

Extracts from the slides used to discuss comparison of telephony systems – hosted vs on-site

Instructors

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why ?

- Companies need to maximize their investments
- As legacy voice systems reach end of life a replacement has to be found
- ISDN circuits from the telcos are being retired and some legacy systems do not support VoIP
- Costs for maintaining legacy systems are very high and tied to manufacturers - VoIP standards allow a much more flexible manufacturer-agnostic and low cost {zero license fees} approach to telephony



Main Elements for Discussion

- Hardware of the PBX
- Telco trunks
- Telephone end-points
- Call costs
- Support
- Moves, adds and changes

Hardware of the PBX

Does the company have existing ;

- Legacy voice systems and handsets ?
- Expensive telco circuits ?
- Expensive support contracts ?

How important are these items to the company;

- Seamless migration to new telephone system
- Integration of existing legacy systems to maximize investments



Hardware of the PBX



It often makes good economic sense to maximize investments made in the legacy telephony systems and make use of it where possible to integrate and migrate to the new telephony system at a pace suited to the business.

Migrate to the new feature rich system without disruption.

PBX Considerations



- Features required by the business {now and future}
- SLA and need for high availability
- Level of support on site
- Level of support from external consultants
- **Cost of hardware and maintenance**
- End of life predictions for hardware
- Security and updates for hardware and software Integration and flexibility

PBX Considerations



Does the company need public Internet access on the voice systems for enum and inbound Internet calling ?

Does the company need web API to integrate their web site into the voice systems {voice, video, chat etc.} ?

What level of SLA can be expected from a hosted service and/or what level of redundancy will be required for on-site PBX hardware ?

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Essential Features



It is often useful to list the essential features required by the company voice system ;

IN ADDITION TO 'standard features' such as transfer, music on hold etc... IVR menus, ACD features, unified voicemail, conferencing, directory services, CTI integration, CRM integration, video calling, video conferencing, call groups, agent groups, agent skill levels, reporting, voice recording, ease of moves adds and changes...... DISCUSS

Essential Features



Are there any other essential features for the business;

- PCI compliance and security
- Voice recording encryption
- Agent time reporting
- Muti-tenant contact center functionality
- Remote agent and virtual call center functionality
- One number reach follow me
- Hot desking etc



The expensive ISDN trunks are due to retire within the next couple of years. They were based upon a TDM technology and had fixed numbers with very limited redundancy or failover options.

SIP trunks use IP technology and provide all of the redundancy and re-routing abilities of IP.

{show and discuss trunking diagrams}



What are the SLAs offered by the trunk provider ?

Does the business require multiple providers and multiple diverse routes to ensure the high availability of the telephony trunks ?

Is the provider just a middle-man re-selling a trunk service = will this make reporting and resolving problems an issue



The company need to decide if their existing telephone numbers have to be ported to the new providers or if they are OK to be allocated new numbers.

This obviously has implications for stationary and business publications etc. PLUS DID numbers can now be allocated in many different countries and areas - not limited to the business location. Freephone and premium numbers can also be considered.



The company MUST provide a local emergency service number that will contact the correct LOCAL emergency services. This can be setup on some SIP trunk providers or can be provided using a single analogue telephone line.

Other analogue considerations include - alarm circuits, CCTV lines, automation, FAX and franking machine lines etc...



Discuss options for the SIP trunk provision

IF using existing Internet, is QoS configured ? IF using existing Internet, is secure NAT configured ?

- IF using new Internet provision, has it been sized correctly {approx 100kbps per voice channel} ?
- Is QoS configuration required within the company WAN?

Telephone End-Points



The company need to decide if their existing telephone handsets are to be used either during migration or to be fully integrated with the new voice system.

This is not only a cost and investment consideration, but user training and feature requirements consideration.

{discuss options}

Telephone End-Points



The company also need to consider if they are currently using any custom telephone end-points such as conference telephones, FAX machines etc..

Modems, FAX, franking machines etc. need special consideration to ensure they will work through any new VoIP systems.

Call Costs



Assuming the expensive ISDN circuits have been removed, there are several considerations for the new trunk providers

Monthly charge for DID numbers

- Monthly charge for CallerID and other features
- Call rate charges for each geographic area of interest

AND important to note if the provider bills using the actual call time or rounds up to 20 seconds or 60 seconds, which can make a vast difference

Support Considerations



Does the company have the right level of expertise to support the new telephony system for 1st line support?

Does the company have the right level of expertise to support the new telephony system for every day moves adds and changes ?

If not - what are the costs for training or remote support

Support Considerations



What are the support options for the new voice systems ?

Are there ties to a single manufacturer with the added costs and hassles involved with licensing, end of life and compatibility with other systems ?

Moves, Adds and Changes



Can everyday MACs be done in-house at no cost and when it is convenient for the business ?

Is remote support available to assist with any questions and dialplan changes ?

Can complex changes be done to match the business requirements at short notice and at low cost ?



WAN failover

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101 Troubleshooting

Discuss how the following can be used

- MAC address tracing
- ACL counters
- ASA connection details
- NAT tables

Watch out for

- ICMP blocks
- PBR routing
- QoS drops



NMS

Alerts vs SNMP vs syslog vs ICMP polling
limitations

NAT PAT ALG

- Understand NAT terminology
- When is ALG required ?
- How does the proxy work ?
- What are the NAT timeouts ?



DNS



Internal DNS and SRV records

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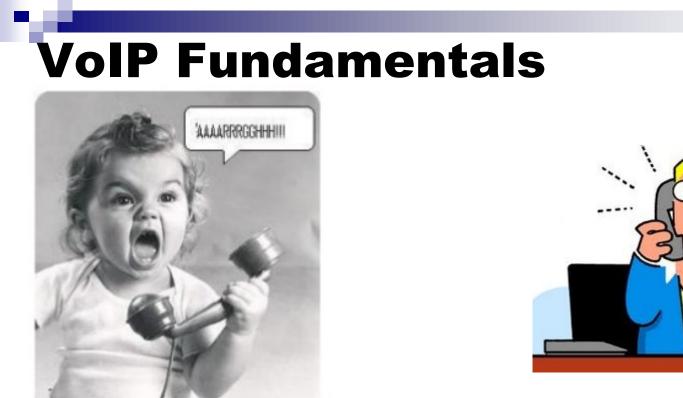


Call Center Requirements

Agent requires basic esentials;

- Telephone {hardphone, softphone, ATA, webRTC}
- Dialtone service {legacy PBX, provider, voip server, hosted}
- PC and screen {web connected, local CRM, CTI integrated}

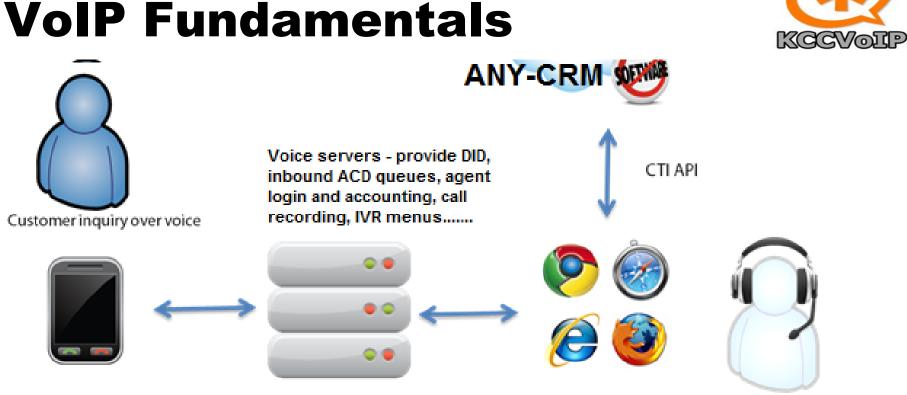
Draw and discuss the alternatives for each and how they are handled through the network





In the beginning the call center had no CTI and no CRM. The customers were very unhappy and no one wanted to be a call center agent

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Press 1 to proceed in English. Press 2 to proceed in Spanish.

Agent receives call and call information





Now the customers are amazed and happy with the knowledge and speed of the call center agent

In House vs Cloud



IN HOUSE

Full control of functionality Requires in-house experts Low cost per seat Very flexible More complex

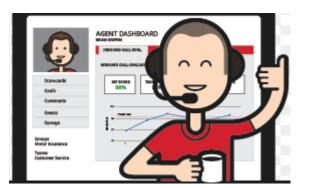
CLOUD PROVISIONED

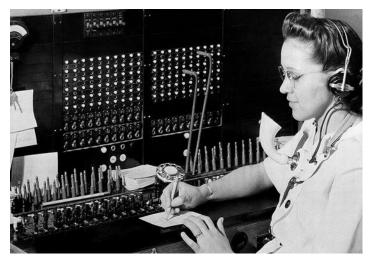
Limited to provider Relies on provider High cost per seat Limited by provider Off the shelf provision



Happy Agents

The more information you can provide to the agents the happier they will be and they will work more efficiently and keep the customers happier.





VoIP Fundamentals



Telephone calls are routed much the same as data

An ip address routes the data to a destination hop by hop 10.32.111.23/32 \rightarrow routed by 10.32.0.0 \rightarrow 10.32.162.4

A telephone number routes the call to the destination

- +1 646 570 1658 + = international prefix \rightarrow 00
 - 1 = USA \rightarrow
 - 646 = NY \rightarrow

explain

QoS



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Call Flow



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Resources and Wrap-Up

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