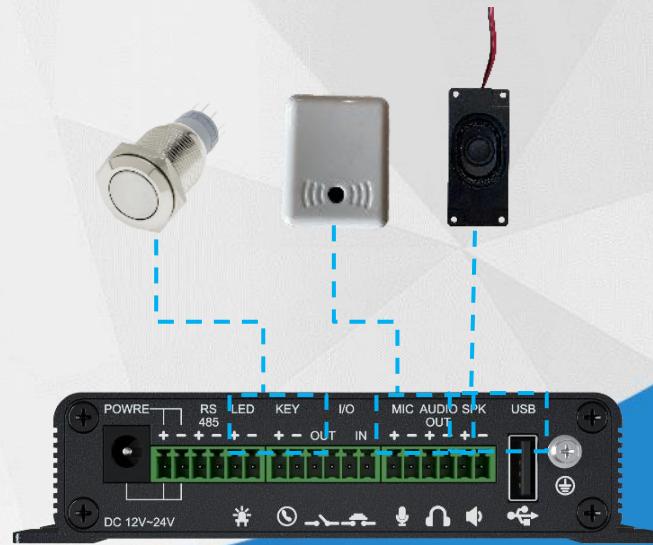


Onvif



Toolbox for DPA

- 1* 300W IP camera
- 1* DSS Button
- 1* Microphone
- 1* 2W passive Speaker



Coming Soon



HD Voice



PoE



2 SIP Lines



Onvif

DP81L-D



DP81L-S



DP81L-E



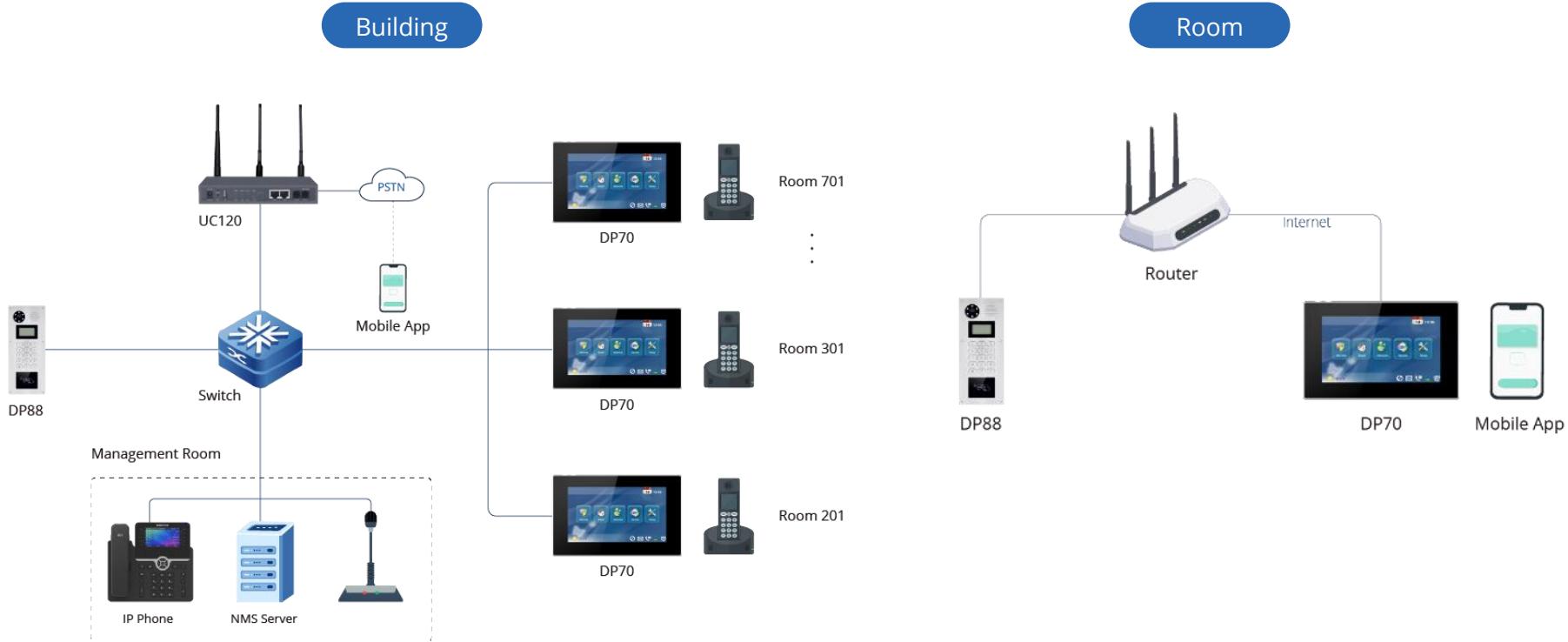
What's in the box?

- 1 set DP88
- 1 set DP83
- 1 set DP70
- 1 set DPA
- 1 set UC120
- 1 set POE Switch
- 1 set Sensor
- 1 set ceiling Speaker
- 3 set Ethernet Ports (Support External IP Phone or Analog Phone)
- 6 set Ethernet Cables



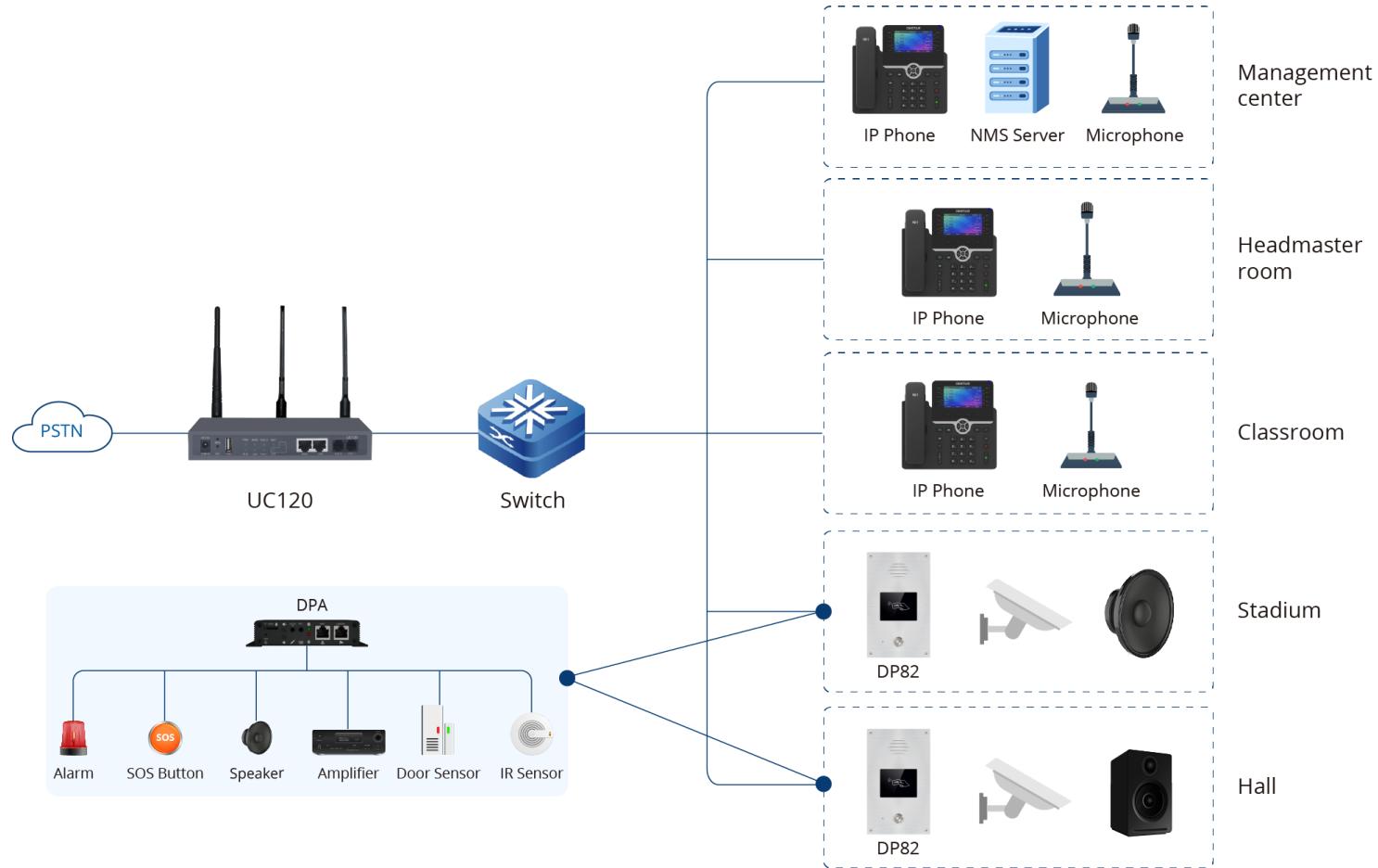


SIP Intercom Application for Residential Buildings



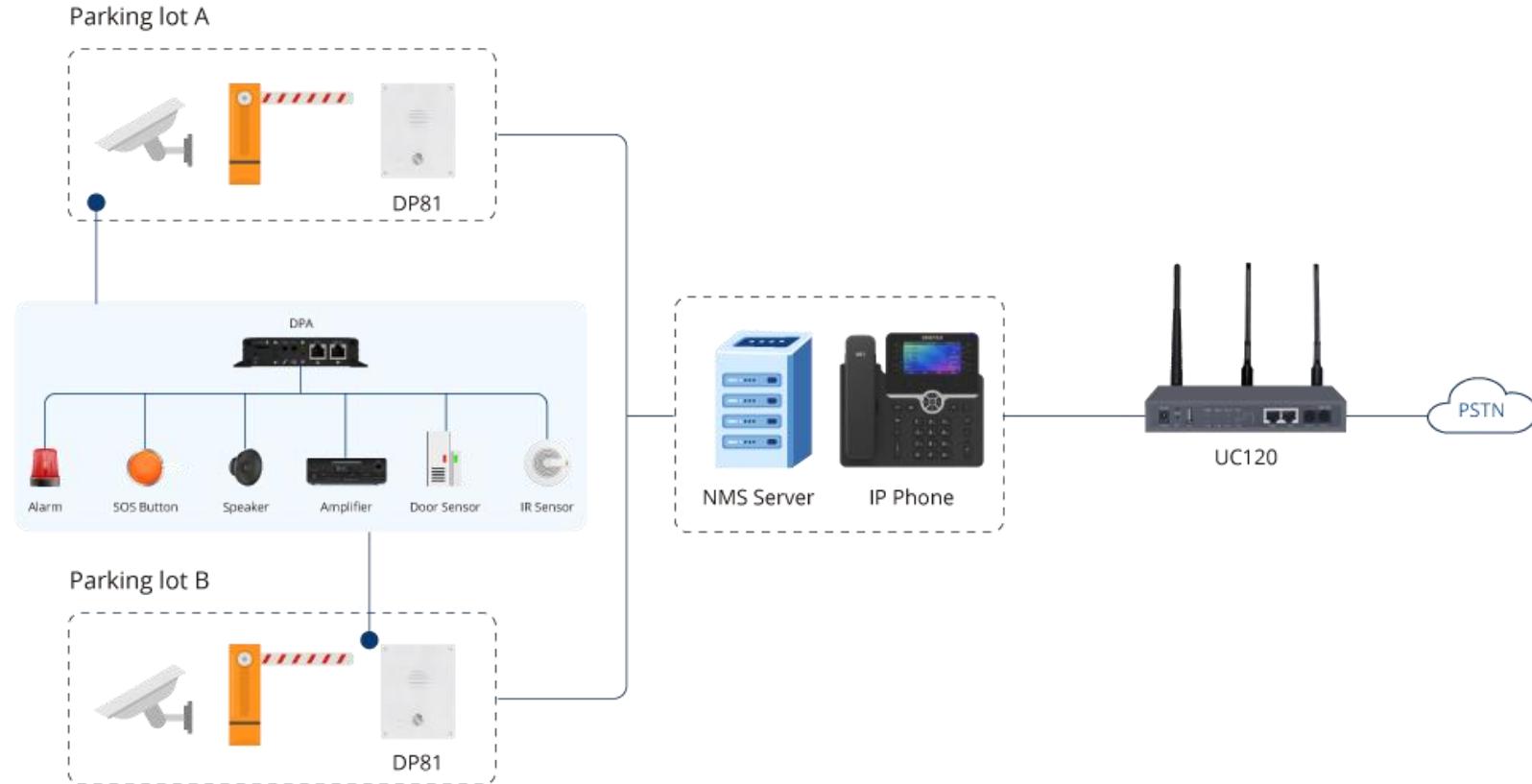


SIP Intercom Application for School



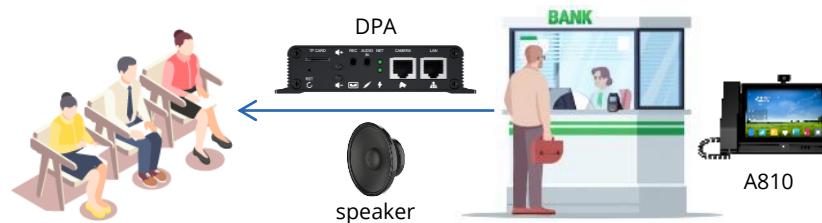


DINSTAR SIP Products Application in Parking

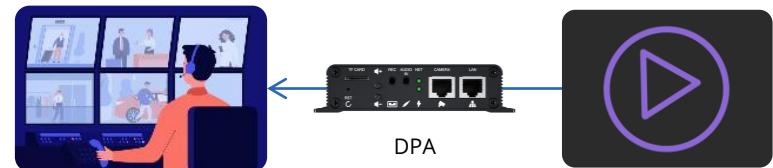




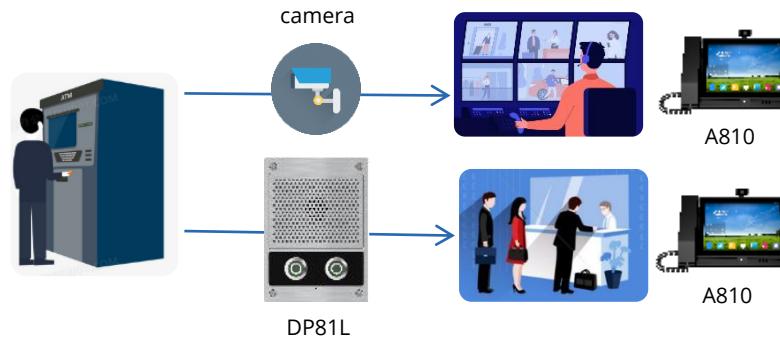
DINSTAR SIP Products Application in Bank



Broadcasting business handles queuing, registration notices, etc



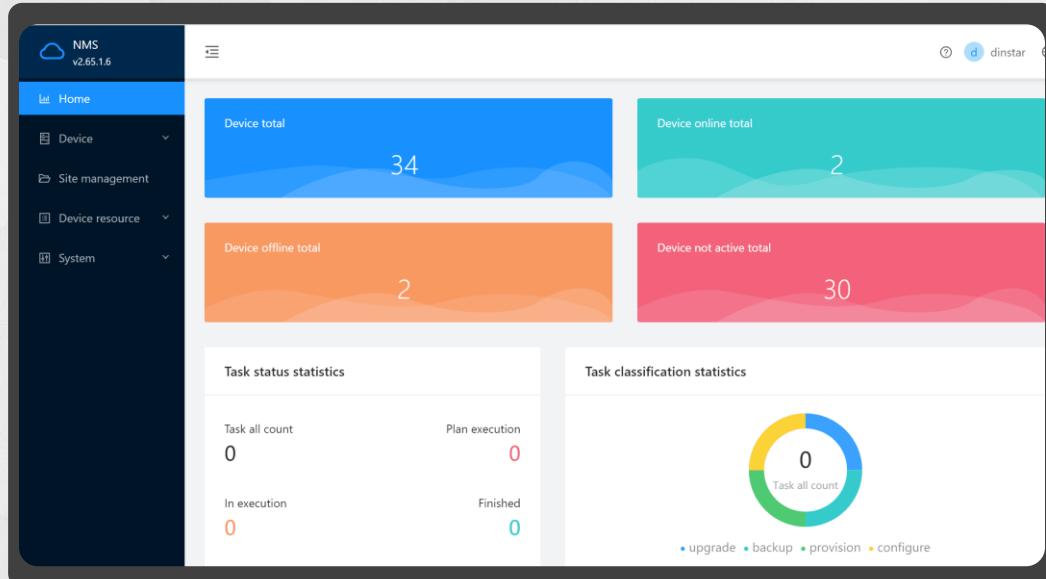
Real-time audio and video recordings



Emergency incidents (swallowing cards, coins stuck, machine failures, etc.), information consulting



Dinstar Network Management System (NMS)



DINSTAR

Device Management

IP PBXs - UC series
IP Phones – C Series IP Phone
Analog VoIP Gateways- DAGs
Digital VoIP Gateways- MTGs
Session Border Controllers-SBCs
GSM/LTE VoIP Gateways - UC2000 series
SIP Door Phones – DP series

Configuration & Firmware Management

Auto Provision for Devices
Firmware Upgrade
Auto Backup for Devices

Real-time Monitoring

Device Monitoring
Network Monitoring
Status Monitoring
Analytical System Reports
Remote Web & Telnet

Remote Management

Remote Configuration
Remote Firmware Upgrade
Remote Web GUI & Telnet
Remote Trouble-shooting



Excellent User Experience, Quick Response



Within 30 mins
Common Problems



Within 120 mins
Complex Problems



Email
support@dinstar.com



Skype
dinstar001, instar002, instar003, instar005





Thank You !



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