





ANANT

Software based Unified
Communication Server

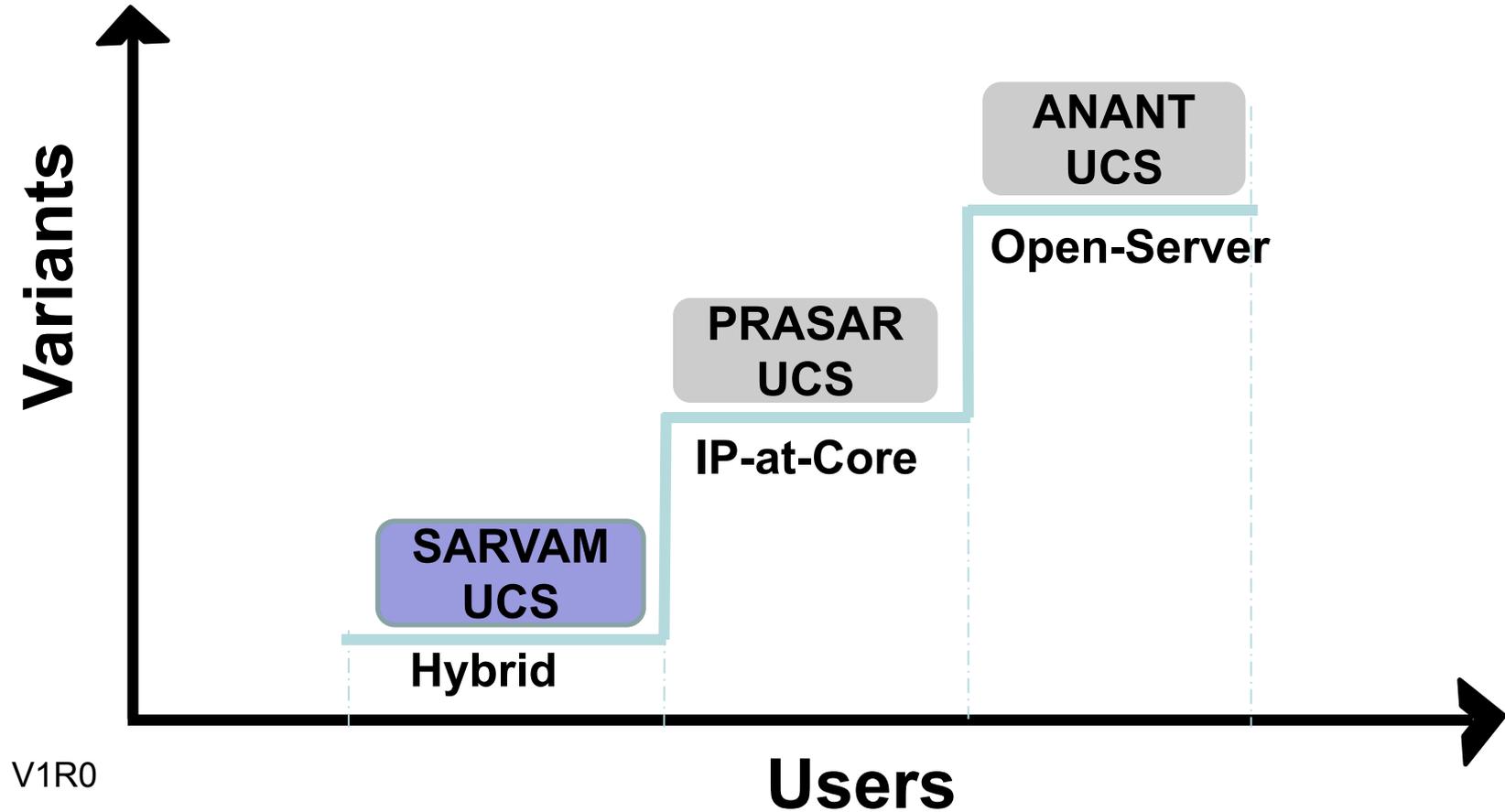
Why Server based PBX?

- Future-Proof Technology
- Hardware Independent
- Scalable
- Reduced IT Infrastructure
- Optimum Resource Utilization

What is ANANT UCS?

- Next Generation IP-at-core Software based PBX
- Server-Gateway Architecture
- Unified Communication Platform
- Highly Scalable
- Open SIP Platform
- Hardware Independent
- Proprietary Software with Secured LINUX OS

ANANT Positioning



Product Highlights

5000

IP Users

1024

Concurrent
Calls

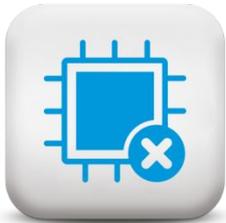
512

Three Party
Conference

64

Simultaneous
Call Recording

Key Features



Hardware Independence

- Hardware of Choice
- Industry Standard Server
- Maximum Resource Utilization



Active-standby Redundancy

- Uninterrupted Communication
- Increased Productivity



RAID1 and RAID5 Support

- Fault Tolerance
- Easy Data Recovery



Mobility

- Anywhere, Anytime Connectivity
- Faster Decision Making

Key Features



Auto Sign-in for Softphone

- One Touch Configuration
- Reduced Installation Time



LDAP Client Support

- Centralized Directory
- Easy Management



Built-in Auto-attendant and Voicemail

- No Missed Messages
- Customer Retention



Auto Sync Configuration

- Quick Configuration
- Smoother Operations

Key Features



Scheduled Back-up of Voice-messages

- Secured Messages
- Hassle-free Back-up



Integration with VoIP Gateways

- Integrated Solution
- Utilize Legacy Infrastructure

Security with ANANT UCS

- SIP Security with TLS and SRTP
- Encrypted Communication
- Blacklisting Unauthorized Access
- Allow SIP Traffic only from trusted source
- Password Ageing
- Role Based System Access and Management

Mobility with ANANT UCS



Voicemail to
Email



Instant
Messaging



Screen Sharing



Outlook
Integration



Flexible Phone Options



IP Phones



Android/iOS
Softphone Clients



IP-DECT



SIP Phones



PC Soft Phones

Integration with VOIP Gateways



VOIP-FXO-FXS

- 32 SIP Accounts
- 32 VoIP Channels
- 32 FXS Ports
- 32 FXO Ports
- 5 Dedicated Variants



VOIP-GSM

- 9 SIP Accounts
- 12 VoIP Channels
- 4/8-GSM/3G SIMs



VOIP-T1/E1 PRI

- 32 SIP Accounts
- 32 VoIP Channels
- 1 T1/E1 PRI Port
- Network Clock Sync

Up to 32
VOIP
CALLS

VoIP
SECURITY

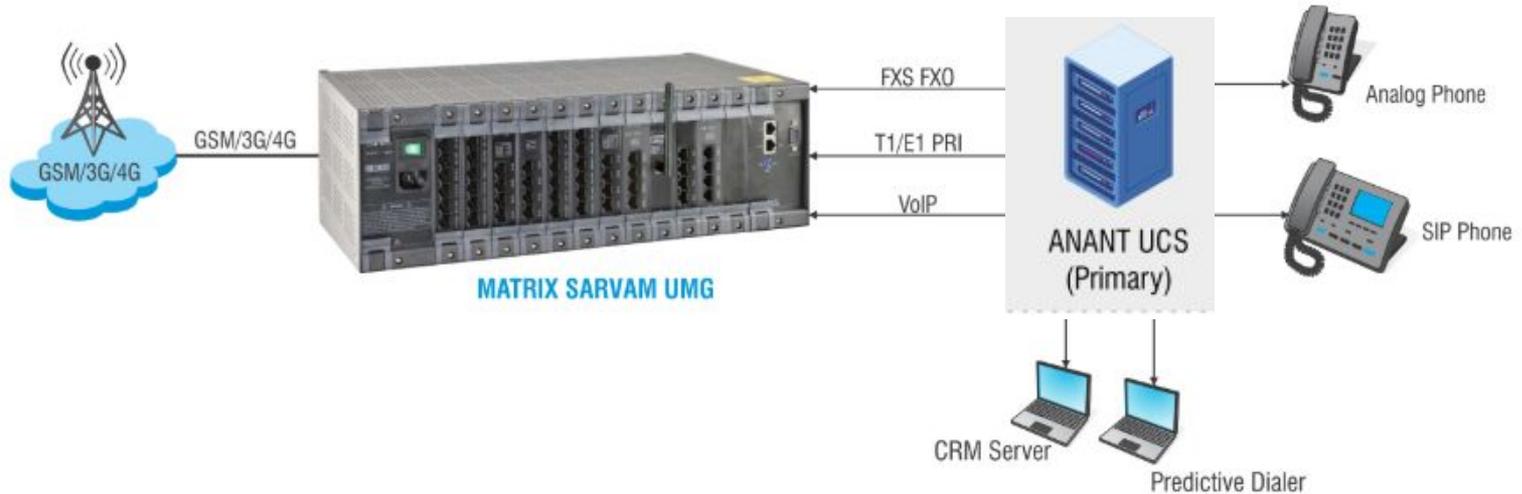
AUTO
Provisioning

STAND
ALONE
Application

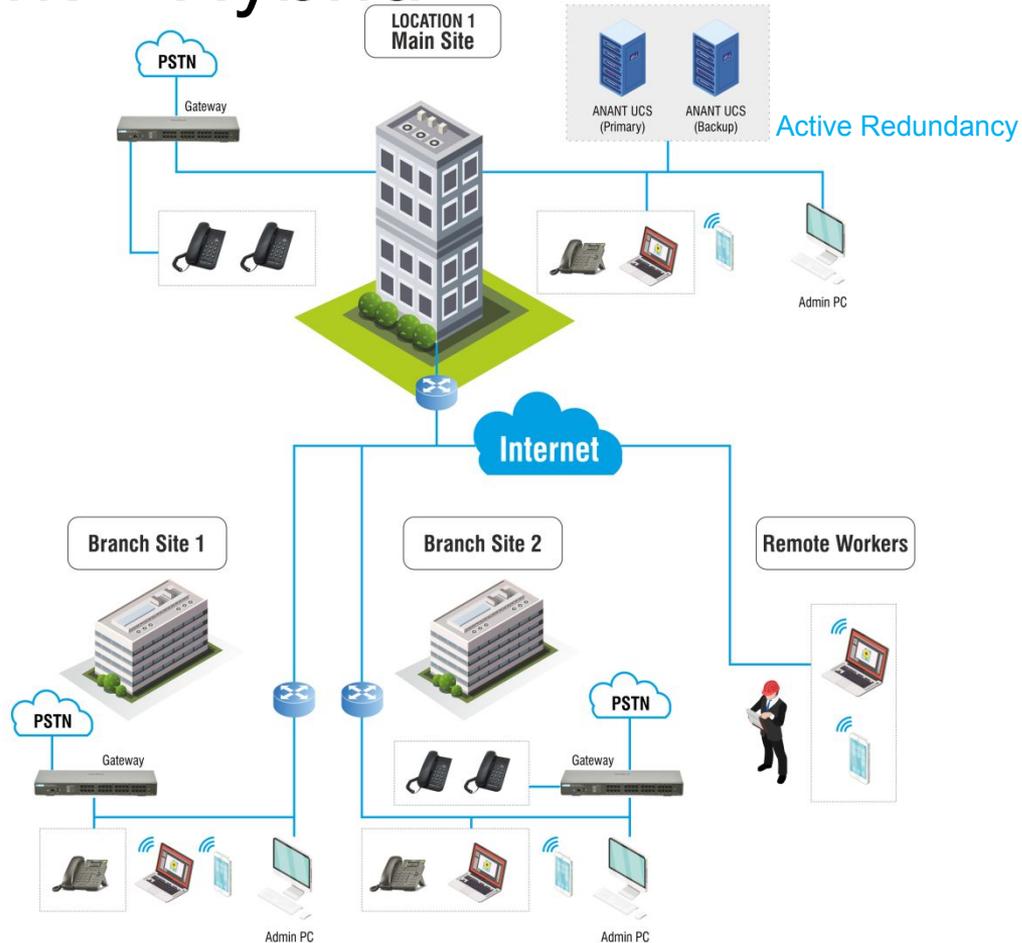
Built-in
PBX
Functionalities

SIP
Interoperability

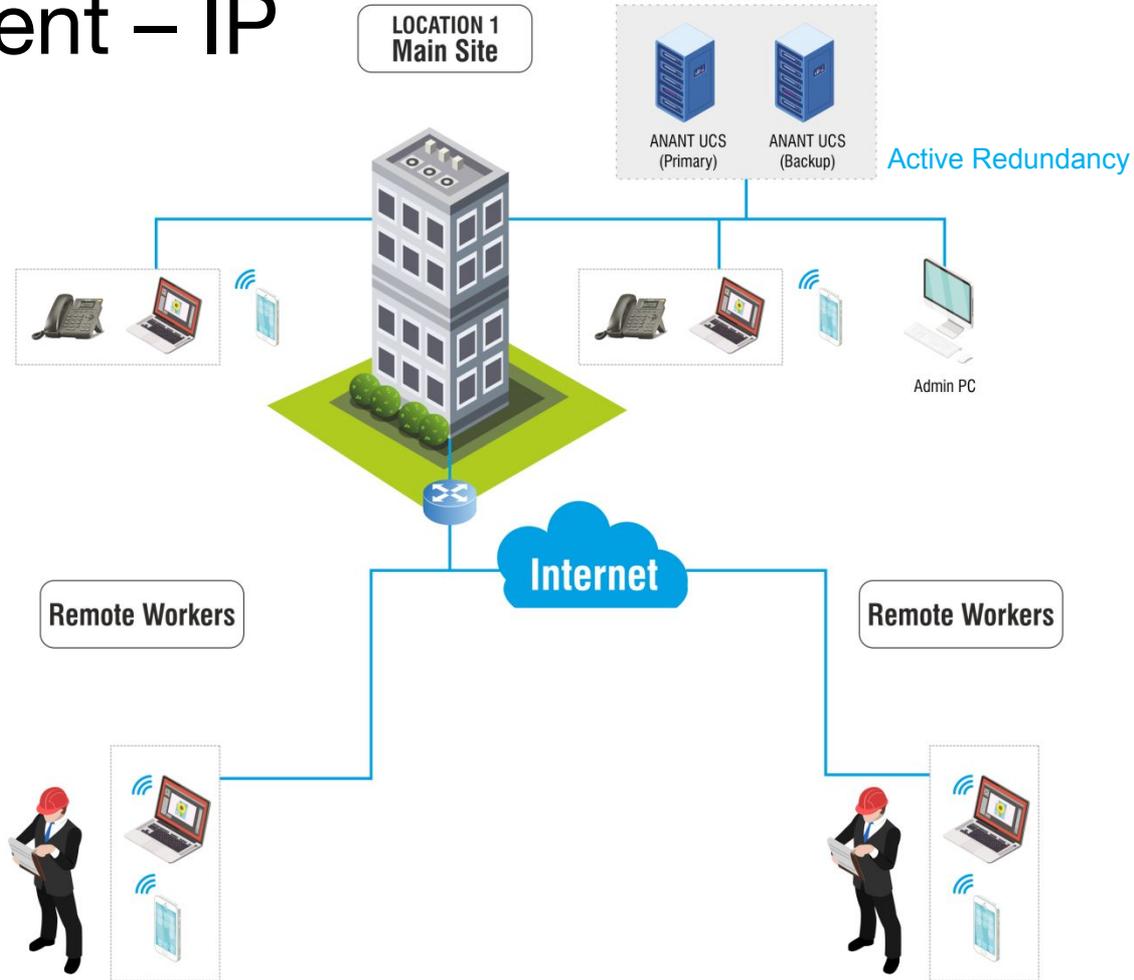
ANANT with Universal Media Gateway



Deployment – Hybrid



Deployment – IP



Technical Specifications

Parameter	Details
SIP Extensions	5000
SIP Trunks	99
Maximum Transcoding Audio Calls	Based on Resources (Max. 1024)
Maximum DRTP/RTP Relay Audio Calls	1024
Maximum DRTP/RTP Relay Video Calls	102
Maximum Voicemail Channels	64
Maximum Conference Participants (System wide)	1536
Maximum 3-Party Conference	512
Maximum Participants in a Single Multi-Party or Dial-In Conference	64
Maximum Call Tapping or Recordings	64

Server Specifications

Parameter	Details
Server Model	Any
Processor Model	Intel Xeon E-2134 or above
CPU Speed	3.5 GHz or above
CPU Core/Thread	4/8
Max Turbo Frequency	4.5 GHz or above
Cache	8 MB Smart Cache
RAM	8 GB DDR4 ECC
Hard Disk	1 TB (1 No.) - Without RAID Support 1 TB (2 Nos.) - With RAID1 Support 1 TB (3 Nos.) - With RAID5 Support
Form Factor	1U (RACK)
Raid Controller (Optional)	H330 Raid Controller*

Industry Applications: Manufacturing

- COSEC Door Integration
- PA System Integration
- Multi-location Connectivity



Industry Applications: Large Enterprise

- 1024 Concurrent Calls
- Conference
- Central Call Processing and Management
- Hot Redundancy



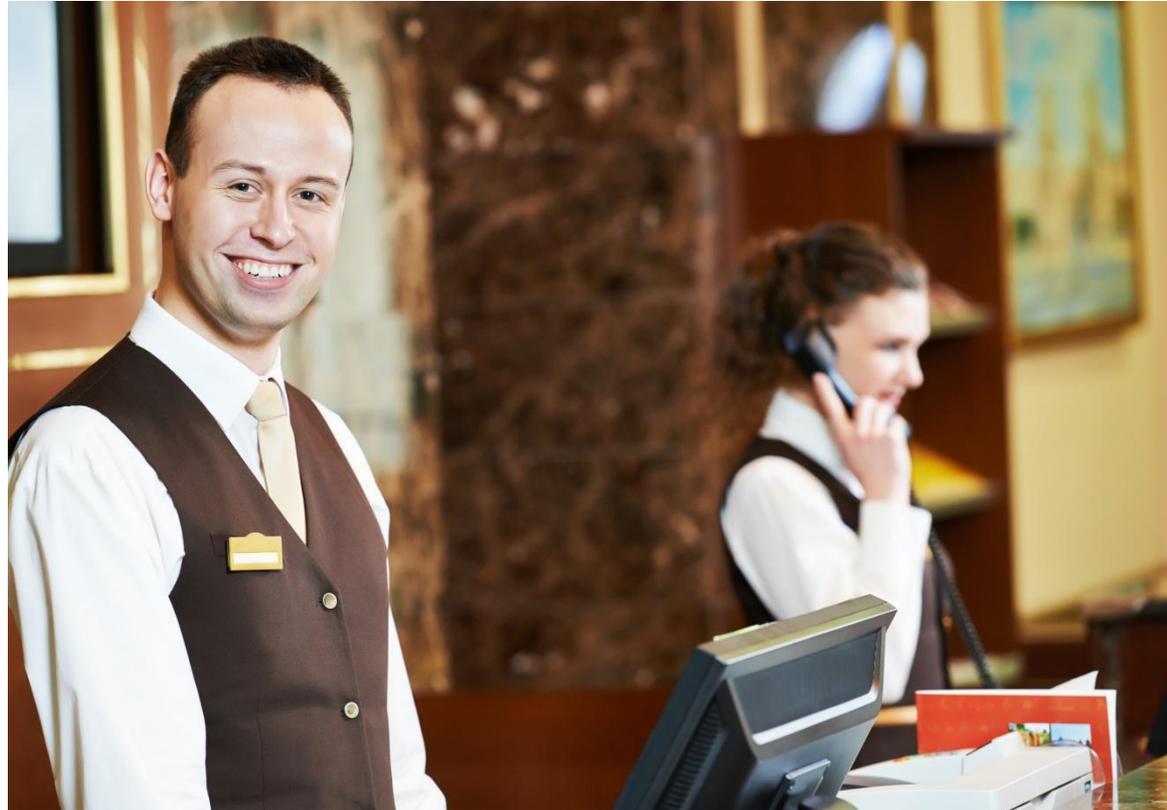
Industry Applications: Mobile Workforce

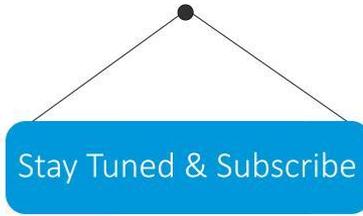
- Mobility
- Collaboration
- Voicemail



Industry Applications: Hospitality

- Intercom
- Auto-attendant
- Voicemail
- Integration with renowned PMS Software





@Matrix Comsec



MatrixComSec



@Matrix_Comsec



@MatrixComSec

www.MatrixTeleSol.com

THANK YOU!



www.MatrixComSec.com