

# iPECS UCM Introduction

PLM & Planning Team  
2018-08-30

# Contents

- Market Trends
- Overview
- Upgrades/Improvements: iPECS CM vs. iPECS UCM
- New Application Integration
- Application Integration
- Terminal Integration
- Basic Features
- New Special Features
- Special Features



**iPECS**  
Your Communications Solution

# Market Trends

# Communication Technology Trend

Key market requirement : low TCO, smart-office support, and rich applications.

→ Cloud, Mobility, and UC are new keywords to meet those market needs.

## Cloud Computing



- Cloud computing and virtualization enable effective and efficient usage of resources
- Ecosystem based on the open architecture is needed to fit the Cloud environment

## Mobility



- Anytime, Anywhere service is expected
- Changes the way users prefer to communicate
- Ubiquity of mobile technology: text messages, voice, video, and applications for work

## Unified Communications



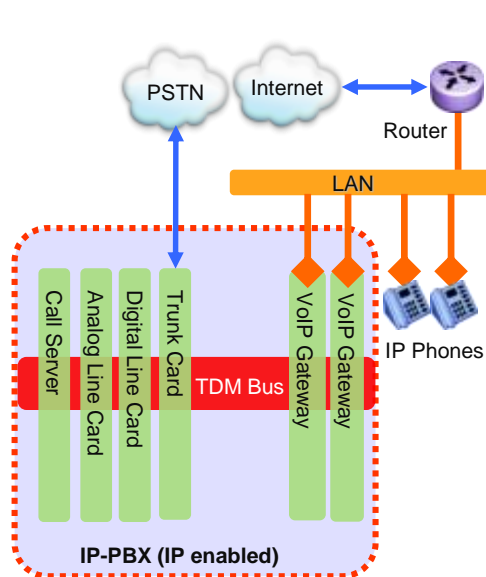
- Think Service, not Device (SaaS)
- Collaboration and enterprise application needs to be connected to each other
- Easy communication from any device, anywhere, anytime

# IP Communication Evolution

IP-PBX platform is evolving from IP enabled type to All-IP type in order to adopt various applications and devices.

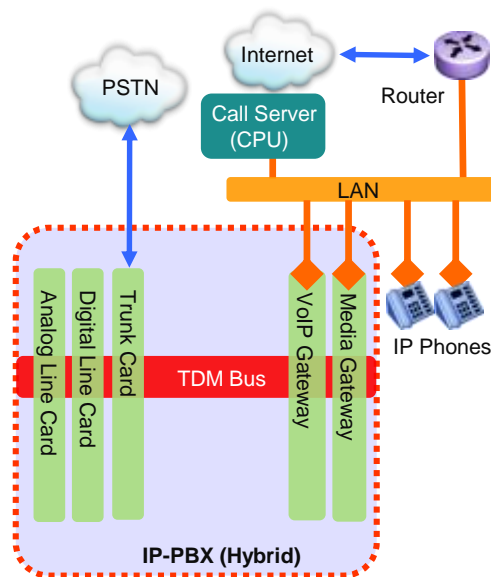
## TDM based IP-enabled

TDM Backbone Cabinet with built-in Call Server & VoIP Gateway for IP trunk and phone



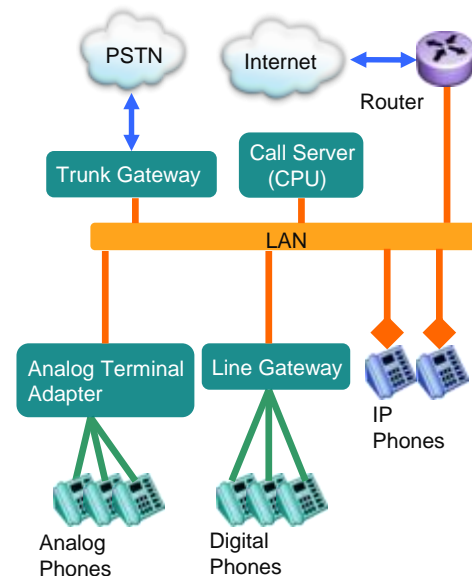
## Hybrid

TDM Backbone Cabinet with external Call Server with IP interface, VoIP Gateway for IP trunk & Media Gateway for IP phone



## All-IP

IP Backbone & VoIP Call without any interface modules or Gateways for TDM network and phone interface





# Overview

# iPECS UCM (replaces iPECS CM)



iPECS UCM

- **Next Generation Communication Platform**
  - Fully distributed architecture based on IP-backbone
  - Software-based solution running on x86 Server
  - Centralized communication and single point Management
  - Flexible Scalability from 2,000 port to 960,000 port (UCM) [30,000 port (CM)]
  - High availability with
    - Server clustering, Power/Ethernet Link/CTI/SMDR redundancy
    - Local Survivability & Failover to PSTN
    - Dual Call Server Profile on Media Gateways as well as IP Phones
- **Rich Features and Flexibility with Large Deployment Base**
  - No.1 market share in Korea and used in dozen countries world wide
  - Large scale reference sites of more than 10,000 users with multi-site configuration, UC, FMC, Contact Center, etc
  - Secured VoIP Communication with sRTP/TLS, VPN, and SBC
- **Moving to Hosted/Centrex Solution on Cloud Environment**
  - Supports virtualization
  - All-IP platform with centralized management
  - Geographical Redundancy
  - Independent Numbering Plan & System Configuration for each Tenant
  - Support variable Open Standard for 3<sup>rd</sup> party solutions

# iPECS UCM Structure

- iPECS UCM consists of a collection of call servers, gateway modules, media gateway cabinets, and software licenses.

## Software License

Reside on the call server as a file for call features and applications control

## Gateway Module

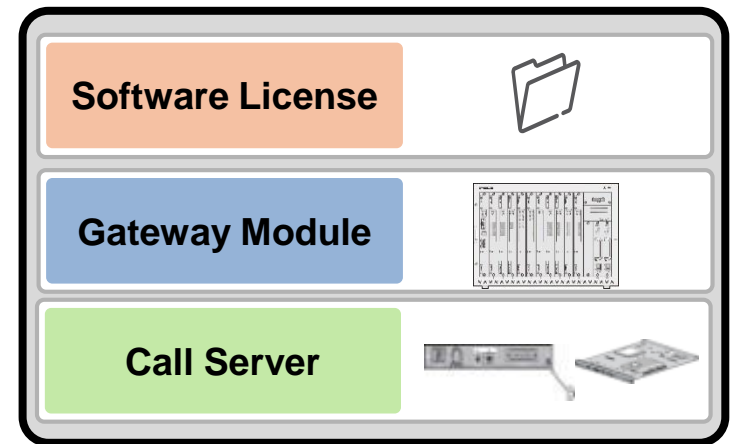
Gateway Modules to interface with PSTN/ISDN network and terminal.

## Call Server

Call control platform with CPU

## Cabinet

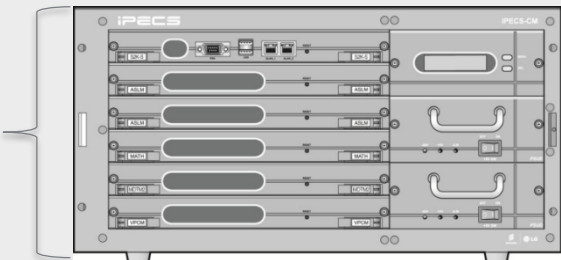
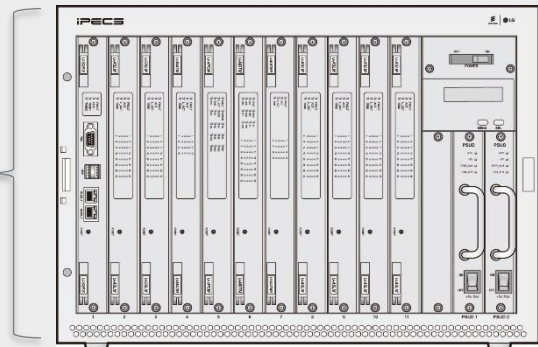
Gateway modules are inserted in Cabinet (1U/7U type)





**Cabinet**




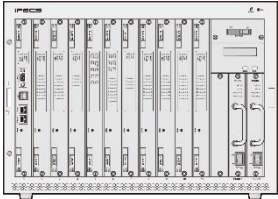

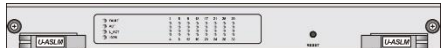
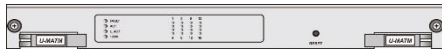
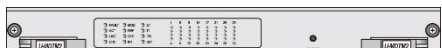

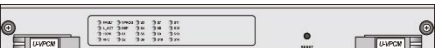
## High Density Gateway System for LME Market

H/W changes	OLD (iPECS CM)	NEW (iPECS UCM)
Media Gateway Cabinet	<b>CM-MGC2</b> 6 Slot 5U 	<b>UCM-MGC3</b> 11 Slot 7U 
Analog Subscriber Line I/F	<b>CM-ASLM</b> (32x Analog extension) <b>CM-ASLM-MW</b> (32x Analog extension + Message Waiting)	<b>UCM-ASLM</b> <b>(32x Analog extension + Message Waiting)</b>
VoIP Processing	<b>CM-VPCM</b> (VoIP processing w/ G.711/723.1/729/722)	<b>UCM-VPCM</b> (VoIP processing w/ G.711/723.1/729/722+Opus)
Digital Trunk I/F	<b>CM-MDTM</b> (1x T1/E1 Digital Trunk)	<b>UCM-MDTMX2</b> <b>(2x T1/E1 Digital Trunk)</b>

# iPECS CM H/W Component (Old)

		Design	Specification
Call Server	CM-S2K-S		<ul style="list-style-type: none"> <li>• ELG branded Industrial Computer</li> <li>• Can be installed in CM Media Gateways</li> </ul>
Media Gateway Cabinet	CM-MGC2		<ul style="list-style-type: none"> <li>• 6 universal slot, 19" Rack mountable (5U)</li> <li>• AC or DC Power Redundancy</li> <li>• Giga Ethernet Redundancy</li> <li>• LCD Alarm &amp; Fault indication</li> <li>• Built-in Ethernet Switch</li> </ul>
	CM-1URMC		<ul style="list-style-type: none"> <li>• 1 universal slot, 19" Rack mountable (1U)</li> <li>• Single AC Power</li> </ul>
Media Gateway Module	CM-ASLM (MW)		<ul style="list-style-type: none"> <li>• Analog Extension 32ports</li> <li>• CID</li> <li>• Length 8Km, Bridge Max.10ea</li> <li>• Message Waiting Lamp (CM-ASLM-MW)</li> </ul>
	CM-MATM		<ul style="list-style-type: none"> <li>• Analog Trunk 16ports</li> <li>• Support CO/E&amp;M/LD/RD with single FW</li> <li>• 4 types of sub-module (CO/E&amp;M/LD/RD)</li> </ul>
	CM-MDTM2		<ul style="list-style-type: none"> <li>• Digital Trunk 1 Link (30 Ch)</li> <li>• Support E1/T1, PRI, R2, CCS No.7 with single FW</li> </ul>
	CM-VPCM		<ul style="list-style-type: none"> <li>• Voice Conference 256 Ch</li> <li>• Voice Announcement &amp; Prompt 256 Ch</li> <li>• Voice Mail 1G (16G/32G Expandable)</li> <li>• sRTP, Transcoding 256 Ch</li> <li>• IP Packet Redirecting (NAT)</li> </ul>

# iPECS UCM H/W Component (New)

	Model Name	Specification	
Call Server	UCM-S2K-S*	<ul style="list-style-type: none"> <li>• ELG branded Industrial Computer</li> <li>• Can be installed in UCM Media Gateways</li> </ul>	
Media Gateway Cabinet	UCM-MGC3**	<ul style="list-style-type: none"> <li>• 11 universal slot, 19" Rack mountable (7U)</li> <li>• DC Power Redundancy</li> <li>• Giga Ethernet Redundancy</li> <li>• LCD Alarm &amp; Fault indication</li> <li>• Built-in Ethernet Switch</li> </ul>	
	UCM-1URMC*	<ul style="list-style-type: none"> <li>• 1 universal slot, 19" Rack mountable (1U)</li> <li>• Single AC Power</li> </ul>	
Media Gateway Module	UCM-ASLM**	<ul style="list-style-type: none"> <li>• Analog Extension 32ports</li> <li>• CID</li> <li>• Length 8Km, Bridge Max.10ea</li> <li>• Message Waiting Lamp</li> </ul>	
	UCM-MATM*	<ul style="list-style-type: none"> <li>• Analog Trunk 16ports</li> <li>• Support CO/E&amp;M/LD/RD with single FW</li> <li>• 4 types of sub-module (CO/E&amp;M/LD/RD)</li> </ul>	
	UCM-MDTM2*	<ul style="list-style-type: none"> <li>• Digital Trunk 1 Link (30 Ch)</li> <li>• Support E1/T1, PRI, R2, CCS No.7 with single FW</li> </ul>	
	UCM-MDTMX2**	<ul style="list-style-type: none"> <li>• Digital Trunk 2 Link (60 Ch)</li> <li>• Support E1/T1, PRI, R2, CCS No.7 with single FW</li> </ul>	
	UCM-VPCM**	<ul style="list-style-type: none"> <li>• Voice Conference 256 Ch</li> <li>• Voice Announcement &amp; Prompt 256 Ch</li> <li>• Voice Mail 32 GB</li> <li>• sRTP, Transcoding 256 Ch</li> <li>• IP Packet Redirecting (NAT)</li> </ul>	

\* New faceplate & Old H/W

\*\* New faceplate & New H/W

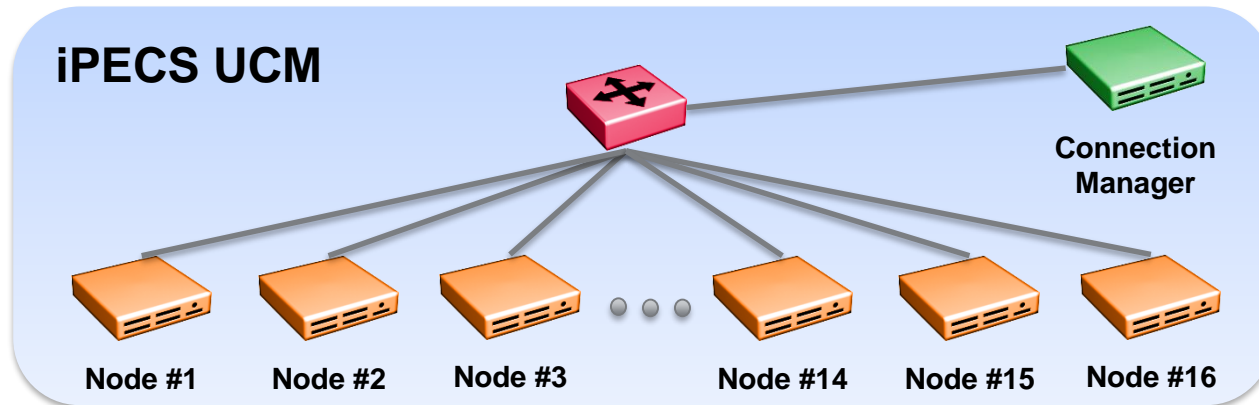
# iPECS UCM S/W Highlight

**Distributed DB architecture**

**Capacity up to 960K users (16 nodes)**

**Active-Active failover**

**Cloud ready (Virtualization)**



**Reliable and secure**

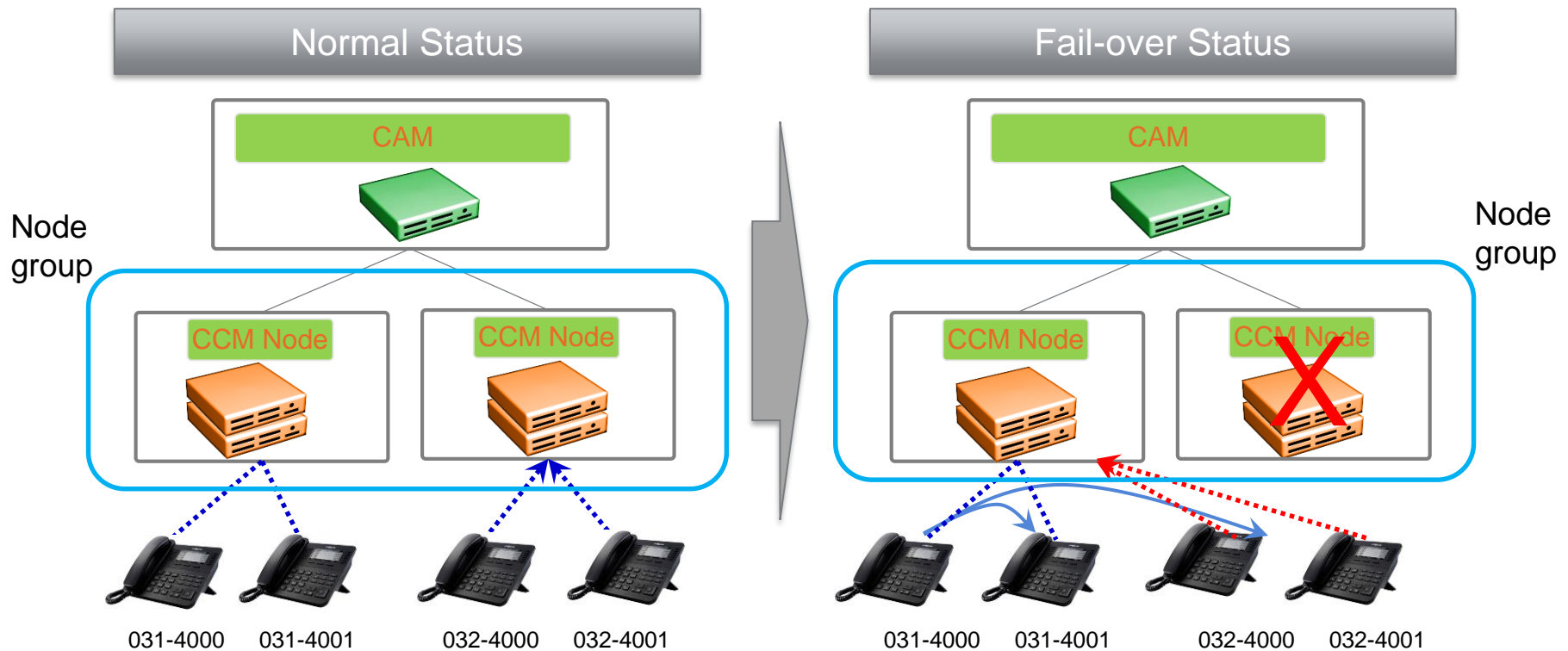
**Scalable and flexible**

**Versatile multi-call/messaging control**

**APIs for 3rd party interworking**

# iPECS UCM High Reliability (A-A)

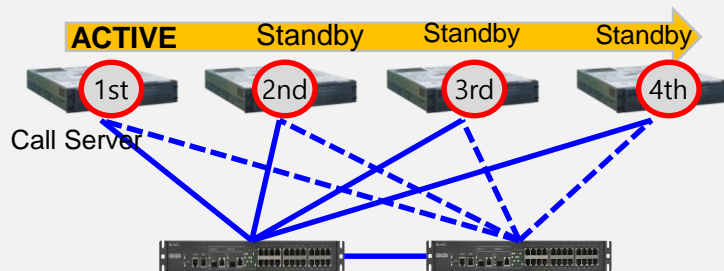
- Active-Active mode support (16 node / 8 node groups)
- Per node redundancy support (4 per node / up to 64 standby servers)
- Per node local survivability support (1024 per node / up to 8192 LCM sites)
- CAM server: local redundancy only



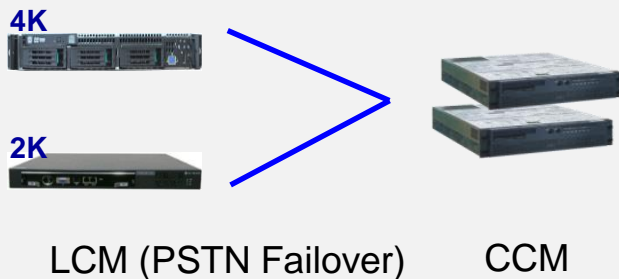
# iPECS UCM High Reliability (A-S)

- Local and Geographical Redundancy (16 server/8 region)
- Redundancy for Ethernet Link, Power, CTI Link, SMDA Link
- Local survivability and PSTN Fail over for WAN Fail
- Backup Call Server for GW and IP Phone in case LCM Fail

## Server Redundancy



## Remote Survivability



## Major Parts Redundancy

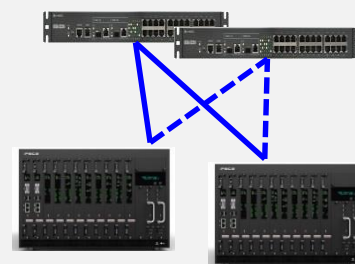
### Call Server



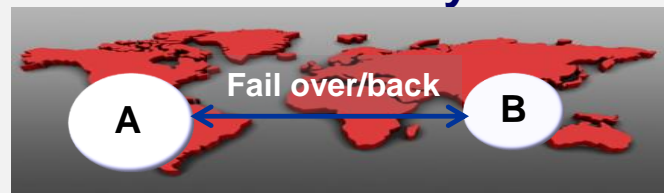
### Media Gateway



Power LAN Link



## Geo-Redundancy



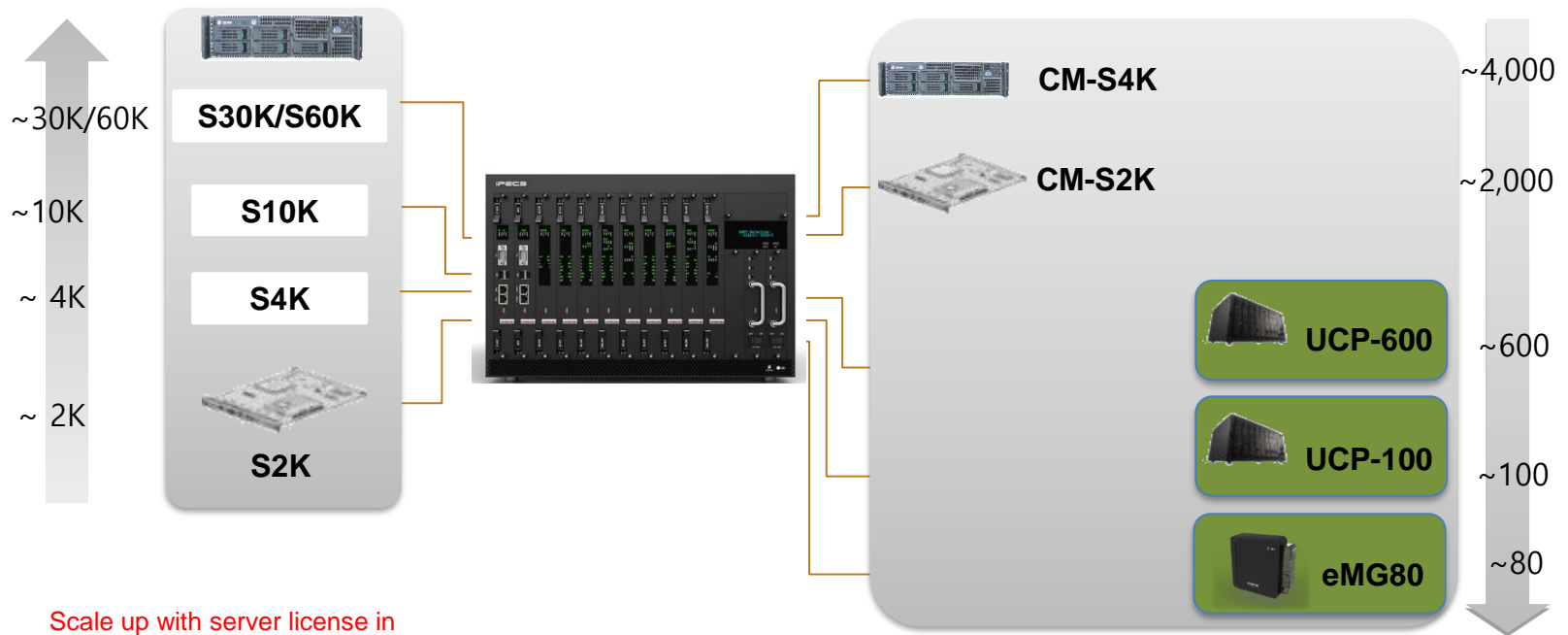
Master Region  
(Headquarter)

Stand-by Region  
(One of branch)

# iPECS UCM Flexible Scalability

- CCM node can be scaled up from 2K to 60K ports.
- LCM, which can be installed in up to 1024 sites per CCM, can be scaled down from 4,000 ports to 80 ports so that customer can choose very cost effective branch office solution.

## Scale Up for CCM



Scale up with server license in  
x86 server

## Scale Down for LCM

# iPECS UCM Security

**iPECS UCM is a proven secure communication solution with encryption, enhanced authentication, and access control.**

## Security management on Call Server

- System access control using **ACL(Access Control List)**
- **Subscriber authentication**

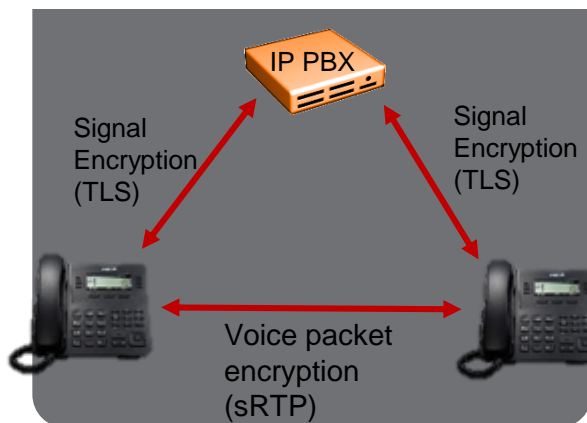
**IP address range control**

**SIP authentication**

**Strong User/Admin password**

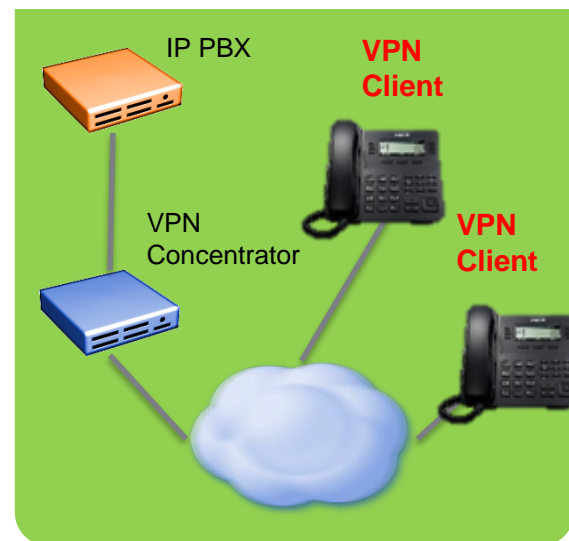
## Encryption between IP-PBX and IP Phone

- Protocol : TLS, sRTP
- **Algorithm : AES, ARIA (Korea)**
- **Key Exchange : ECC, RSA**
- Certificate : DER, PEM



## IP Phone VPN & 802.1x

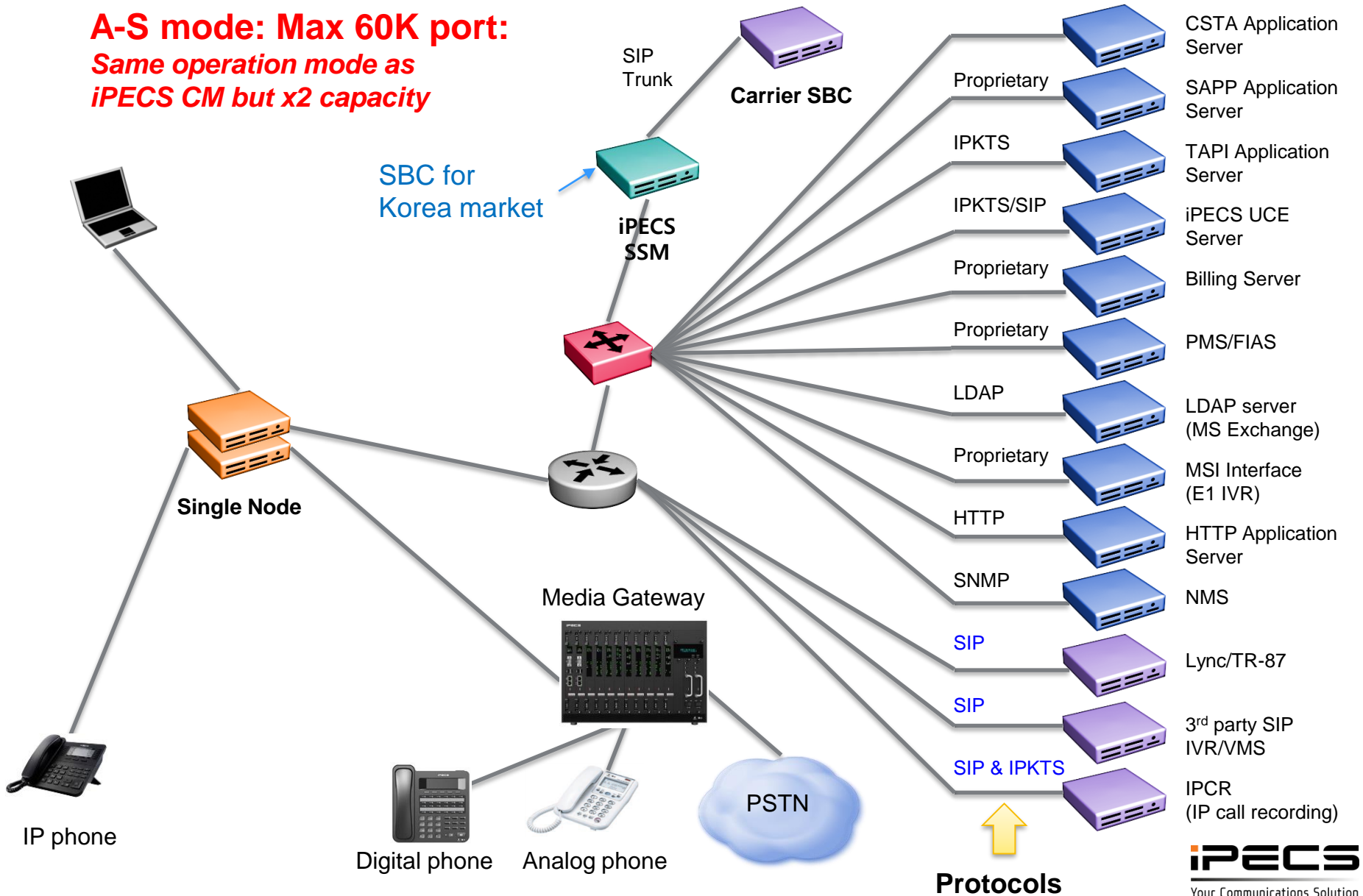
- **IP Phone + embedded VPN client**
- 802.1x : enhanced access control





# iPECS UCM Network Configuration

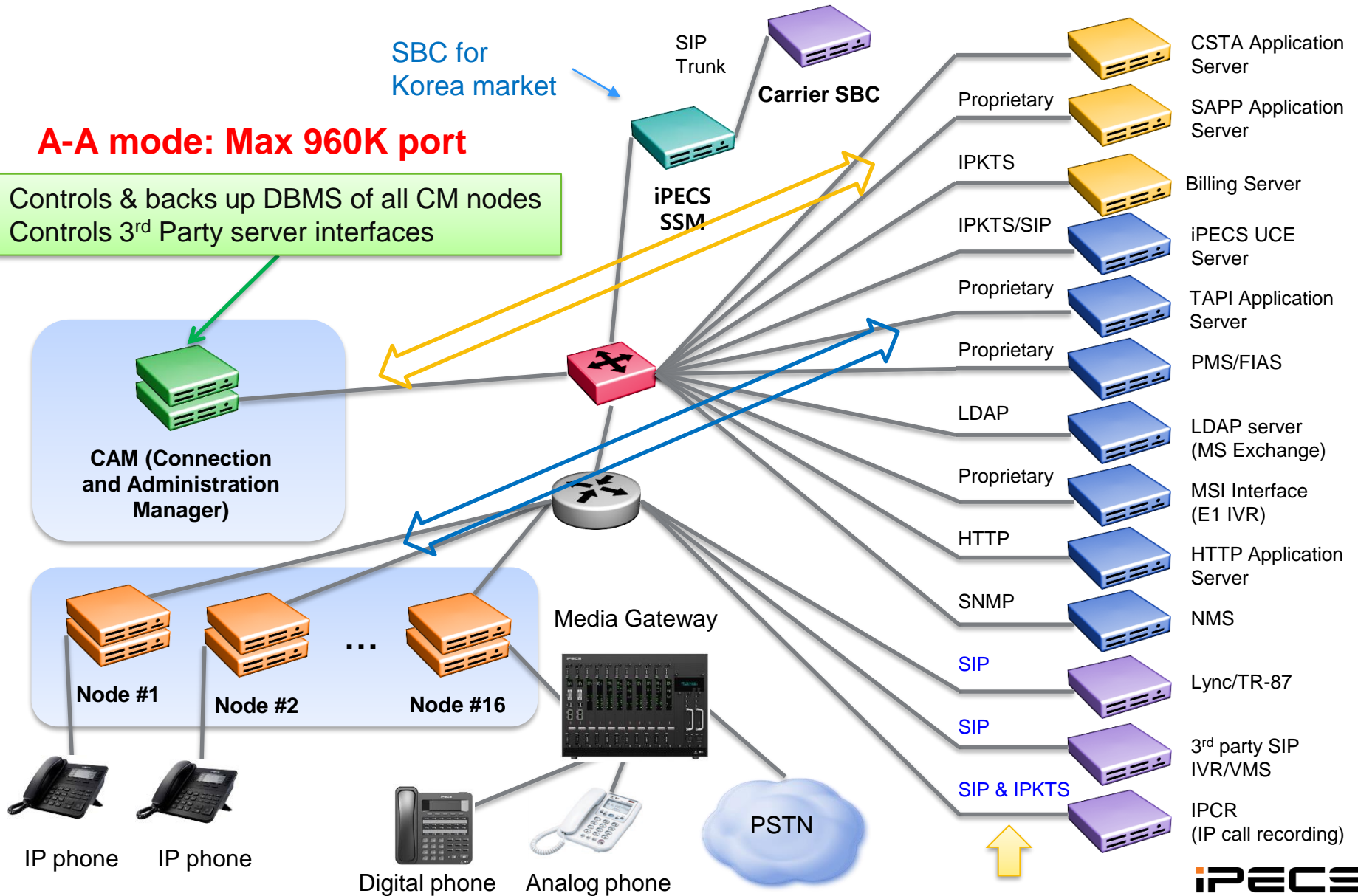
**A-S mode: Max 60K port:**  
**Same operation mode as**  
**iPECS CM but x2 capacity**



# iPECS UCM Network Configuration

## A-A mode: Max 960K port

Controls & backs up DBMS of all CM nodes  
Controls 3<sup>rd</sup> Party server interfaces



# iPECS UCM Feature Highlight

**Wide range of IP telephony features**

**Multi-tenant support up to  
65,000 tenant groups**

**Multiple interfaces for integrating  
external VMS/IVR**

**Virtual subscriber service**

**Spam call protecting features for  
specific phone number**

**CID Conversion and  
ARS digit conversions**

**Automatic QoS through  
LLDP and LLDP-MED for IP Phones**

**Multiple phone numbers for  
one SIP Phone, up to 30 numbers**

**Broadcasting and interphone  
for 100 party/group**

**Push-to-Talk (PTT) for 128 party/group  
(99 group/tenant)**

**Built-in voice conference with  
2000 rooms/tenant (128 party/room)**

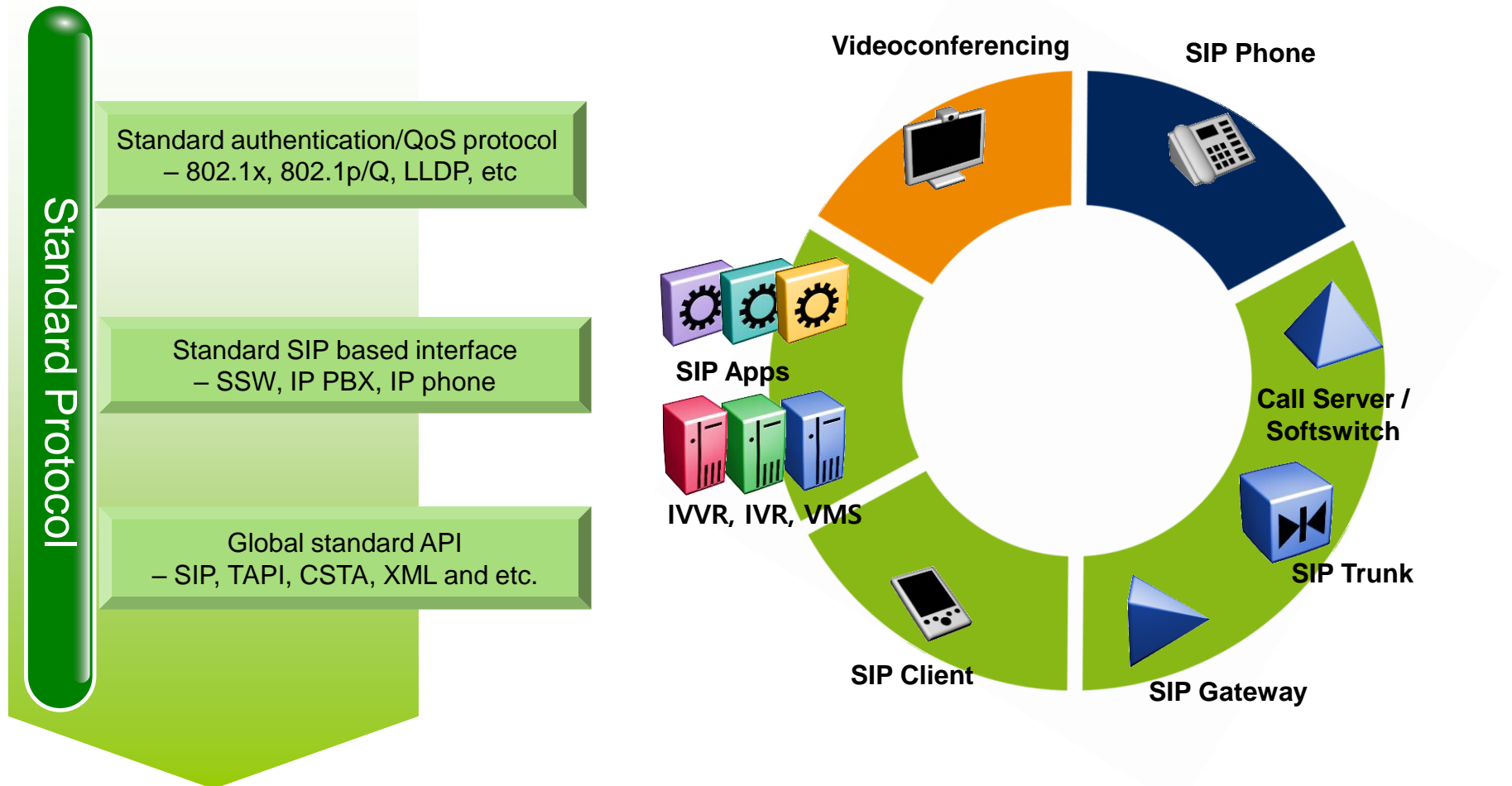
**Built-in auto-attendant and  
voice mail with email notification**

**Mobile extension**

**Hot desk for sharing a single desktop  
phone with multiple users**

# iPECS UCM Openness

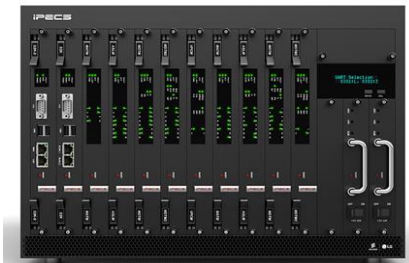
- Flexible and converged configuration is available based on standard protocol.



# iPECS UCM External VMS/IVR Interface

- Multiple interfaces to integrate the external VMS/IVR are supported to meet various customer environment

## iPECS-UCM



MDTM

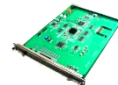


1

E1 Digital Trunk (Max 30)

LAN (UDP/IP) for MSI

MATM



2

Analog Trunk (Max 32)

3

VPCM (through LAN) SIP Trunk

Apply as a standard  
SIP phone

## External- VMS/IVR/IVVR



***Up to 8 VMS Links (16 VMS in case Redundancy) are supported***

# iPECS UCM Voice Networking

- Ability to interconnect with other PBX through both QSIG and H.323 in order to expand total system capacity supporting various supplementary services.

	QSIG on PRI (ETSI Standard)	H.323 on IP
Interface Gateway	MDTM-PRI (signal & media)	H323M (signal) VPCM (media)
Codec	TDM	G.711 & G.723 & G.729
Supplementary Service Protocol	Standard QSIG	Standard H.450
Supported Features	Basic Call(Net Station Call), Transit In/Out, Net Line Identification Net Name Identification, Net 3 <sup>rd</sup> Party Conference, Net Call Transfer By Reroute, Net Call Completion(Call Back) Net Call Offer, Net Call Intrusion, Net Call Forward (register/cancel/service) Net Do Not Disturb, Net Message Wait Indication(MWI) Centralized VMS(Voice Mail system), Centralized ATD(Attendant) Service(CAS)	

# Comprehensive Mobility Solution

- Multiple choices of mobility solutions for office and mobile environments

## IP-DECT

- GDC-800Bi / GDC-800H

## Hot Desk

## UCE Mobile Clients

- Unified Comm. enabled Mobile Client
- IM, Presence, Video call, LDAP Integration...
- Call back / Call Through / VCC

## 3<sup>rd</sup> party mobile Apps

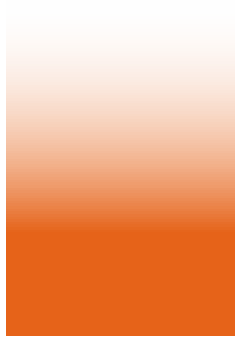
- Supported with 3<sup>rd</sup> party SIP license
- Recommended App: CounterPath Bria

## Mobile Extension

- Make Mobile phone as a PBX station
- One-Number Service

Office

Mobile



**iPECS**

Your Communications Solution

# Upgrades/Improvements iPECS CM vs. iPECS UCM



# iPECS UCM Capacity Upgrade

Item	CM 5.5	UCM	
		ACT-STB	ACT-ACT
			(w/ Max 16 nodes)
System Port	30,000	60,000 / system	60,000 / node (960,000 / system)
Extension	30,000	60,000 / system	60,000 / node (960,000 / system)
Trunk Port	10,000	40,000 / system	40,000 / node (640,000 / system)
BHCC*	450,000	720,000 / node	
Media Gateway Module	8,000	65,000	
LCM	255	1,024 / node	
Tenant	254	65,000 / system	
System Speed Code	3,000 / tenant	9,999 / tenant	
Conference Room	100 / system	2,000 / tenant	

\* 3.5GHz x86 server needed


# iPECS UCM Improvements (1/4)

Category	iPECS CM P5.5	iPECS UCM
Server Redundancy	ACT-STB Failover	ACT-STB Failover <b>ACT-ACT Failover</b> Local Survival Solution: LCM registration priority option is added (CAM server: local redundancy only)
Server Configuration	CCM : 1 LCM : 250	CAM : 1 CCM : 16 Node <b>LCM : 1024/CCM, max 8192/system</b>
Performance (BHCC)	450,000 BHCC/system	720,000 BHCC/node
Capacity (IPKTS)	30K subscriber/system	60K subscriber/node <b>Max 16 node/system (960K/system)</b>
Capacity (SIP)	UDP 10,000 user/system TLS 10,000 user/system SIP 10,000 session/system	UDP 60,000 user/node TLS 40,000 user/node <b>SIP 20,000 session/node</b>
SIP Auto Provisioning	Not supported	SIP phone setting is automatically provisioned by User ID/PW or MAC
Application Server	NMS	eNMS (upgraded NMS) eVQM (Voice Quality Monitoring) eCSM (Call Statistics Management) E-SBC (standalone SBC) E-VM (standalone Voicemail System)

# iPECS UCM Improvements (2/4)

Category	iPECS CM P5.5	iPECS UCM
DBMS	Proprietary Relational DBMS	<b>SQL Relational DBMS (Firebird) with capacity increase</b> <ul style="list-style-type: none"> <li>• Full ACID compliant transactions</li> <li>• Referential integrity</li> <li>• Multi Generational Architecture</li> <li>• Support for External Functions (UDFs)</li> <li>• SQL activity can send asynchronous notification events to clients</li> <li>• Third-party tools, including GUI administrative tools and replication tools</li> <li>• Careful writes - fast recovery, no need for transaction logs</li> <li>• <b>Many access methods:</b> native/API, dbExpress/FireDAC drivers, ODBC, OLE DB, .NET provider, JDBC native type 4 driver, Python module, PHP, Perl</li> <li>• Incremental backups</li> <li>• Full cursor implementation in PSQL</li> </ul>
Data Security	No DB encryption	<b>DB encryption for personal data</b>

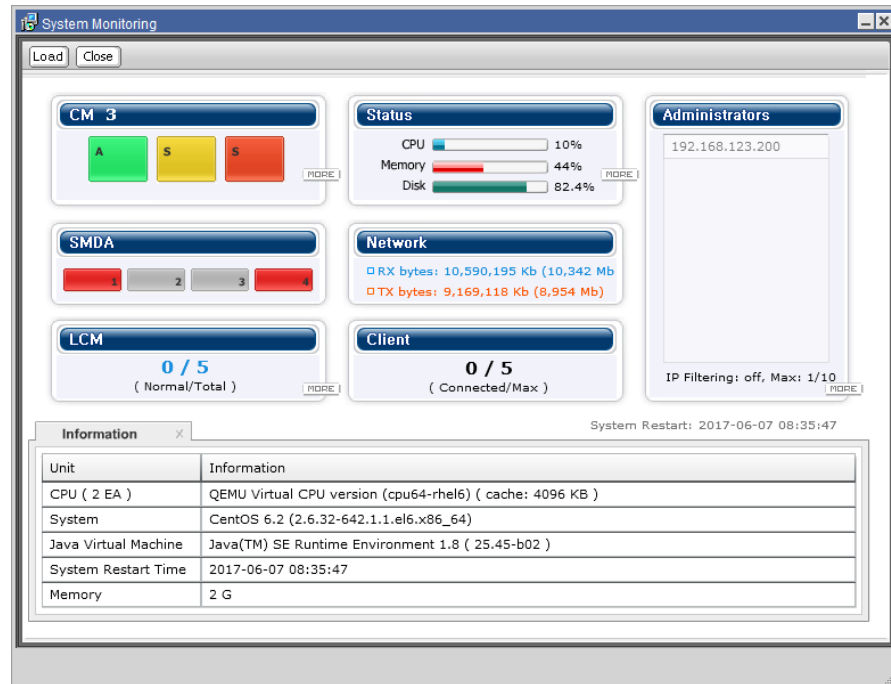
# iPECS UCM Improvements (3/4)

Category	iPECS CM P5.5	iPECS UCM
Easy Admin	N/A	<ul style="list-style-type: none"> <li>Improved access speed (e.g. x4 speed improvement for mass data generation, summary information)</li> <li>System Monitoring UI upgrade</li> <li>System Configuration UI upgrade</li> <li>Simplified VPCM UI</li> <li>Expanded virtual gateway UI</li> <li>Alarm/Fault/Status summary window is added</li> <li>More user friendly UI</li> </ul> 
3 <sup>rd</sup> party API	TAPI CSTA SAPP	TAPI CSTA SAPP <b>Restful CM API (2018 Q1)</b>
Easy Upgrade	Manual DBMS upgrade is needed for system S/W upgrade	Manual DBMS upgrade is <b>NOT</b> needed for system S/W upgrade
Web Brower	ActiveX Support Browser Uses JVM	<b>HTML5</b> Support Browser JVM not used

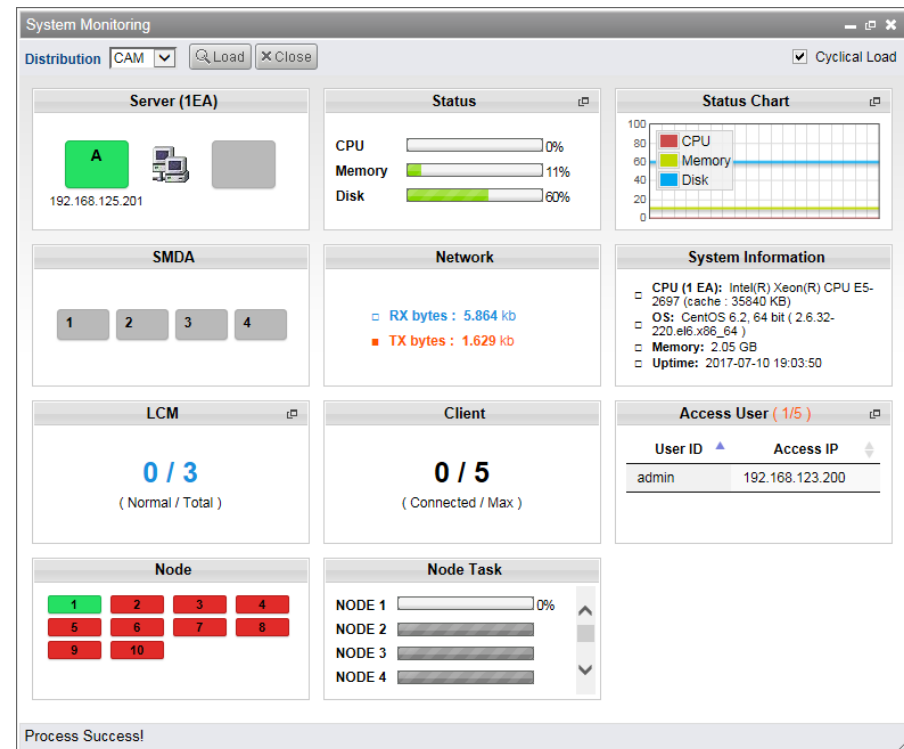
# iPECS UCM Improvements (4/4)

Category	iPECS CM P5.5	iPECS UCM
Enhanced Audio Conference bridge (P2 : coming soon)	N/A	<ul style="list-style-type: none"> <li>S/W based &amp; virtualized (x86 server)</li> <li>Authentication with Security Code</li> <li>Participant Name Recording &amp; Play</li> <li>Mute / Unmute</li> <li>Web-based conference management</li> </ul>
VPCM delay reduction (P2 : coming soon)	N/A	<ul style="list-style-type: none"> <li>IP Redirection</li> <li>MDTM loopback</li> </ul>
VQM support for G/W & IPKTS phone (P2 : coming soon)	N/A	<ul style="list-style-type: none"> <li>SIP RTCP Summary Report (RFC6035)</li> <li>RTCP-XR info</li> </ul>
Active directory user info field upgrade (P2 : coming soon)	Name & Mobile number	<ul style="list-style-type: none"> <li>Name &amp; Mobile number</li> <li>Title field &amp; Department field</li> <li>E-mail field</li> </ul>
S/W VPCM (P2.x : 2018.1H)	N/A	<ul style="list-style-type: none"> <li>S/W based &amp; virtualized (x86 server)</li> <li>Smaller footprint than H/W VPCM in datacenter</li> <li>Higher codec density</li> </ul>
ACD CTI Feature (P3 : 2018.2H)	IPKTS supported	<ul style="list-style-type: none"> <li>IPKTS &amp; SIP supported</li> </ul>

# System Monitoring UI upgrade

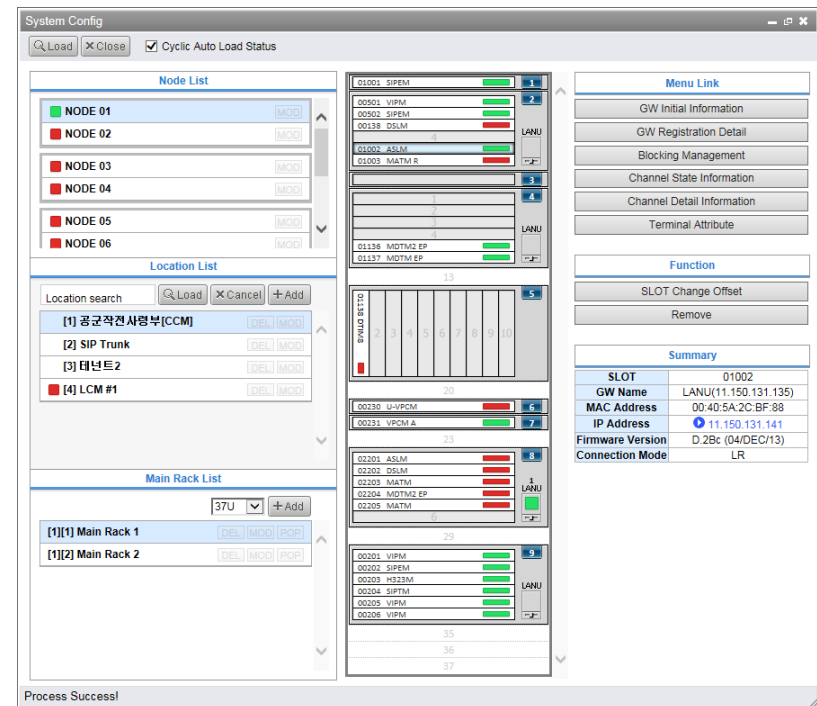
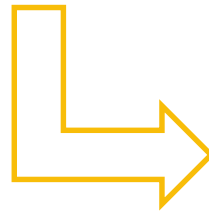
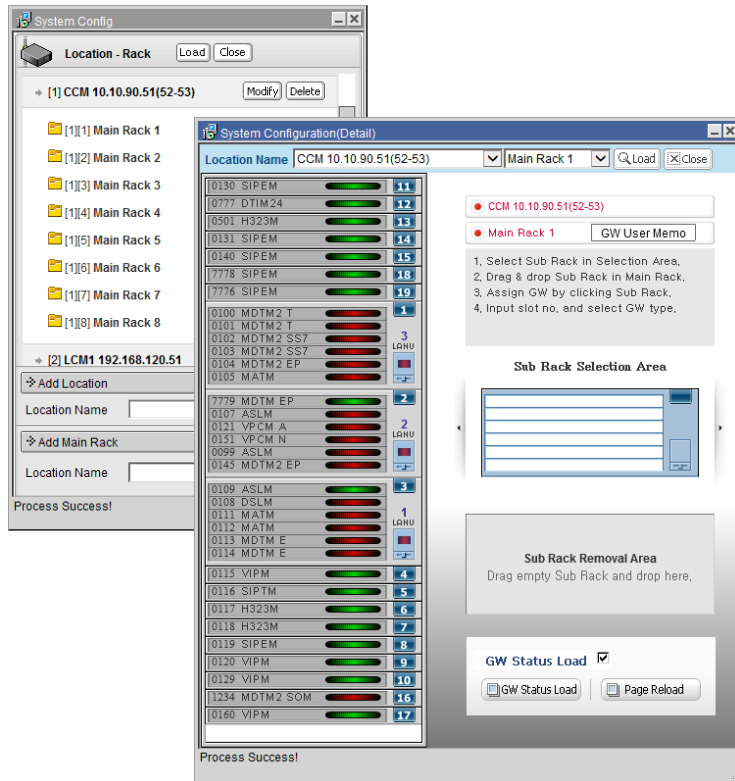


- UI change & custom layout
- CPU/Memory/Disk usage graph



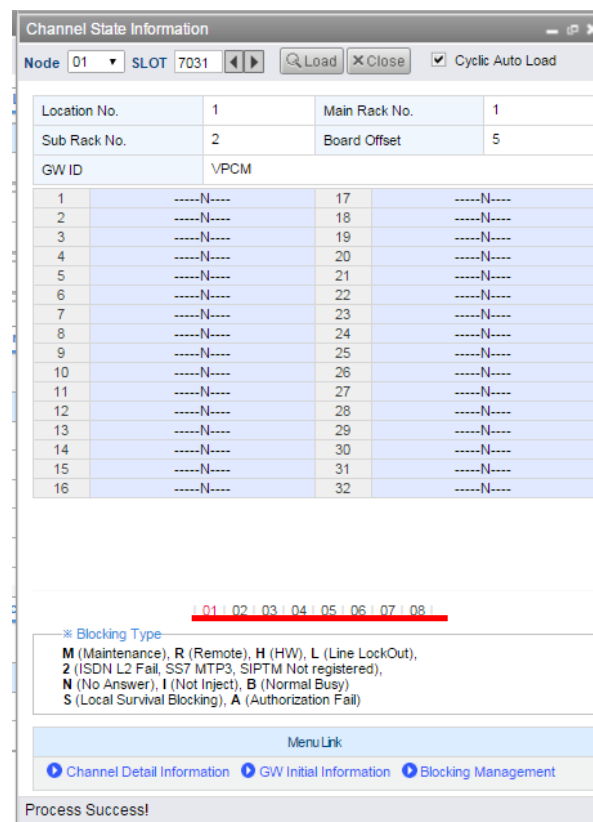
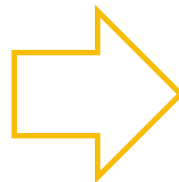
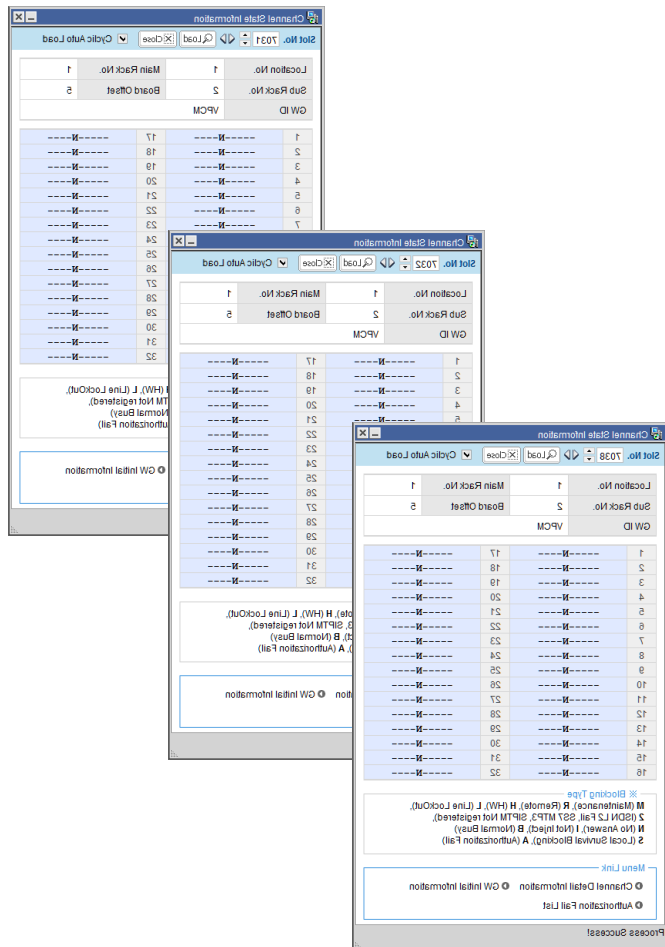
# System Configuration UI upgrade

- All info access in one screen
- LCM status icon
- Gateway summary and related menu display
- And many others ...



# Simplified VPCM UI

32 channel 8 slot → 265 channel 1 slot



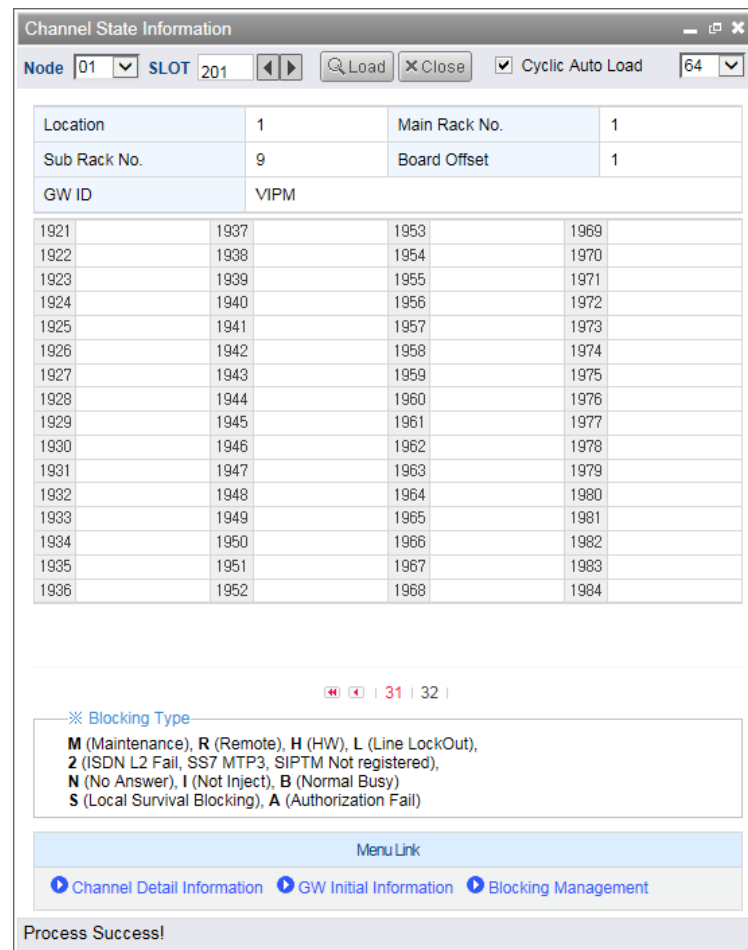
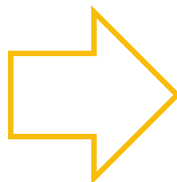
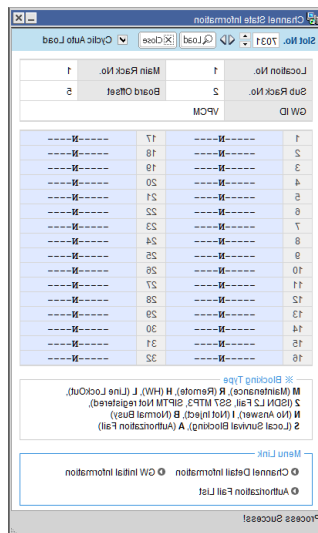


# Expanded virtual gateway UI

CM Virtual G/W (VIPM, SIPEM, SIPTM, H323M) slot used 32 channel

→ UCM Virtual G/W slot can use 8192 channel

(VIPM: IPKTS Phone, SIPEM: SIP Phone, SIPTM: SIP Trunk, H323M: H323 Trunk)



# Alarm/Fault/Status summary

## Alarm/Fault/Status summary with filters

- Easier error message tracking
- Easier temporal tracking

Alarm/Fault/Status Message Summary

Type 

Alarm

 Level 

ALL

 Date 

2016-01-30

00

 : 

00

 ~ 

2016-01-30

23

 : 

59

 Keyword 

Load

Close

Total Count 44 , 1/5 Page

Date	Type	Code	Grade	Information
2016-01-30 11:43:50	ALARM	<a href="#">A0018</a>	Major	LANU NO ANSWER FAIL ALARM ON LOC:NODE01/LANU/LANU0040/LANU INF:S_LOC0001/M_RACK0001(32U)/S_RACK04(CM-MG)
2016-01-30 11:43:50	ALARM	<a href="#">A0018</a>	Major	LANU NO ANSWER FAIL ALARM ON LOC:NODE01/LANU/LANU0021/LANU INF:S_LOC0002/M_RACK0001(32U)/S_RACK02(CM-MG)
2016-01-30 11:43:50	ALARM	<a href="#">A0018</a>	Major	LANU NO ANSWER FAIL ALARM ON LOC:NODE01/LANU/LANU0020/LANU INF:S_LOC0001/M_RACK0001(32U)/S_RACK02(CM-MG)
2016-01-30 11:43:50	ALARM	<a href="#">A0018</a>	Major	LANU NO ANSWER FAIL ALARM ON LOC:NODE01/LANU/LANU0013/LANU INF:S_LOC0004/M_RACK0001(32U)/S_RACK01(CM-MG)
2016-01-30 11:43:50	ALARM	<a href="#">A0018</a>	Major	LANU NO ANSWER FAIL ALARM ON LOC:NODE01/LANU/LANU0012/LANU INF:S_LOC0003/M_RACK0001(32U)/S_RACK01(CM-MG)
2016-01-30 11:43:50	ALARM	<a href="#">A0018</a>	Major	LANU NO ANSWER FAIL ALARM ON LOC:NODE01/LANU/LANU0011/LANU INF:S_LOC0002/M_RACK0001(32U)/S_RACK01(CM-MG)
2016-01-30 11:43:50	ALARM	<a href="#">A0018</a>	Major	LANU NO ANSWER FAIL ALARM ON LOC:NODE01/LANU/LANU0010/LANU INF:S_LOC0001/M_RACK0001(32U)/S_RACK01(CM-MG)
2016-01-30 11:43:26	ALARM	<a href="#">A0004</a>	Critical	GATEWAY NO ANSWER FAIL ALARM ON LOC:NODE01/GW/SLOT07041/PCM INF:S_LOC0001/M_RACK0001(32U)/S_RACK02(CM-MG)/OFFSET06
2016-01-30 11:43:26	ALARM	<a href="#">A0004</a>	Critical	GATEWAY NO ANSWER FAIL ALARM ON LOC:NODE01/GW/SLOT07031/PCM INF:S_LOC0001/M_RACK0001(32U)/S_RACK02(CM-MG)/OFFSET05
2016-01-30 11:43:26	ALARM	<a href="#">A0004</a>	Critical	GATEWAY NO ANSWER FAIL ALARM ON LOC:NODE01/GW/SLOT07021/PCM INF:S_LOC0001/M_RACK0001(32U)/S_RACK01(CM-MG)/OFFSET06

01 | 02 | 03 | 04 | 05 |

Process Success!

# More User Friendly UI

Actual used items are displayed

Tenant Use List

Node 01

System Numbering Plan Tenant Base (Alone) Total count 7

Tenant	Prefix	Dial Tone	Tenant Name
1	*1	Use	
2	*2	Use	
3	*3	Use	
4	*4	Use	
5	*5	Use	
6	*6	Use	
7	*7	Use	

( 01 )

[MenuLink](#)

[Overall Numbering Summary](#) [Numbering Plan](#)

Process Success!

Menu is divided in to multiple tabs

Tenant Attribute

Node 01 Tenant 1

Normal Information Charge Information

Tenant Name		Default Codec	G.711a
ACNR Retry Count	2	Full SMS Trunk Call	
Call Forward Chain Allow Step	3	Tenant for ATD Service	1
Call Forward Apply from Attendant Call	No	ATD Call To Executive	To Secretary
ATD Answer Priority	Recall->Trunk->Extension		
G.711 Packetization Time (msec)	20	G.723 Packetization Time (msec)	30
G.722 Packetization Time (msec)	20	G.729 Packetization Time (msec)	20
Wake-up Retry Count	2	Wake-up Retry Interval(min)	2
Tenant Name Display on LCD	No	LCD Back Light Usage in Idle	Day(O), Night(X), Timed(X)
Pick-up between ATD	No	Redial List Use	Not use (All Terminals)
Conference Member Add Method	Immediately On Answer	Call Forward, DND, Absence MSG Duplication Use	Use
Time Zone No. <a href="#">?</a>	1	Extension Message Wait Use	Not Use
DTMF Bypass on Transit Call	No	Extension Voice Message Wait Use	Not Use
Free Zone ARS Use	Not Use	Free Zone Use	Not Use
Trunk Incoming Call on Bandwidth Limitation	Call Release	Free Zone Call Use	Not Use
Terminal Lock for Unauthorized Access	Not Use	Privileged Feature Access Method	No

Process Success!

One window for day/night menus

Incoming Route Error Destination

Node 01 Route No. 1

	Day	Night	Timed
Busy	System Tone	System Tone	System Tone
No Answer	System Tone	System Tone	System Tone
Wrong Number	System Tone	System Tone	System Tone
Transfer no answer	System Tone	System Tone	System Tone
Recalling no answer	System Tone	System Tone	System Tone
DND	System Tone	System Tone	System Tone
Extension Lock-Out	System Tone	System Tone	System Tone
ETC	System Tone	System Tone	System Tone

Process Success!





**iPECS**  
Your Communications Solution

# New Application Integration

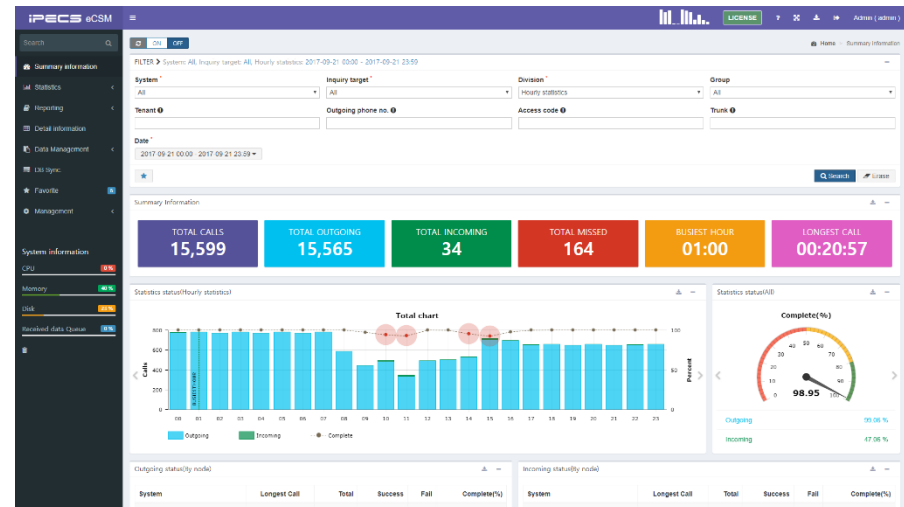
- eVQM (enhanced Voice Quality Monitoring) provides tools to monitor, troubleshoot, resolve, and prevent quality issues that negatively impact service levels and user experience.

## ■ Essential VQM data

- MOS Info
- Talk Delay
- Packet Loss
- Jitter
- Signal Noise, Level
- Configuration Data



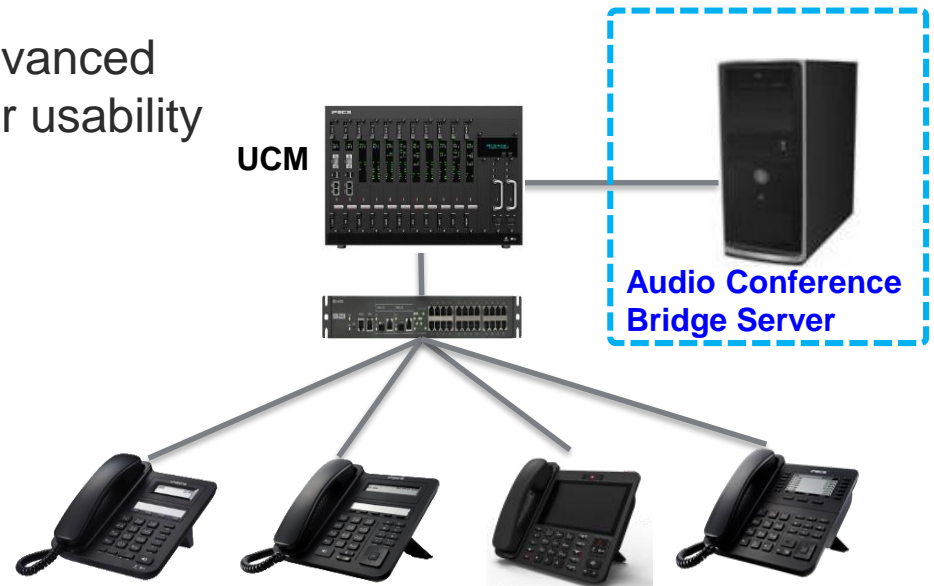
- eCSM (enhanced call statistics manager) provides tools to monitor, troubleshoot, resolve, and prevent call connection issues that negatively impact service levels and user experience.
- Comprehensive statistics data
  - Total/per-node/per-tenant/per-route-group/per-prefix/per-trunk)
  - Statistics for (monthly, daily, hourly, every 5-minutes)  
: outgoing/incoming/total/subscriber/subscriber-group
  - Report chart / Top 5 report/daily report/monthly report
  - Tenant, route-group, prefix, trunk name configuration
  - Statistics group configuration
  - Call tracking function
  - Excel, PDF, PNG file output
  - Daily summary report
  - User, group, privilege management



Category	Feature	Description
Mailbox Features	Maximum Message Length	Max 3600 seconds
	Days Messages Kept	1~365 days (default: 30days)
	Urgent Voice Message	priority : urgent or normal.
	Language Options	prompt language is selectable based on tenant.
Logins	Log-in	Mailbox number and Passcode will be requested to log-in.
	Fast Log-in	When an user access voice message from his office phone, only Passcode will be needed
	Password administration	option to set/change password
Greetings	Personal Greeting	Multiple personal greeting can be defined based on user-defined time schedule.
Voice Title	Personal Voice Title	personal voice title can be defined
Message Playback Options	Play Urgent Voice Message first	If there is any Urgent Voice messages, those message plays first.
	Play Message	automatically enters the subscriber into the Play menu if new messages are present.
	Play message header	Listen to header of message (date/time/length/calling party of message)
	Play Voice Title	Voice Title will be played before message play
	Call Back	call back to the number of the person who left you the message
	Reply a Message	after listening to a user message, user can reply with a voice message
	Forward a Message	Lets you forward a message to a phone number
	Skip a Message	Lets you skip the message is playing.
	Save Message	a message will remain in the mailbox for 31 days
Sending Messages	Create and Send Message	Lets you create a new voice message from scratch and send it to another number
Notifications	Message Waiting Indicator (MWI)	When a new message is present, the indicator light on the phone is illuminated.
	Email Notification	You can set the phone to send you an email whenever a new voice message arrives
User Account	VM User Account Creation	Create VM User account automatically by DB-Sync with CM
VM Web	Web Management	System / Tenant / User Admin Web

- Audio conference bridge provides advanced conference control interface for better usability and security :

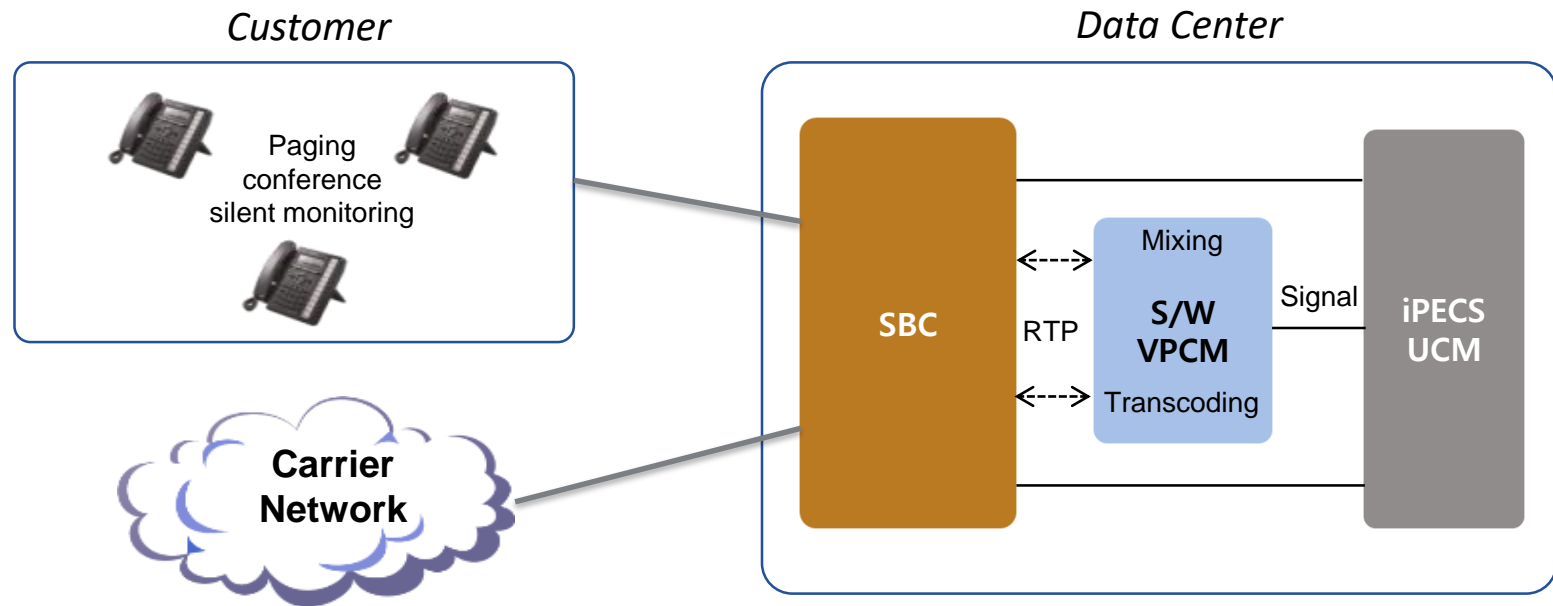
- › Dynamic Conference Creation
- › Authentication with Security Code
- › Participant Name Recording & Play
- › Mute All
- › Unmute All
- › End Conference
- › Conference Statistics
- › Web-based management\*



DIGIT	ADMIN USER	NON-ADMIN USER
1	Mute/Un-Mute Self	Mute/Un-Mute Self
2	Lock/Unlock Conference	Disabled
3	Eject last user who joined conference	Disabled
4	Decrease Listen Volume of Conference	Same for user
5	Resets the caller's listening volume to the default level.	Same for user
6	Increase Listen Volume of Conference	Same for user
7	Decrease Talk volume	Same for user
8	Leave Menu	Same for user
9	Increase Talk volume	Same for user
0	Allows an Admin to mute/unmute all non-admin participants in the conference	Disabled
*	Play menu options	Same for user
#	Leave Conference	Leave Conference

\* Web-based management feature will be added in 2018.1H





- S/W VPCM replaces existing VPCM for removing dedicated H/W on Data Center
  - Used for voice mixing (for add-hoc conference), paging, and silent monitoring
  - Transcoding for different codec between trunk and terminal
  - Tone generation for voice prompt and announcement



**iPECS**  
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# Application Integration

## ■ Enhanced IP Attendant

iPECS Attendant is a software application designed to enhance the attendant feature by visualizing the attendant call handling and control functions.

## ■ Embedded soft phone functions

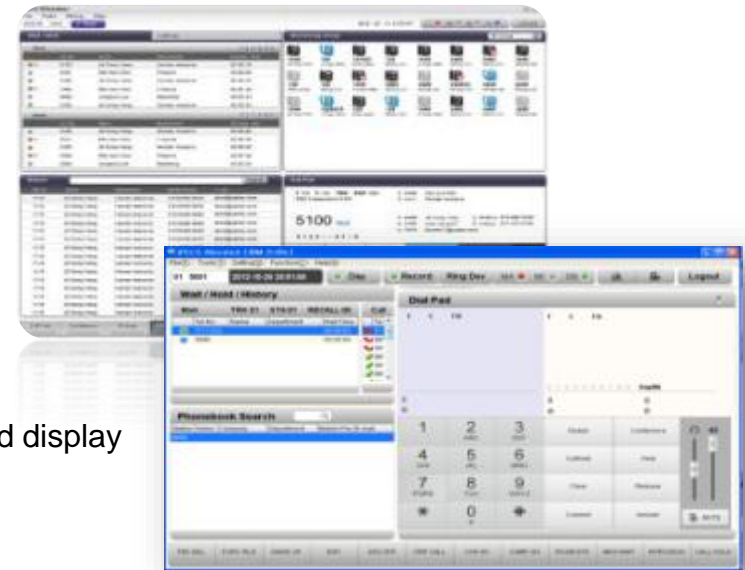
- Utilizing various call features of iPECS UCM via IP soft phone
- High quality voice communication using a PC
- Flexible call handling and operating
- Value added service such as video call, call recording

## ■ Easy to use interface

- New graphic user interface for easier and more intuitive design and display
- Flexible display layout

## ■ Hospitality features

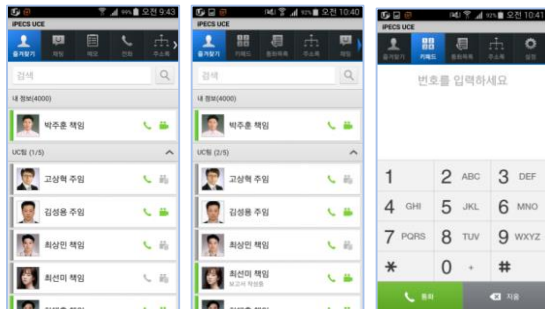
- New Check-in/out, wake up call, Group registration, room status, room cut off etc.



- iPECS UCE is specialized Unified Communication Solution for LME office environments.



**UCE Admin for mode selection**

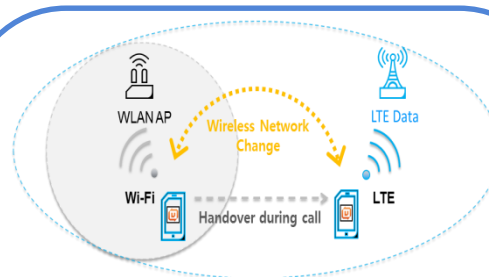


UC Client

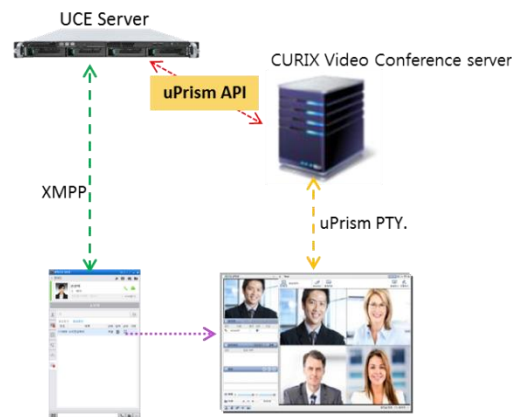
FMC Client

FMX Client

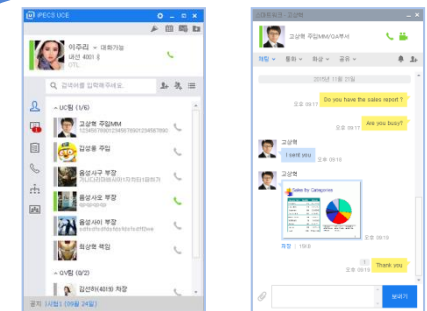
**UC/FMC/FMX Mode**



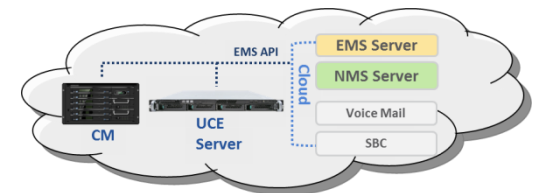
**Handover / LTE Only**



**Video Solution Integration**



**GUI/UI Enhancement**

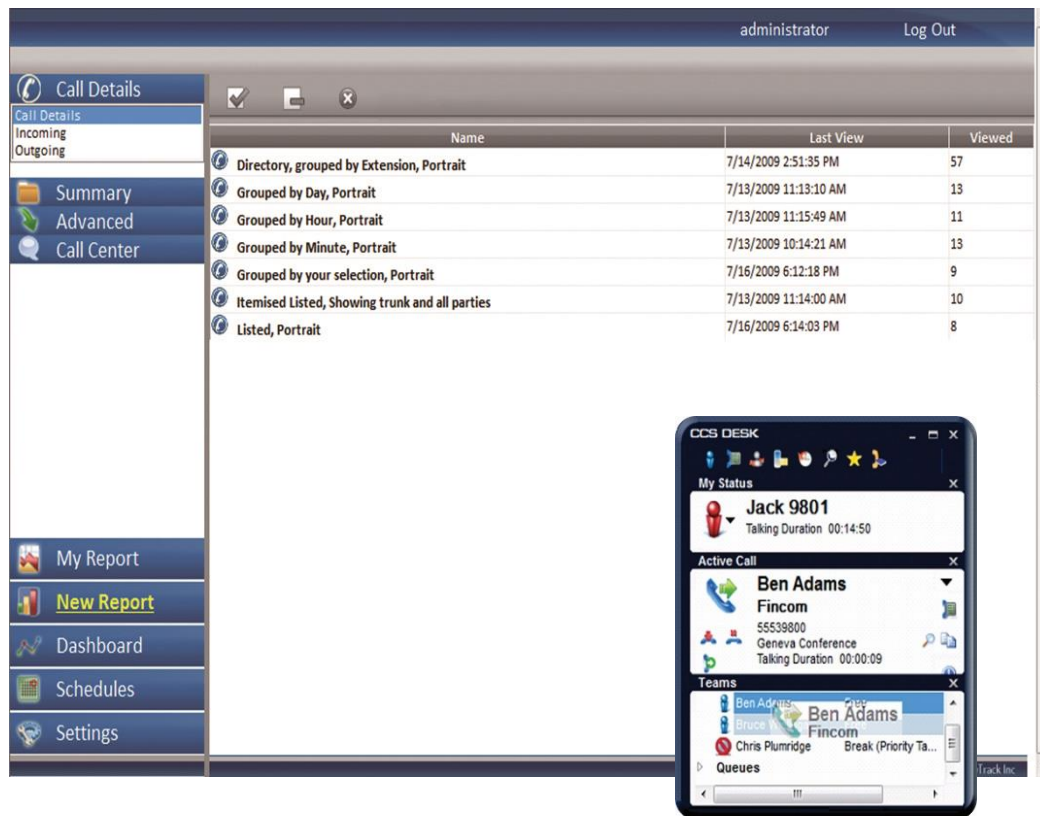


**Cloud Service Integration**



**XML Service for New IP-Phones**

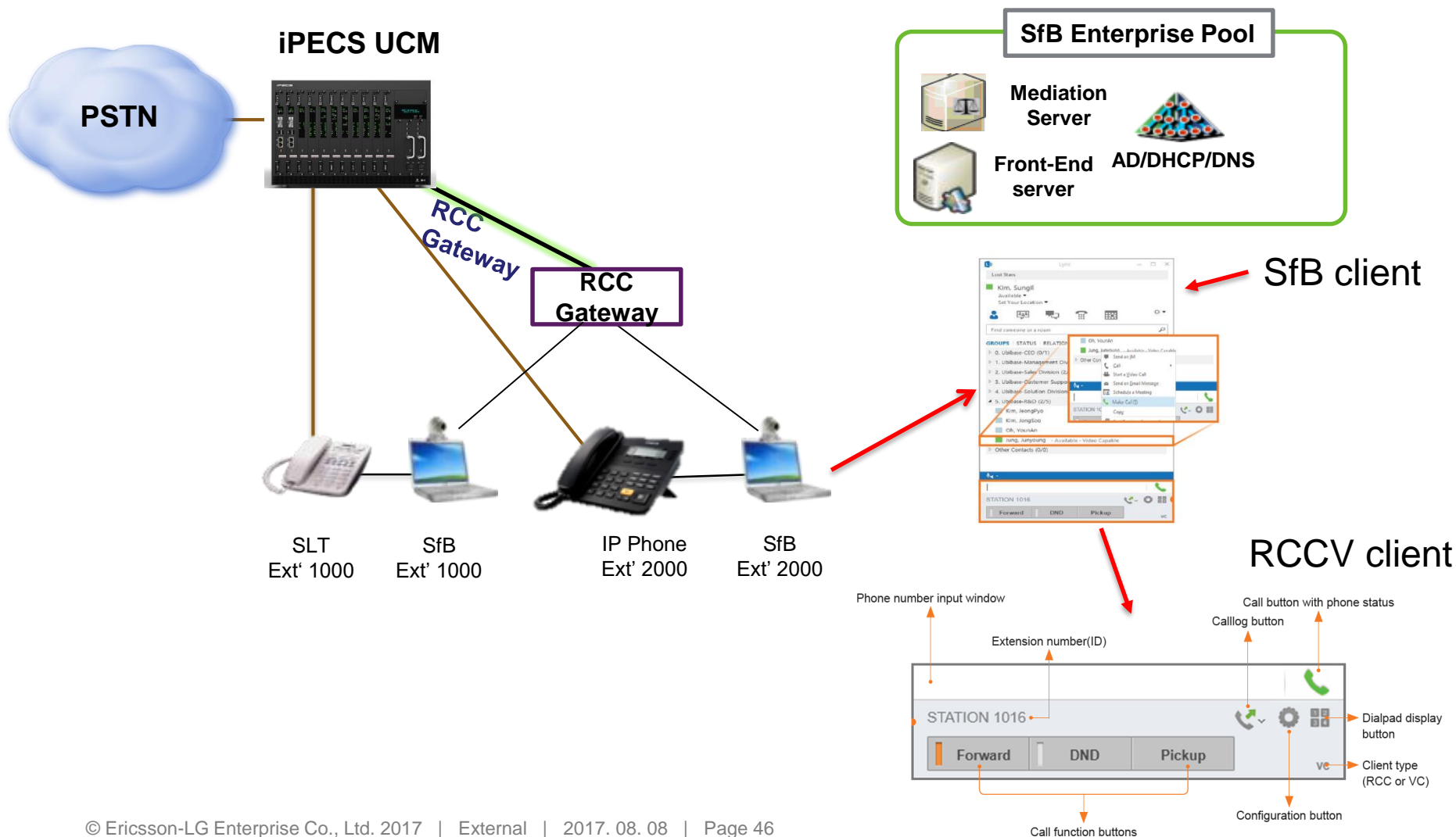
- iPECS CCS is the comprehensive Contact Center Suite optimized for small and medium contact center. It is designed as the solution with all IP (i.e., VoIP), multimedia (Call, Fax, E-Mail, Web), social networking (Facebook, Twitter) and modular base to fit to the variable customer needs.



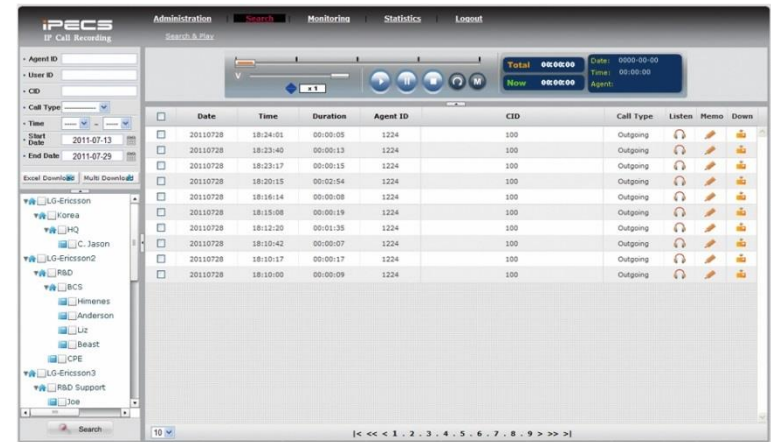
- Business Intelligence Reporting
- Skills Based Routing
- Real-Time Configurable Dashboards
- Unlimited queues
- Remote & Multi-Site Support
- CRM Integration
- Powerful Add-Ons
  - Call-back in Q (Virtual Hold)
  - Web Call-back
  - Email, Voicemail & Fax queuing
  - Voice Recording
  - IVR
  - Tele-Marketing (Outbound CC)
  - Web Chat

# iPECS RCC/RCCV

- iPECS RCC/RCCV Client can complement Microsoft Skype for Business client by providing IP phone control or soft-phone add-on



- iPECS IPCR is IP Call Recording solution optimized for iPECS Call Platform
- **Cost effective all call recording**
  - No additional circuit I/F board required
  - No extra cabling required
  - Centralized or distributed call recording
  - Encryption enabled call recording
  - All Terminal recording
  - Trunk-port-based call recording
  - Multi party Conference call recording up to 128 party access
- **Supervisor Call recording control**
  - Easily remove recording file when customer doesn't want to record
  - Supervisor/Manager can stop agent All Call Recording
  - Supervisor/Manager can start one click call recording during On-Demand Recording



# Fidelio PMS I/F for Hotel

- FIAS (Fidelio Interface and Application Protocol) is implemented to support direct interface with Fidelio PMS which is one of the most popular PMS for large hotels.

- **Supported Features**

- Check in / check out notification
- Room move
- Guest information support for name and language
- Class of service handling (bar / unbar an extension)
- Room status
- Voice mail notification from iPECS
- Message lamp handling
- Do not disturb handling
- Simple posting of phone charges based on total amount
- Minibar charge by total and by article number
- Wake up calls handling

- **Compatible MICROS-Fidelio PMS versions**

- Suite8 PMS version 8.6.x onwards
- OPERA Suite PMS version 4.0.04.x onwards
- OPERA Suite PMS version 5.0.x



Front Desk Station

00 : GUEST NAME  
01 : GUEST GRADE  
02 : CHECK-OUT DATE  
03 : LCD LANGUAGE  
04 : HOTEL ROOM CUT-OFF  
05 : DND  
06 : WAKE UP TIME  
07 : TOLL CLASS  
08 : TRANS CLASS  
09 : PMS GROUP  
10 : MSG WAIT NO  
11 : MSG SENDER



# XML Service

- LIP-9000 series provide XML API to allow customer to combine communication & Information in order to improve productivity.



## Major Features

- Multimedia CID
- Directory Search
- Corporate Notification
- Call History
- Catch-Call



Directory Search

# WMS (Web Management System)

- WMS is designed with very intuitive user interface, which helps administrator to install and maintain iPECS UCM easily.

- **Editing simultaneously**

- Multi-login, Multi-working, Log-in History
- Multi Windows I/O
- Tree Structure

- **Monitoring for real time defect**

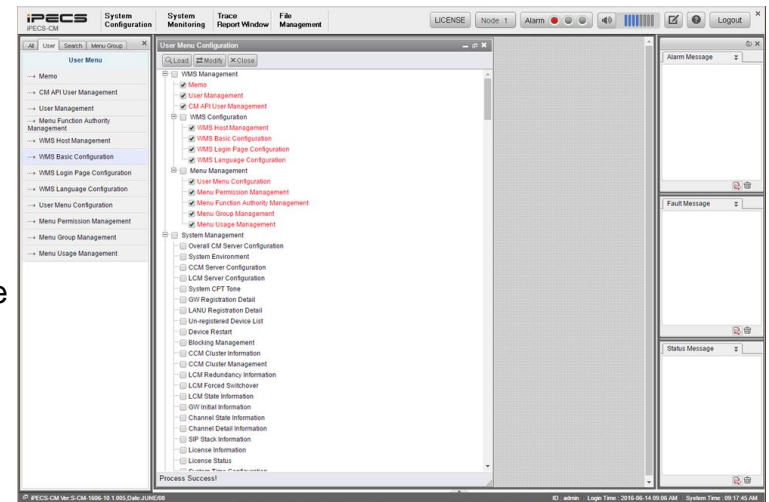
- Real-time Display
- SMDA, 3<sup>rd</sup> Party App., CPU, Memory, Hard Disk Usage etc

- **Print Real-time Message**

- Alarm, Fault, Notification System Fault using Led, Speaker

- **Remote Maintenance**

- Providing remote system control, Fault Monitoring, Statistics
- Securing certification process about unknown users



## ► Web Based Local & Remote Maintenance Solution

**Call Trace regarding specialized extension or trunk**  
**Message Trace regarding Status, RTP Connection etc**

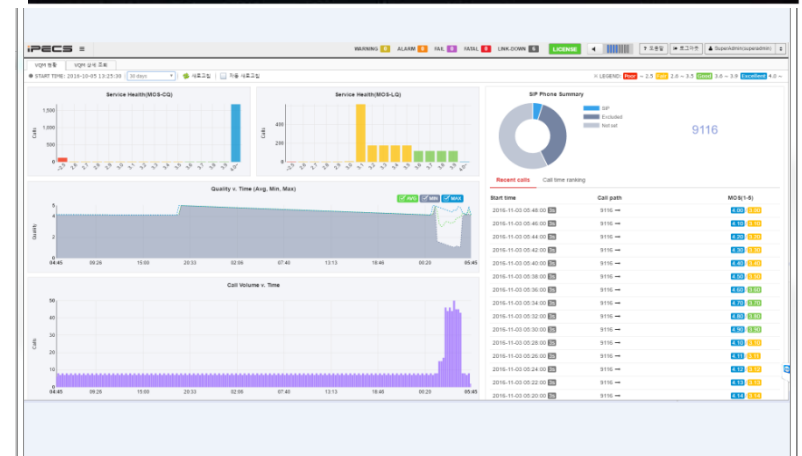
- eNMS (Network Management System) is a powerful tool for managing a fault information, monitoring real time status, maintaining call statistics and databases of multiple appliances.

## SNMP based unified monitoring

- Call server and gateway module monitoring
- Terminal data management  
DB sync. with iPECS UCM  
Managing terminal status, IP address and etc
- Data switch status monitoring
- Customized Dash Board

## Error monitoring and real time notification

- SMS/E-mail notification
- Management of trouble history





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# Terminal Integration

# LIP-9000 Series

## LIP-9000 Series

- Full line-up in the new design
- Intuitive user interface in high resolution LCD
- Enriched presence information display
- Optional color variation available
- HD voice quality for all models



**Premium**  
**LIP-9071 SIP/IPKTS**



**LIP-9040C SIP/IPKTS**  
**LIP-9040 SIP/IPKTS**

**Advanced**



**LIP-9030 SIP/IPKTS**

**Standard**



**LIP-9020 SIP/IPKTS**

**Basic**



**LIP-9010 SIP/IPKTS**



**LIP-9008 SIP/IPKTS**

**Entry**



**LIP-9002 SIP/IPKTS**

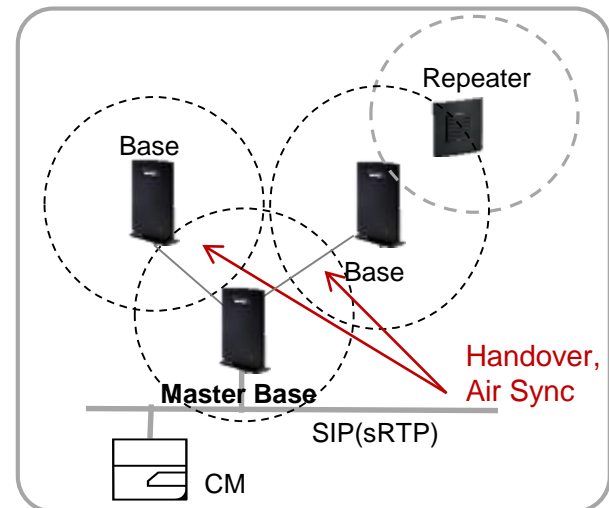


**LIP-9012DSS**

**LIP-9024DSS**

**LIP-9024LSS**

- **Standard SIP based IP DECT for cost effective mobility and easy maintenance as well as enhanced features**
- **High-tech IP DECT solution**
  - 128 BS & 1000 HS
  - Up to 8 calls / BS
  - SUOTA (SW Upgrade Over-the-Air)
- **Cost effective multi-site mobility solution**
  - Multiple locations become integral parts of centralized voice communication infrastructure.
  - IP DECT at remote locations does not need additional remote gateway module
- **Easy Installation & Management**
  - Centralized Web management tool
  - Simplified installation and operation
- **Simple to expand coverage using repeaters**
  - Up to 1 repeater per base





# Basic Features

# Basic Feature Groups

## Call Control

1:1/ 1:N / Hold / DND / Redial  
Speed Dial  
Forward / Transfer  
Call Park / Pick-up  
Class of Service  
Emergency Call  
Multi-Numbering  
CID Conversion  
Virtual Subscriber  
ARS Digit Conversion  
Executive/ Secretary

## Numbering

Directory Number  
Numbering Plan

## Station Group

Key number  
Hunt  
Pick up  
PTT  
Interphone  
Paging

## DISA

Auto Attendant  
Voice Prompt

## Hot Desk

Virtual office

## Networking

SIP  
QSIG

## Mobile Extension

Extension Paring with Mobile  
Call Through / Call Back  
VCC

## Conference

Ad Hoc  
Meet me  
Command Call

## Voice Mail

Email Notification  
Greeting  
Voicemail Backup  
VM Paging

## System Security

VPN, Black list, Call SPAM, ACL, Auto QoS

## Hotel Feature

Wake up, Emergency Call Notification, Hunt Group Display

## Cloud Service

Licensing Structure based on tenant

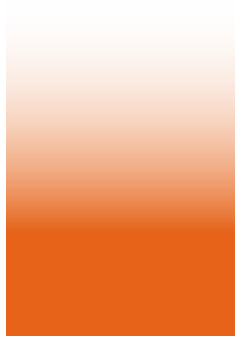
## System Operation & Management Tool

WMS  
NMS  
Statistic



# Basic Feature List

Call Processing	Call Processing	Call Processing	Trunk
Call Hold	Trunk Queuing	Auto Attendant / Digital Receptionist	Trunk Release Guard Time
System Hold	Step Call	Web-based Management Console	Trunk Ring Detect
Automatic Hold	Voicemail	Real Time Web-based System Status	Dial Pulse Signaling
Music on Hold	Record New Voice Prompts From Phone	Integrated Web Server	DID (direct in dial)
Music on Hold Playlist or Line In	Call Recording	Backup and Restore	DISA (direct inward system access)
Do Not Disturb	IP call recording	Vmware compatible	DTMF tone control
Call Logging	MWI – Message Waiting Indicator	Temporary Station Lock	IP trans-coding & rtp relay
Call Reporting	Message wait/call back	Security	IP trunk
Call Transfer	BLA (Bridge Line Appearance)	TLS/sRTP	H.323 Trunk
Transfer Directly to VM	Class of Service	Terminal lock for unauthorized user	ISDN
Blind Call Transfer	Dial Plan	Localization	CLI Display
Attended Call Transfer	Emergency Calls from Unnamed IP phone	Multiple language support	ISDN supplementary services
Call Forward	Fax over IP	Speed dial	ISDN Call Deflection
Pilot hunt call forward	Hot Line Service	Speed Dial Pause Insertion	ISDN Malicious Call Id
Preset call forward	Automatic Answer	Station Speed Dial	ICLID ROUTING
Call Forward on Busy	Last Number Dialed	System Speed Dial	Automatic Network Dialing
Call Forward on No Answer	Line Lockout	Tenant	Alternative Route Selection
Call Routing (DID)	call override	Tenant prefix	Individual Trunk Access
Call Routing (Caller ID)	Short Message Service : SMS	Tenant group	Route Selection For Each Trunk
Alternative Route Selection (ARS)	Message Wait Reminder Tone	Command call	Trunk Route Service
Auto Called Number Redial (ANCR)	BLF Status Updates	Command call	Incoming Route Option
Caller ID	Intercom call	cost control	Outgoing Route Option
CID dependant ring	Intercom call hold	Account code	Alternative Incoming Route Service
Trunk IC/OG CID change	Intercom call answer mode change	Authorization code	Alternative Outgoing Route Service
Conference Calling	Intercom step call	CDR/SMDR	Route Transit Service
Conference Rooms	Paging	LCR: least cost routing	Trunk Access Code Service
Multi-Party Voice Conference	Internal Page	data control	Trunk Own Code Service
Call Intrusion	Meet Me Page Answer	Data line security	Route Cli Service
Call Intercept	Push-to-talk paging	Fax bridge	Trunk Direct Transit Service
Call Parking	Ring Extension and Mobile Simultaneously	Trunk	H.323 Multi Route Service
Call Pickup	Executive/secretary	CO line flash	Trunk Inter Digit Timer Service
Call Back	Executive/secretary forward	Trunk Route Groups	Trunk Dtmf Duration Service
Call Waiting	Mobile extension	Trunk Preset Forward	R2 Comfort
CampOn	Attendant	Trunk Ring Assignment : Dil	CAC : Call Admission Control



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# New Special Features

# VQM support for G/W & IPKTS phone

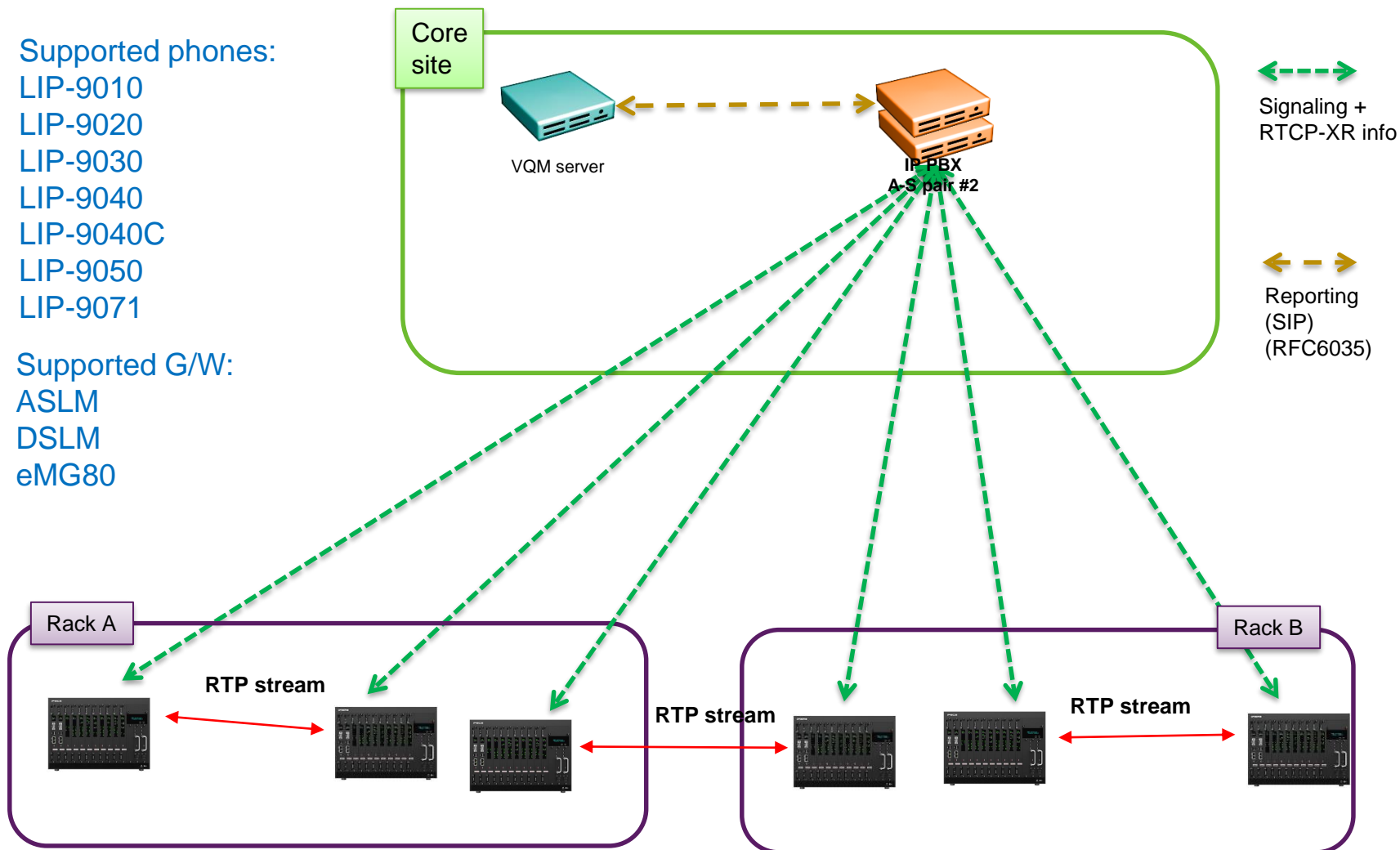
- Voice quality of gateways and IPKTS phone can be measured

Supported phones:

LIP-9010  
LIP-9020  
LIP-9030  
LIP-9040  
LIP-9040C  
LIP-9050  
LIP-9071

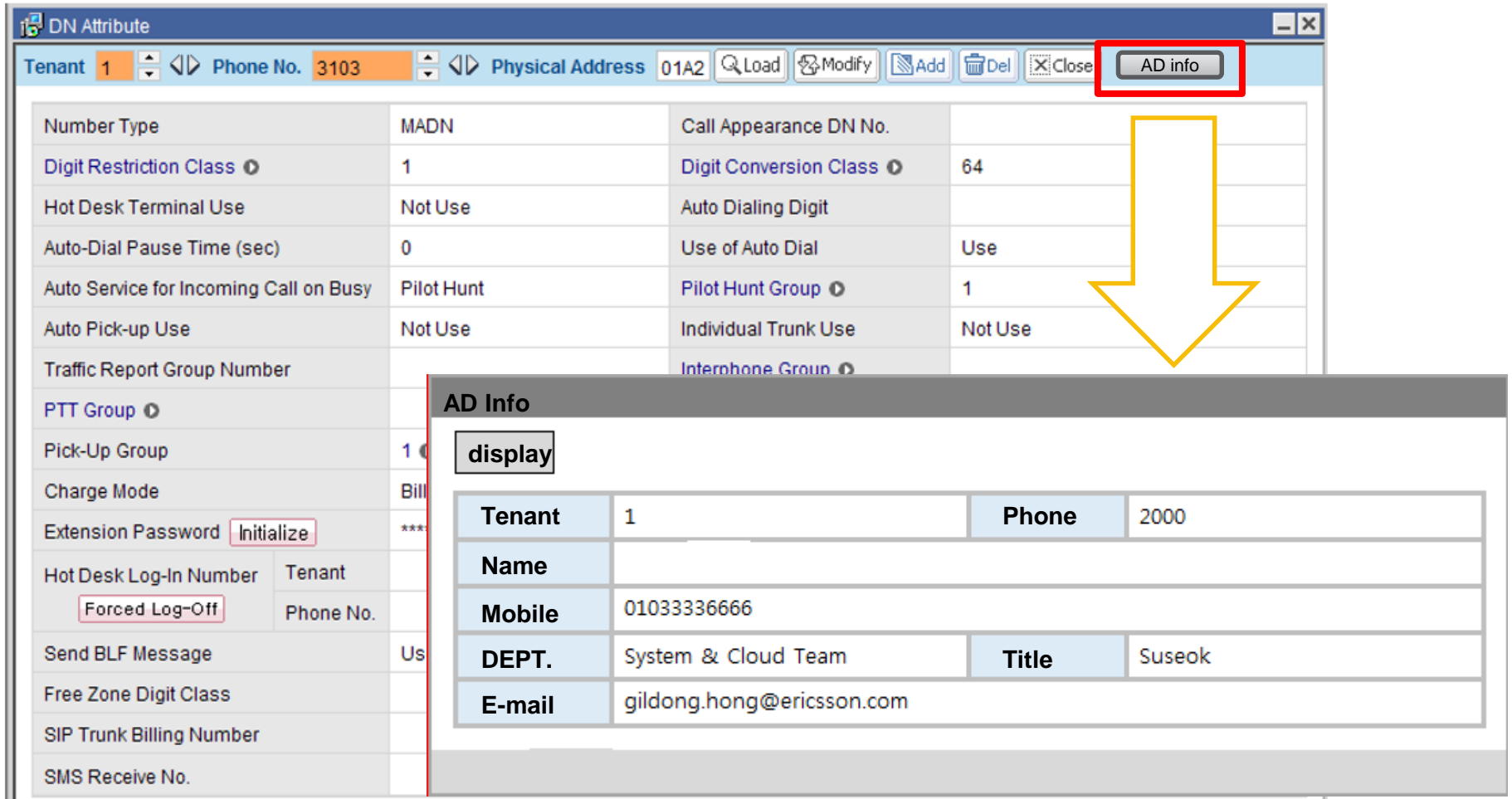
Supported G/W:

ASLM  
DSLW  
eMG80



# User Info Field Upgrade

- More info about extension user is provided (lookup API also supported)



The screenshot shows the 'DN Attribute' configuration window. The 'AD info' button is highlighted with a red box. A yellow arrow points from this button to the 'AD Info' pop-up window. The 'AD Info' window displays a 'display' button and a table with user information.

AD Info			
<b>display</b>			
<b>Tenant</b>	1	<b>Phone</b>	2000
<b>Name</b>			
<b>Mobile</b>	01033336666		
<b>DEPT.</b>	System & Cloud Team	<b>Title</b>	Suseok
<b>E-mail</b>	gildong.hong@ericsson.com		



# Special Features

# Hot Desk

- Hot Desk feature provides seamless communication at anywhere, anytime

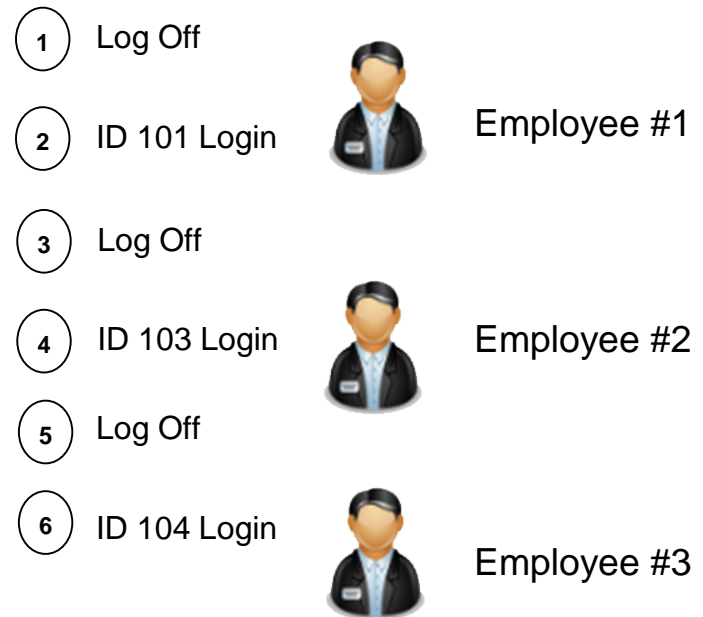
- Hot Desk



Anytime



Anyone



Anyplace

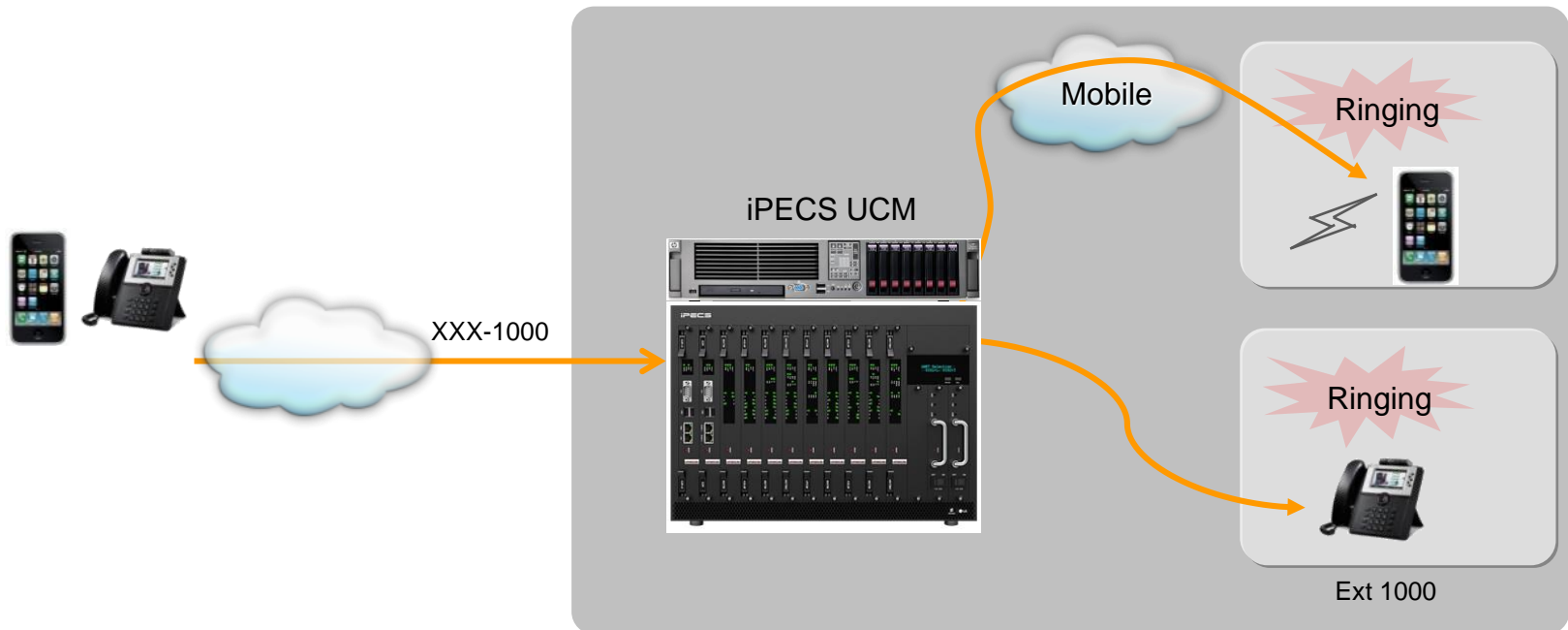


Seat #1    Seat #2    Seat #3

\*Flexible button map and station speed dial information of deskphone is changed to match Hot Desk ID

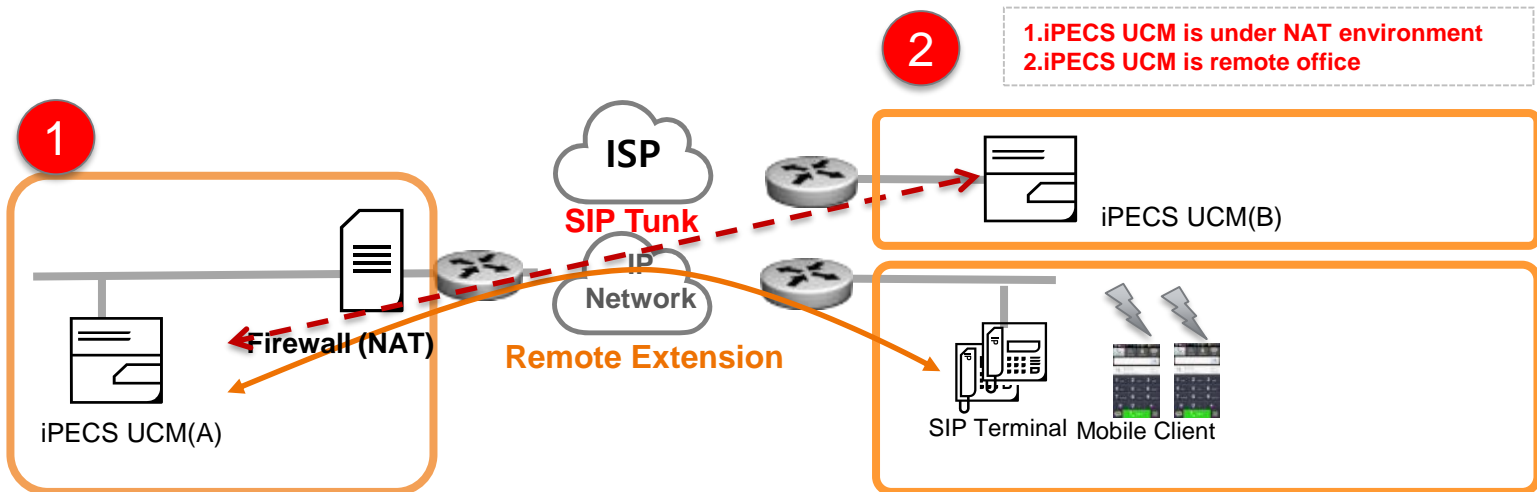
# Mobile pairing

- Paired mobile phone provides one-number service for office worker



# Remote Extension Enhancement

- SIP Trunk & Extension is supported in enterprise network environment to increase system availability
  - SIP extension to IP phone, Mobile Client out of NAT/Firewall is now available
  - NAT feature on SBC is not needed

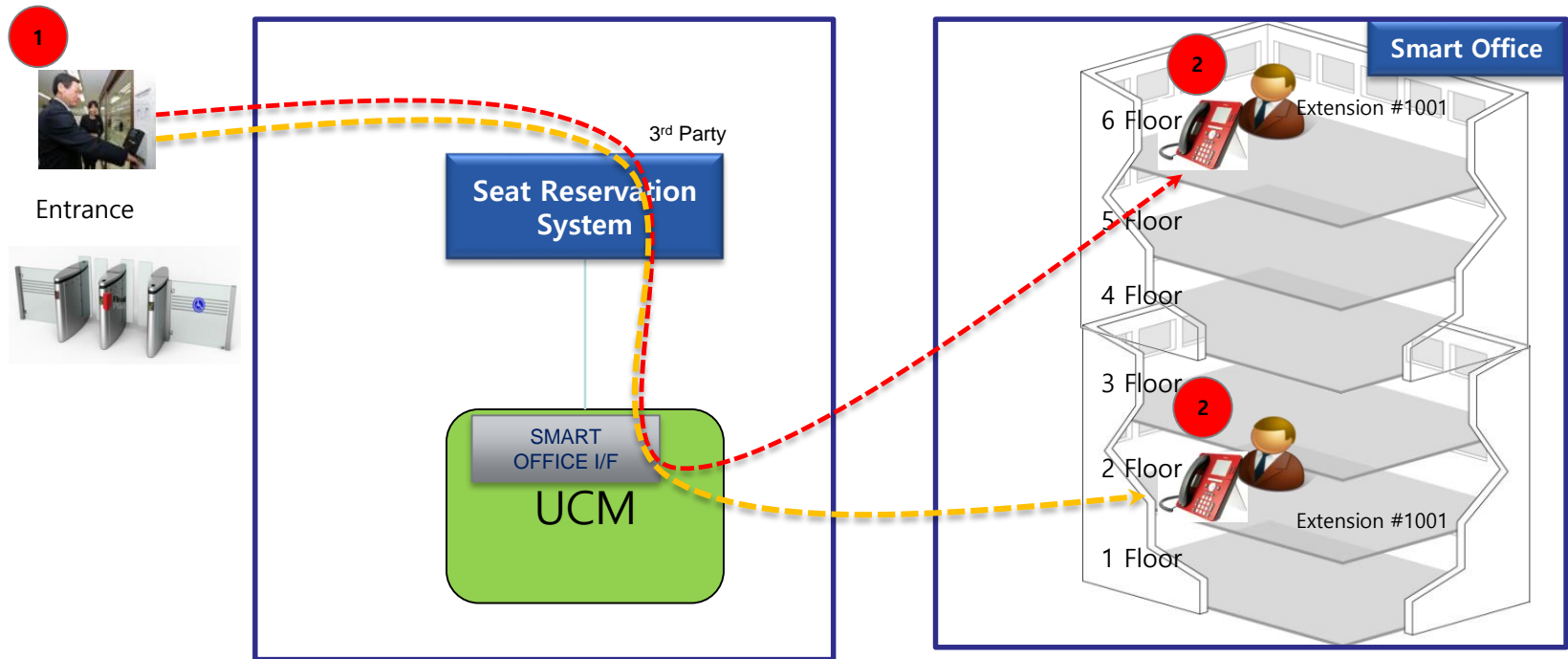


- iPECS UCM can set firewall IP for SIP registration and call control of remote phone.
- Proxy Server IP should be set to firewall IP of SIP Phone.



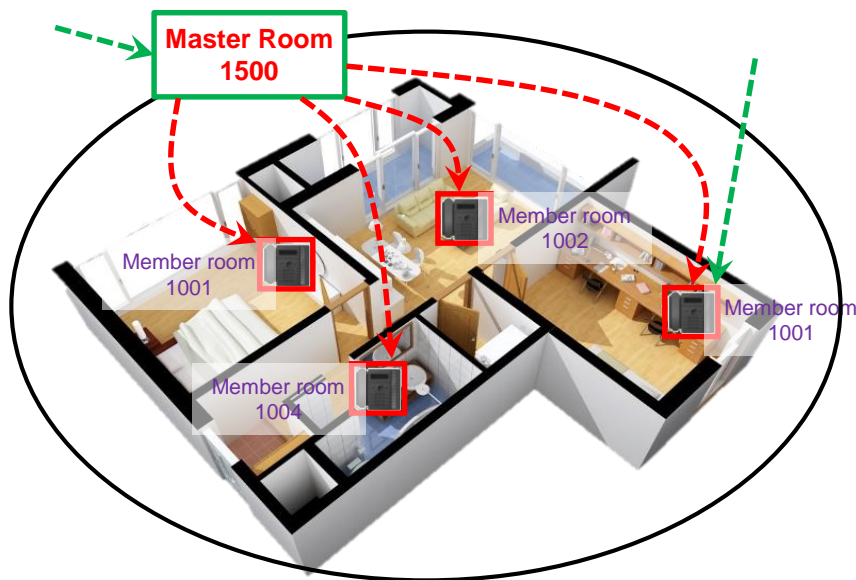
# Smart Office

- Utilize Smart office I/F in Smart Mobility Office environment



1. Register your seat
2. Automatically deskphone on the seat you reserved will be changed to your extension.
3. UCML-SMO will be needed soon but not needed now

# Suite Room



Index	Tenant	Phone No.	Ring Service	Wake Up Service
1	1	1001	Immediate Incoming	Immediate Incoming
2	1	1002	Immediate Incoming	Immediate Incoming
3	1	1003	Immediate Incoming	Immediate Incoming

WMS > Data Management > Hotel Information > Suite Room Information

1. Master Room Number of Suite Room
2. Up to 60 Member Room Number per Suite Room
3. Ring option for Normal Call and Wakeup Call can be defined:  
Immediate Incoming, 3~30 sec Delayed, No Incoming

- Suite Room is consist of several Member Room numbers and one virtual Master Room number which is used for controlling room status and interworking with PMS, Hotel Front-Desk or Hotel Attendant.
- Incoming call only with either Master Room Number and Member Room Number.
- Outgoing call with any phone but with Master Room Number including its COS, CID, Name, Check In/Out information.
- Up to 4 concurrent calls can be made per Suite Room.
- SIP phone is not allowed to be member of Suite Room

# IP Attendant Headset Link

Call Control



Voice



Voice



- Call Initiation by IP ATD, not the phone
- Some of feature will not be supported such as Call Recording by IP ATD, Voice Conference and so on
- With SLT or LIP Phone

Soft Phone Link

Load Add Del Close

Slot No. 3 CH No. 21 Tenant No. 1 Phone No. 5001 Physical Address 6B

Terminal Main Type	ATD		
Terminal Sub Type	iPECS-ATD		
Hard Phone	Tenant	1	
	Phone No.	70000002	

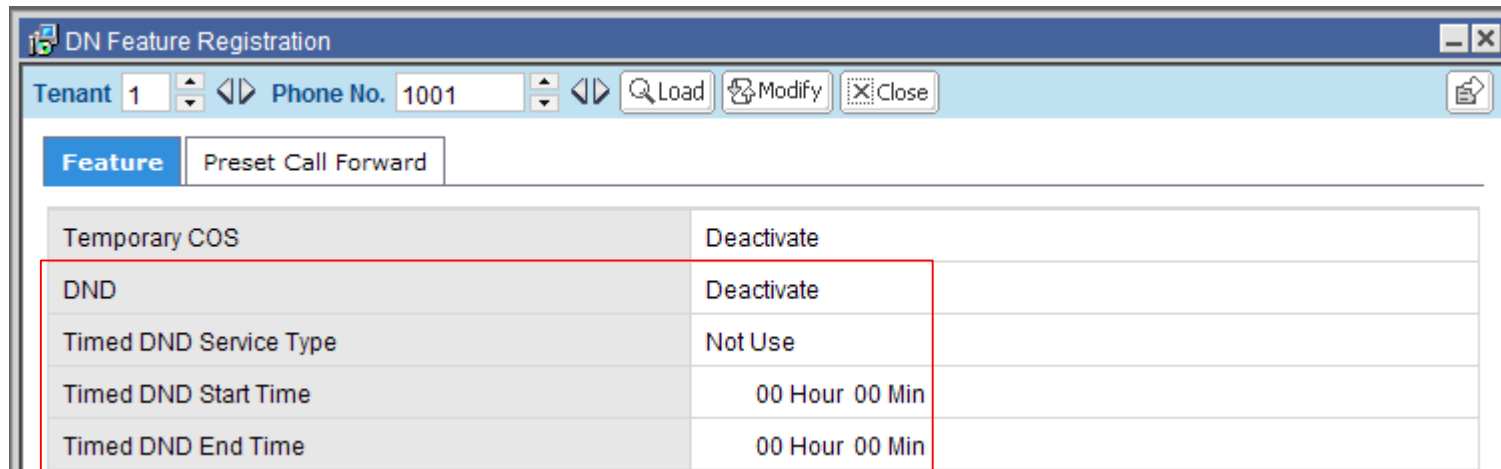
Menu Link

Soft Phone Link Summary

Process Success!

# Timed Do Not Disturb (DND)

- This feature allows users to set DND for range of designated time.
  - Five (5) DND options are supported: Once, Daily, Monday~Friday, Monday~Saturday, or a Date.
  - DND registration for Daily, Monday~Friday or Monday~Saturday is maintained until it canceled by the user.
  - DND registration for Once or Date mode is canceled automatically after end time.



The screenshot shows a web-based interface titled "DN Feature Registration". At the top, there are fields for "Tenant" (set to 1) and "Phone No." (set to 1001), along with buttons for "Load", "Modify", and "Close". Below this, there are two tabs: "Feature" and "Preset Call Forward". The "Feature" tab is active, displaying a table with the following settings:

Temporary COS	Deactivate
DND	Deactivate
Timed DND Service Type	Not Use
Timed DND Start Time	00 Hour 00 Min
Timed DND End Time	00 Hour 00 Min

A red rectangular box highlights the "DND", "Timed DND Service Type", "Timed DND Start Time", and "Timed DND End Time" rows in the table.

# Catch Call Service

- UCM can inform you of the incoming call information when the status of user is 'Busy' or 'DND' or 'Forward' or the incoming call number is registered to 'Spam'.

The screenshot shows the 'Terminal Option' configuration window. At the top, there are fields for Slot No. (2), CH No. (1), Tenant No. (1), Phone No. (4000), and Physical Address (20). Below these are various configuration options. The 'Catch Call Log Save Option' is highlighted with a red box and is set to 'Busy'. Other options include Data Line Type (Voice Line), Message Wait Indication Method (Not Used), Differential Ring Program (Apply All(Normal, Recall, Forward, Transfer)), Differential Ring ID for Internal Call (1), Differential Ring ID for External Call (1), SLT Dialing Type (DTMF Only), Digit Restriction Class for S-DN (Sub Number), Hook Action During Transfer (Transfer cancel), Off-Hook under Paging (Page Listen), Flexible Button Pick-up Action (Pick-up(Group)), Automatic Button Label Display (LIP-8040LD) (Use), Apply P-DN Function Registration to SADN (Sub Number), Lockout / Speaker Mode Release Type (Lockout manual, Speaker mode auto), Incoming Call Indication on Conversation (Mute Ring/Ring LED (Except LIP8004)), Internal Call Timer Display (Not Use), LIP8050V Default Media (Audio), Pick-Up Call Log (Not Use), AOC Metering (Display Cost (Normal:Increase, Prepaid:Decrease)), and Group Speed Dial Add Authority (Not Use). The status bar at the bottom indicates 'Process Success!'.

Option	Value
Slot No.	2
CH No.	1
Tenant No.	1
Phone No.	4000
Physical Address	20
Data Line Type	Voice Line
Message Wait Indication Method.	Not Used
Differential Ring Program	Apply All(Normal, Recall, Forward, Transfer)
Differential Ring ID for Internal Call	1
Differential Ring ID for External Call	1
SLT Dialing Type	DTMF Only
Digit Restriction Class for S-DN	Sub Number
Hook Action During Transfer	Transfer cancel
Off-Hook under Paging	Page Listen
Flexible Button Pick-up Action	Pick-up(Group)
Automatic Button Label Display (LIP-8040LD)	Use
Apply P-DN Function Registration to SADN	Sub Number
Lockout / Speaker Mode Release Type	Lockout manual, Speaker mode auto
Incoming Call Indication on Conversation	Mute Ring/Ring LED (Except LIP8004)
Internal Call Timer Display	Not Use
LIP8050V Default Media	Audio
Catch Call Log Save Option	Busy
Pick-Up Call Log	Not Use
AOC Metering	Display Cost (Normal:Increase, Prepaid:Decrease)
Group Speed Dial Add Authority	Not Use

Process Success!

# Call Recording using IP Phone Conference

- To decrease required number of VPCM, SIP phone which activates call recording performs voice mixing internally and send mixed voice packet directly to the call recording device.

## Related WMS Menu

[To set use of Call Recording in System]

Data Management > Number (DN) Information > Voice Mail Information > Two-way Record Device

Data Management > Number (DN) Information > Voice Mail Information > Two-way Record Start Mode

Version Management > SIP Phone Provisioning > SIP Phone Type Configuration > Call Recording Use : Use

The screenshot shows the 'SIP Phone Type Configuration' window for an '8830E Phone'. The 'Call Recording Use' option is highlighted with a red rectangle and is set to 'Use'. Below the configuration table, a 'Notice' section provides additional information about upgrading the SIP phone and digit maps.

SIP Phone Type Configuration			
Phone Type: 8830E Phone			
Mention Select		Mention Download	Not Use
Function Sync	Use	DTMF Mode	Outband(Info)
XML Use	Not Use	XML Server URL	
Phone Book URL			
User Mode Use	Not Use	User Mode Password	
Admin Mode Password		XML Phone Book Use	Not Use
TLS Certification Mode	RSA	Backlight Option	Backlight Off
Flexible Button Upload	Not Use	Flexible Button Upload URL	
Call Recording Use	Use		
Voice Mail URL			

**Notice**

You must upgrade SIP phone for changes to take effect

Digit Map (Routing) : 9010xxxxxxxx[901[1-9][2-8]xxxxxx[901[1-9]9xxxxxx[94xxx4xxx3xxx

Digit Map (2nd Dial Tone) : 9

If Predialing is not used, speaker button will be turned on automatically with on-hook dialing.

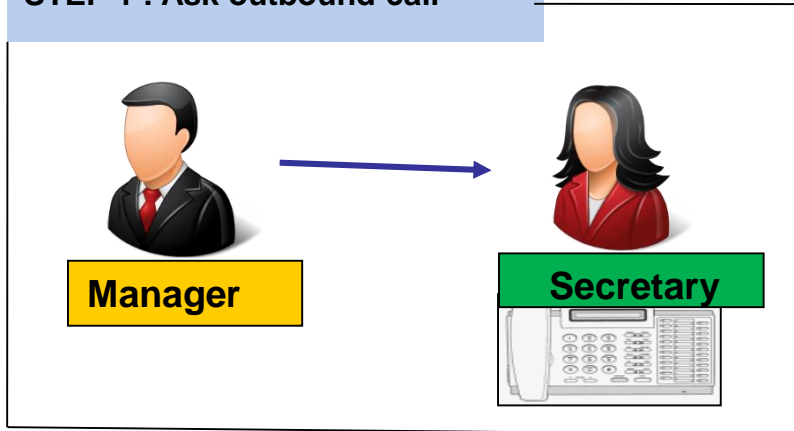
Pause Timer means Inter Digit Timer.

Process Success!

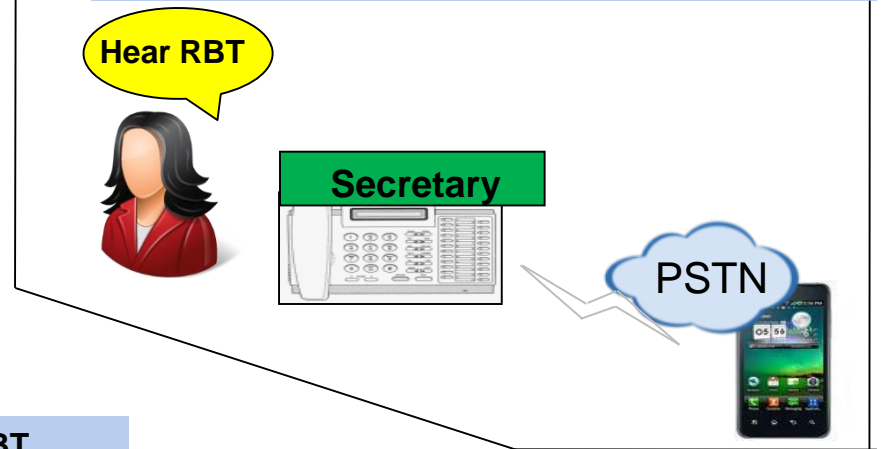
# Transferring a trunk call in alerting status

- Secretary can transfer a outgoing trunk call to Manager before called party answer the call.

## STEP 1 : Ask outbound call



## STEP 2 : Outbound call Hear RBT & press transfer button



## STEP 3 : Manager hear RBT



# Call Priority Control

- **Minimize Call (Min. Communication Level)**

Level of Extension for Minimal Communication. When this level of Extension is higher than the 'Min Communication Level' assigned in the 'System Environment' menu, the Extension cannot place an outgoing call.

- **Intercept Call**

When a called Extension is busy, the calling Extension can connect to the called Extension forcing the release of the connected party. The caller must have an Intercept level that is higher than the called party to activate Call Intercept.

- **Transfer COS (Call Transfer Restriction)**

Extensions may be configured so that they may not transfer a call to other Extensions. When pressing the [TRANS] button, the system may restrict the transfer based on the dialed digits compared to the Digit Restriction Map in the Transfer COS of the Extension.

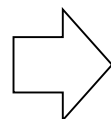
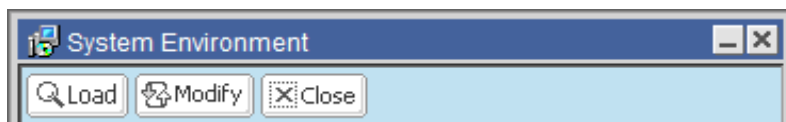
- **Hot/Warm Line**

An Extension can be assigned to access a pre-assigned number (Hot Line/Warm Line) automatically upon an off-hook state. Warm Line may be configured, in which case, the Extension user receives normal intercom dial tone for the Hot Line delay time and may dial based on the assigned dialing restrictions. When the delay timer expires, if the user has taken no action, the Hot Line call is placed.



# Minimize Call

- System can stop making a call from a lower priority station so that it can give high priority stations to guarantee to make a call.



Min. Communication Level

Terminal Attribute

Load Modify Add Del Close Extension Move Extension Exchange ( IP Phone Information )

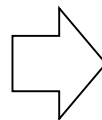
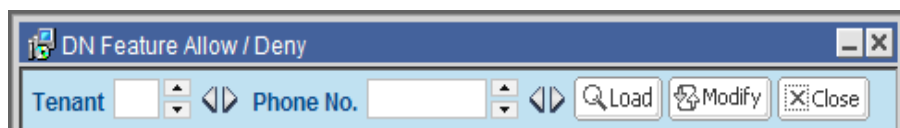
SLOT No. 121 CH No. 4 Tenant No. 1 Phone No. 1004 Physical Address 63

Min. Communication Level

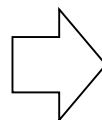
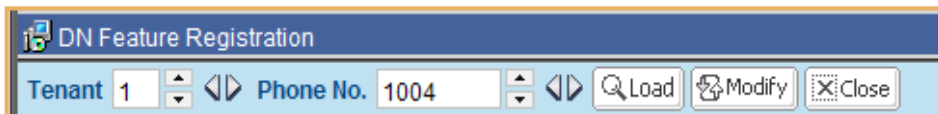
Tenant System Tone Information					
Tenant No. 1 Load Modify Close					
IDX	IndexName	Tone Type	Tone Duration (100 msec)	Tone Port	Prompt No.
61	IC Auto Hold Tone (ATD)	Normal	300	14	
62	Minimize Call Level	Prompt	300	14	4

# Intercept Call

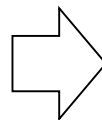
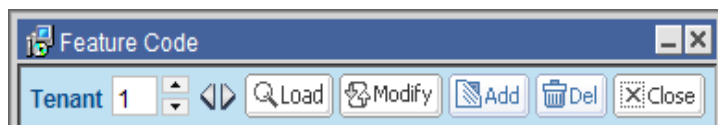
- Pressing “Intercept Request” feature code and extension number will remove the extension user and let the supervisor intercept the call.



Intercept	Allow
-----------	-------



Intercept Level	2
-----------------	---



<input type="checkbox"/>	37	*139	[139] Intercept Request
--------------------------	----	------	-------------------------

# Transfer COS

- A transfer call can be blocked by “TRANSFER COS” settings (prevents call delegating)

The screenshot illustrates the configuration steps to block a transfer call in the iPECS system. The process involves setting the Digit Restriction Class to 1, the Transfer COS to 2, and the Digit Restriction Map to 3, which then blocks the dialed digits 4201.

**Transfer COS Authority**

Use

**DN Feature Allow / Deny**

Tenant 1 Phone No. 1004

**DN Attribute**

Tenant 1 Phone No. 1004

Attribute	Value
Number Type	SADN
Digit Restriction Class	1
Hot Desk	Not Use
Auto-Dial Pause Time (sec)	0
Auto Service For I / C Call on busy	Busy
Auto Pick-Up Usage	Not Use
Traffic Group Number	

**Digit Restriction Map**

Tenant 1

COS	App. Type	Time Band	Change	Deny	Allow
1	All time	All	Edit		
2	All time	All	Edit	3	

**Digit Restriction Digit**

Group Index 3

Name

Number of Registered Digits : 1

Index	Digits
1	4201

**Transfer COS**

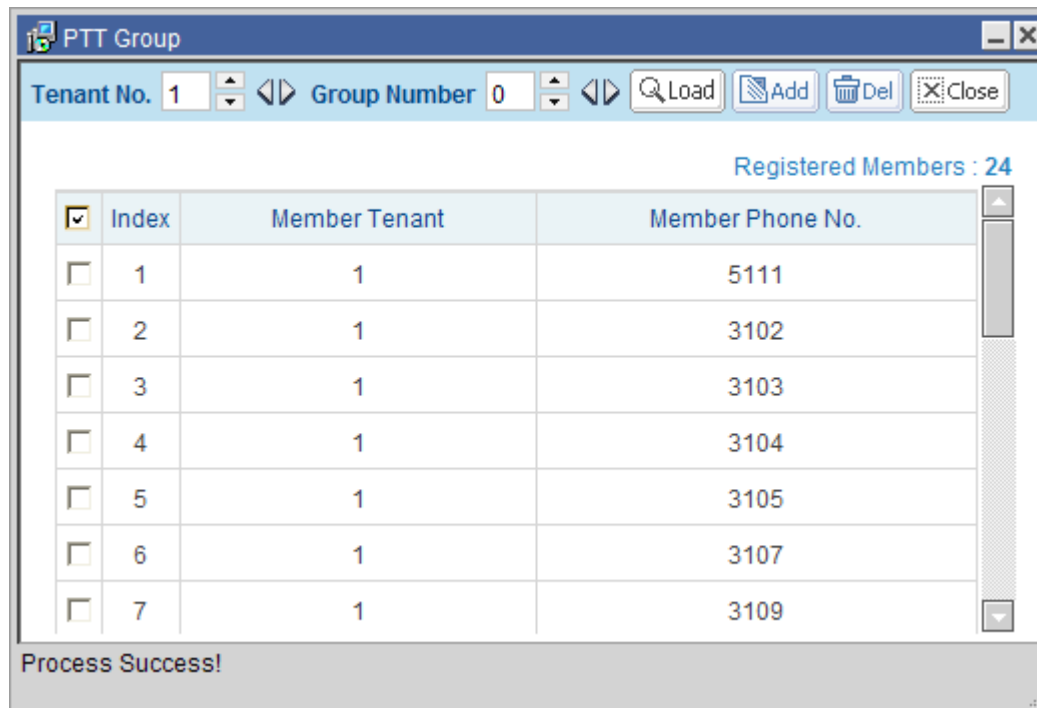
Tenant No. 1

Restrict Class	New Class after Transfer COS Change
1	2
2	1
3	1
4	1
5	1
6	1

Dialed digits(4201) are blocked by Transfer COS

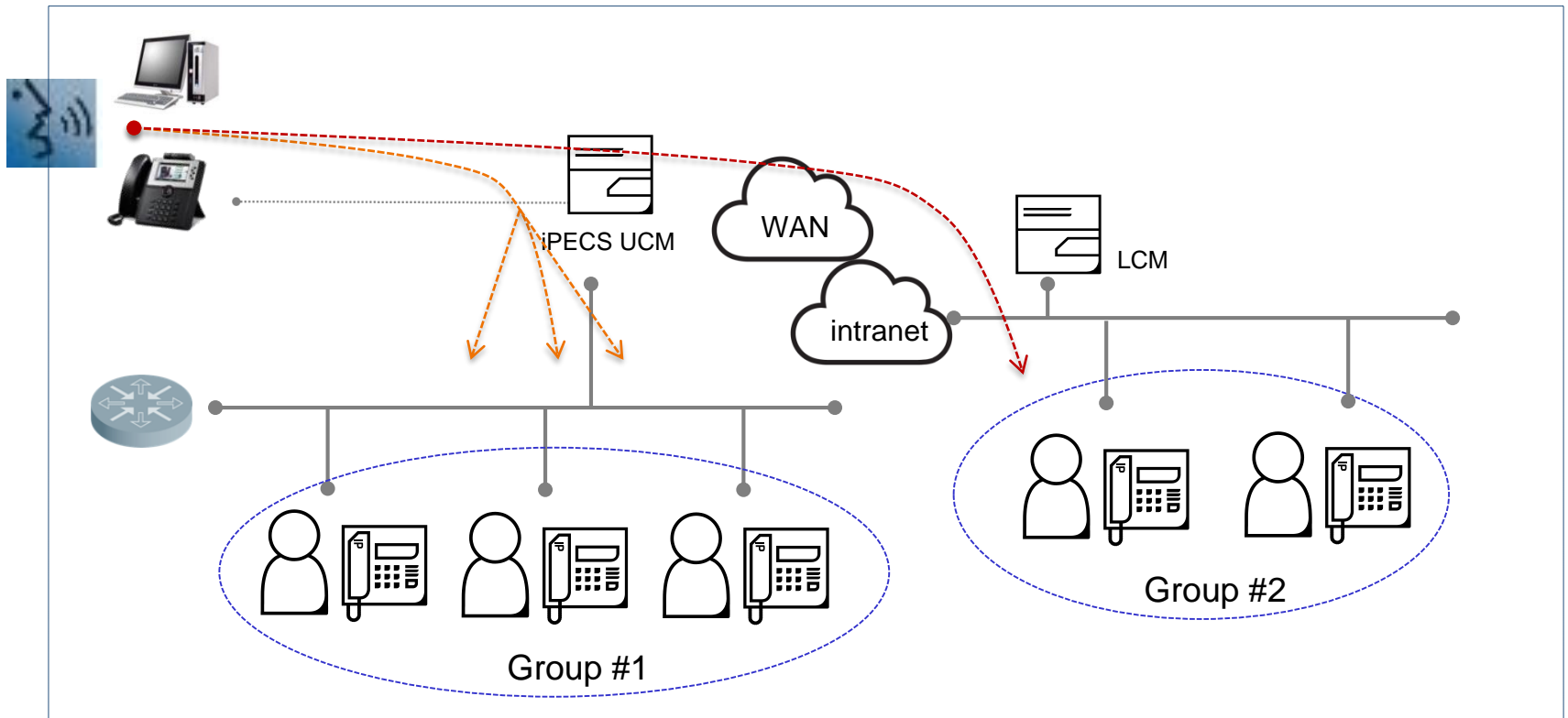
# Push-To-Talk Paging

- iPECS Multi-button phones can be assigned to one or more of ten (10) PTT (Push-To-Talk) groups.
- When logged in, users may send or receive One-way pages.



# Command Call

- A user can place a call to all members in a Command Call Group simultaneously.

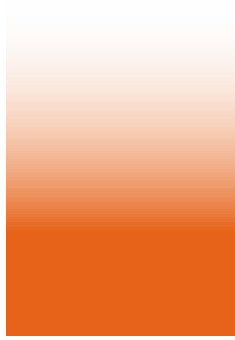




# **iPECS**

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Your Communications Solution



# Appendix

# iPECS Solution Portfolio

## Clients



## Applications



## Telephony Platforms



## Switches





# Contact Center Solution I/F

- Multiple Options for Contact Center Solution to meet each level of Customer needs.

	Routing by	CTI Interface	Applications		License
			Ericsson-LG Enterprise	3 <sup>rd</sup> Party	
<b>Option 1</b>	PBX ACD	N/A	N/A	N/A	N/A
<b>Option 2</b>	PBX ACD & CTI Server	CSTA II	N/A	CSTA CTI Server Package	CTI Interface per CTI agent
<b>Option 3</b>	CTI Server	TAPI	Ericsson-LG TSP	TAPI CTI Server Package	TSP Interface
<b>Option 4</b>	CTI Server	TAPI	Ericsson-LG TSP & iPECS CCS		CTI Interface per CTI agent CCS license