PART 1

Unified Communications – VoIP Assessment

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Unified Communications – VoIP Assessment Part 1

About this assessment:

These assessment tasks provide an opportunity to demonstrate the competencies covered in the VoIP - Unified Communications subject.

- You are allowed to refer to your text books, notes and the Internet during the Assessment.
- The documentation and research work must be entirely your own.
- By commencing this assessment you confirm that you have read and agree to abide by the ACIT Academic Honesty Policy

Successful completion of this assessment contributes towards attaining competency in the following:

ICTTEN512 Design and implement an enterprise voice over internet protocol and a unified communications network
ICTNWK610 Design and build integrated VoIP networks
ICTPMG611 Prepare a detailed design brief
ICTTEN611 Produce an ICT network architecture design

ICTNWK529 Install and manage complex ICT networks

There are 3 parts to this assessment

1. Design and planning
2. Deployment
3. Question and answer

These assessment tasks provide an opportunity for you to demonstrate the competencies required to design, plan and deploy unified communications solutions.

Part 1 - Design and planning

This part requires you to prepare an architecture design of the unified communications solution and a deployment plan.

Scenario

You are employed by Pacific IT Solutions, an IT services company providing systems integration and consultancy services. You have been appointed as an IT consultant for Online Travel Services (OTS) which is a client of Pacific IT Solutions.

Background

OTS is a national organisation within Australia with 7 retail and sales offices in Australian capital cities and a central office and Data Centre on the Gold Coast. All the sites are connected to the central office via VPN. In total OTS employs 120 people across the organisation with the majority of the staff provided with remote access through either a dial-up or a virtual private network (VPN) connection.

Business Profile

Global Travel Pty Ltd (The Company) is a leading provider of Travel services in the Asia Pacific region with headquarters in Singapore. The Company is the holding group for five smaller companies: OTS, Asia Hotel Services, APAC Flight Brokers, Asia Adventure Holidays and Happy Insurance. All of the companies owned by The Company focus on their individual target markets and employ several hundred staff. The Company is currently trying to reduce its costs and decrease the time it takes to break into new markets. OTS is a wholly owned subsidiary of The Company.

The business was setup in 2001 by the General Manager (Jack Slasher) and run from a small office on the Gold Coast until 2004, when the business was bought by Global Travel Pty Ltd and expanded to have offices nationally. This offered the ability for OTS to expand from the online travel services to offer personalized service to clients through the regional offices. OTS targets the provision of specialist travel packages to high value clients, advertising the packages it offers through the website. The business is currently highly successful, with a turnover in the last financial year of over $100 million and a profit of $6 million.

Current technology

OTS does not currently have any unified communications technologies. The company uses a legacy PBX and all calls are made using the PSTN.

All servers are deployed in a VMWare virtualized environment. They have enough spare capacity in the VMWare virtualized environment to deploy any additional virtual servers they may need. All servers are attached to a Gigabit network.

Requirements

OTS has obtained a feasibility report in relation to migrating their telecommunications systems to a solution that uses a relatively low cost SIP trunking service provided by an ITSP. They have decided to proceed with the migration as it will provide substantial savings on their telephone costs. They will start with an implementation of IP telephony at their Gold Coast office. This means that they will replace their current PSTN lines with a SIP trunk, replace their legacy PBX with a VoIP PBX, and replace their legacy handsets with SIP compatible handsets.

You have been given the task of designing the solution. Your contact at OTS is the office manager, Jane Moss, who will also sign off on the project completion. Your design and implementation plan must provide for the following:

An IP telephony solution that will:

1. Use SIP trunking
2. Use SIP compatible handsets
3. Provide flexibility to expand and link to other sites
4. Provide call parking, call queuing, voice mail, ring groups, and an IVR service
5. Achieve these outcomes with minimal cost

Task 1

Research the following VoIP technologies.

- Asterisk
- ShoreTel
- Cisco

Compare the technologies on the following criteria. Create a table/matrix showing the comparison.

- Cost
- Scalability
- Size of online community
- Quality of support documentation
- Ability to integrate with legacy PBX
- Features - voice mail, IVR, etc.

Answer

. Asterisk: Is an open-source framework for building communications applications. Asterisk turns an ordinary computer into a communications server. It powers IP PBX systems, VoIP gateways, conference servers and other custom solutions. It is used by small businesses, large businesses, call centers, carriers and government agencies, worldwide. Asterisk is free and open source.

Is a framework for building multi-protocol, real-time communications applications and solutions. Asterisk is to real-time voice and video applications as Apache is to web applications: the underlying platform. Asterisk abstracts the complexities of communications protocols and technologies.

It can be used to build communications applications, thing like business phone systems (also known as IP PBXs), call distributors, VoIP gateways and conference bridges. Asterisk includes both low and high-level components that significantly simplify the process of building these complex applications.

. ShoreTel: ShoreTel Premises platform is an on-site IP PBX and unified communications (UC) service for the enterprise. The Premises platform has four main components: voice switches, IP phones, ShoreTel Director and ShoreTel Communicator.

The ShoreTel phone system has an interesting approach and, according to ShoreTel, its main advantage is simplified design, administration and high reliability.

The product´s core is the ShoreTel Voice switch. These are devices that integrate voice functionality, communication, call routing, IP phone registration and SIP trunking or telecommunications provider termination (ISDN-PRI/PSTN) in one box.

The ShoreTel phone system currently supports 14 different voice switches, each with its own characteristics, providing a wide range of deployment choices and telco connectivity capabilities.

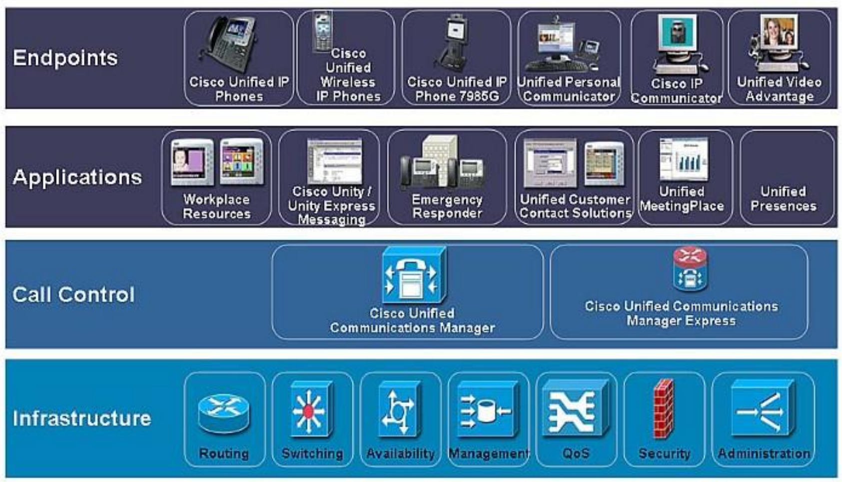
Depending on the amount of IP phones to be supported at a location, customers simply stack multiple voice switches until the desired amount of IP phones has been reached. The highest capacity switch is the ShoreTel Switch 90V, which provides support for up to 90 IP phones, 90 voicemail boxes, up to 12 analog ports and SIP trunking to the Internet service provider. Companies that require more connectivity can purchase and stack multiple switches.

The ShoreTel phone system has an extensive range of IP phones with several features for ease of use. All the phones support Power over Ethernet and have Fast Ethernet (10/100 Mbps) or gigabit Ethernet ports, depending on the model.

MITEL, a global leader in business communications, announced the completion of its acquisition of ShoreTel. With this addition, Mitel has accelerated its move-to-the-cloud strategy, shifting into the #2 market share position for UCaaS (Unified Communications as a Service) globally as customers worldwide look for cloud and cloud -capable solutions to digitally transform their businesses.

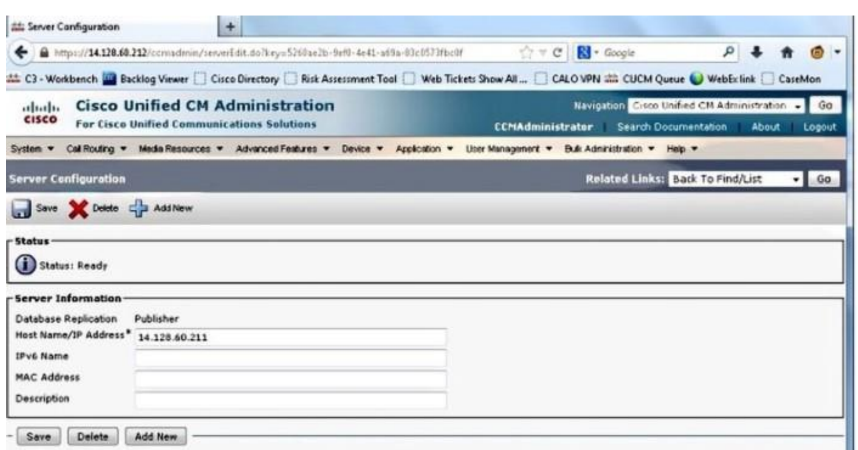
. Cisco: In 1997 Selsius Systems came out with one of the first IP PBX systems consisting of a line of IP phones, VoIP gateways, a server-based call control application called Selsius Call Manager, and many voice applications such as voicemail. One year later Cisco acquired Selsius and renamed their call control application to what we know as Cisco Unified Communications Manager (CUCM).

Cisco Unified Communications (UC) is Cisco´s product for Voice over Internet Protocol (VoIP) communication. This set of products used to be called IP Telephony (IPT). It integrates voice, data, video, and mobility products and applications and represents a paradigm shift like one of the telegraph inventions.



CUCM sometimes referred to as Cisco Call Manager, is a popular call management software that is installed on a Linux platform, as opposed to Windows. Once installed, administrators access the call manager´s graphical user interface (GUI) by entering the domain name or its IP address in the URL bar in a web browser.

Cisco continues to be the leader in IT solutions as VoIP technology continues to grow.



	ASTERISK
Cost	Free, Open Source
Scalability	**As we know, scalability depends on multiple factors such as cpu / ram / storage, bandwidth and gateways. Doing the calculation with our specification proposal we have come to the conclusion that we could have 100 users and 500 simultaneous calls that we could keep running without problems**.
Size of online community	_Worldwide, used by almost 1000 customers and is deployed by more than 170 countries, made up of 86,000 registered users.
Quality of support documentation	**The quality and quantity of documentation and support is very good, it contains a spectacular forum room to discuss ideas and share knowledge, as well as an online help line that is able to respond quickly. It has updated information both audiovisual and diagrammatic. The videos are clear and the website is very well designed so that the search for information is effective through the linking system.**
Ability to integrate with legacy PBX	_As enterprises continue to evolve their communications infrastructures into a pure SIP/VoIP environment, they often discover that there is still the need to connect older non-VoIP equipment to their VoIP infrastructure or directly to the PSTN. Many of these older, but necessary endpoints simply will not function reliably over a VoIP network. There may also be instances where it is more cost effective to use a "Plain Old Telephone" as an endpoint on a VoIP network, rather than an expensive VoIP phone. Ruggedized phones may already be in place on construction sites or in other locations where the risk of damage is great. Entry and elevator phones may already be integrated into existing systems that cannot be easily replaced by an IP endpoint.
Features - voice mail, IVR, etc.	_Voicemail _ Interactive voice response _Conference calling _ Automatic call distribution

	Shoretel
Cost	$200 per user (license cost)
Scalability	The ShoreTel solution comes in one size. Thanks to an ingenious purpose-built IP architecture, the system looks and behaves like a single, unified platform and is easily expanded across as many sites, or even countries, as needed. _ShoteTel is scalable from 5 phones to 20,000. Built from the ground up for reliable IP communications, ShoreTel deploys, scales and grows with ease. End users live its flexibility and ease of use, providing a rich feature set whether you are big or small.
Size of online community	_Provides one-step resource shop for development tools, _Showcase application, discussion forums and developer support
Quality of support documentation	**With more than 70 million users in 100 countries, Mitel´s technical writers have created almost 10,000 user guides, technical documentation, and other useful information to help customers and partners learn how to set up and use Mitel solutions in any workplace.** **While a typical document has about 200 pages, some can have up to 2,000 pages, many of which are translated into a dozen languages. Like many companies, part of Mitel's growth strategy has involved acquiring other companies. As a result, Mitel's technical writing team inherited documents, tools, and portals used by different companies. Mitel decided to minimize changes to documentation and maintain existing portals to deliver the same experience for existing partners and customers.** **The company decided to join XML Documentation for Adobe Experience Manager, to manage its documentation, which I think is very good since** **allows you to create, reuse, and render content, in a single portal. They also decided that the system of administration and creation of documents was simple and centralized and prepared for the future, so they decided to structure the content in DITA format, which favors the reuse of complete themes, as well as partial, through content references. Therefore, it took them little time to create templates and start rendering the DITA content in PDF and HTML5.** **It seems to me that Mitel's search development in how to handle its documents, adapts very consciously to the new times of technology, where we require information that is fast, clear, effective, friendly and in one place, which gives us the confidence that the content is real, without alterations and easy to interpret.**

Ability to integrate with legacy PBX	One of the more interesting aspects of PBX system installation in general and ShoreTel in particular, is the subject of legacy PBX integration. There are a variety of reasons that a new ShoreTel installation might need to integrate with the old, in place or “legacy” PBX phone system. You might be installing the ShoreTel at the first location of a multi-site installation with the rest of the sites coming on line as older equipment leases expire. PBX´s typically use a tandem tie-line to join systems together. The ShoreTel, in this instance, would know the dial plan of the other PBX extension number or receives a call for an extension known to live across the tie-line, the call is sent to the other PBX. The tie-line is typically define as part of a trunk group that outlines a list of “off-premise extensions”. The ShoreTel can also provide digit translation and manipulation to accommodate over lapping dial plans.
Features - voice mail, IVR, etc.	_Voice Mail Functions _Transfer a Call _ IVR _Set call priority _ Call Handling modes (Out of Office) _ Place a Conference Call _ Auto Attendant

Cisco

Cost	$50 to $700 per phone. Installation cost average $500 and $2500 per device, and monthly costs can range from $25 to $250 per user.
Scalability	Cisco has at its disposal an incredibly large and diverse product range to architect this vision. The Cisco Unified Communications range from the backbone of Cisco´s networked audio, video and presence systems. The Smart Business Communications System (SBCS) is geared towards small businesses of up to 50 users and offers a range of features including Outlook and Lotus Notes integrated messaging, voicemail, conferencing and WebEx support as well as basic firewall and remote access functionality. For bigger businesses is the router-based Unified Communications Manager Express and the server-based Unified Communications Manager Business Edition, supporting up to 240 and 500 users respectively. At the enterprise level sits Cisco Unified Communications Manager, supporting up to 30.000 users per cluster and using a centralized system that allows up to 100 units to be combined into a single system.
Size of online community	The community is a hub for connecting with peers and Cisco specialist to ask for help, share expertise, and build network. This month (september 2021) the community surpassed one million registered Cisco Community members. And the tren is accelerating: the past three years almost 70.000 customers on average have become registered members, up from 40.000 per yeat over the previous 20 years.In fact, on average, more than three million people visit the

	Community every quarter, viewing 15 million pages. These pages include more than thirty-five thousand discussions in eight languages, including the new Korean and expanded Portuguese communities.
Quality of support documentation	**Through its support page https://www.cisco.com/c/en/us/support/index.html You can access multiple documents and sources of information on the most varied topics, either by category, communities, licenses and others. I am struck by the effort to keep the community updated, with news, opinion forums, new tools besides being able to create an account where the notifications that Cisco can send can be configured as an inbox. It seems to me a very well thought out alternative that comes from a business mind.** **They have a tool called Cisco CLI Analyzer which is a smart SSH client with internal TAC tools and which was designed to help troubleshoot and check the overall health of a Cisco supported software.** **It has a very good worldwide telephone support https://www.cisco.com/c/en/us/support/web/tsd-cisco-worldwide-contacts.html this is very impressive, here we can appreciate that Cisco is one of the largest IT companies globally.** **From what I can see, it has a very intuitive customer support unit, based on documentation search engines almost like a google but especially focused on all Cisco technology.**
Ability to integrate with legacy PBX	Cisco has long recognized as the technical powerhouse driving innovation, development and business uptake in the VoIP and IP telephony markets. The company´s heritage as a specialist networking vendor has uniquely positions Cisco in the market compared with its telecommunications-based competitors, to aggressively promote the convergence of traditional IP and telephony infrastructures through the 1990´s. With that legacy Cisco is now concentrating on unified collaboration-the ability to offer customers a seamless end-to-end communications experience incorporating both audio and video, across a wide range of hardware and software platforms while removing the technical specifics from the conversation.
Features - voice mail, IVR, etc.	Calling features, Voicemail to Text/Email, Call merging/call waiting, web conferencing, video conferencing, conference calls, business texting, auto-attendant, interactive voice response, integration, google apps, Microsoft office, Microsoft office, Microsoft outlook, skype for business, salesforce, Zendesk

Task 2

Write a description of the solution you envisage. Assume that you have chosen Asterisk as the solution. You should include:

1. A description of the architecture - include VoIP, IP telephony, network, and firewall elements

The plan and architecture that Pacific IT Solutions proposes to carry out the change of the call system of the OTS company will be based on the Asterisk system. This is because it is Open Source, meaning that it is free without the need to install any other hardware or device that requires an additional monetary cost. In addition, it is worth mentioning that we will install FREE PBX that will provide us with the opportunity to work with Asterisk GUI, giving us a better perspective to configure and monitor the system. It will be configured within the virtual machine environment where we have the space and speed indicated for the installation. The Asterisk installation will be carried out with the Free PBX Bistro package, which will give us the opportunity not only to install the program, but also to maintain and update it.

After the installation of Free PBX we will proceed with **the layer-3 24 ports PoE Switch Cisco CBS250 Installation** and the configuration of some extensions thinking about the 3 departments that we have designed for the company: Production, Department of Finances, Department of Marketing and we will supervise that there is connectivity between the different devices calling from one extension to another connected to our own PBX, using softphone software such as Zoiper and Linphone.

Next, we will proceed to the configuration of a gateway **g800 8 port software-selectable T1/E1/PRI appliance that supports up to 240 concurrent calls and we will configure it as firewall also** and then SIP trunks by buying it from an ISTP (Internet Telephony Service Provider), to connect our PBX with PSTN (Public switched telephone network) the classic telephone network where a real-time voice communication is given, ensuring fluidity in network traffic. Additionally, we will proceed to configure trunks under the IAX2 protocol to connect our PBX to any other PBX and thus have the possibility of increasing the scope if required in the future.

After this we will go on to configure Outbound Routes that will allow us to route the calls to the other PBXs that we want to connect and to PSTN.

Finally, we will configure inbound routes to our own PBX to receive calls from PSTN.

As a last procedure, we will configure IVR, call parking, call queuing, voice mail, ring groups, so that the customer experience is efficient, fast and meets the professional requirements of the company. The system will ensure that, through the different dialing options, the client or whoever communicates with the company navigate through it in the most efficient way and can contact the person they want quickly with an effective cost of time.

The architecture that we have previously described is designed to be developed at the lowest possible cost. Asterisk, Free PBX and in general Free PBX bistro is opensource, therefore, it is free. The sip trunk that will be purchased does not exceed $50 per month. And the IP phones will be Grandstream GXP1620, $55 each.

Next, we will describe what is VoIP, IP telephony, network and firewall elements:

Voip: Stands for Voice over Internet Protocol. It refers to making phone calls that are made through the Internet, rather than through a regular landline or a mobile network. A VoIP system works by taking the analogue voice signals, converting them into digital signals, then sending them as data over the broadband line.

Ip telephony: (Internet Protocol telephony) is a term used to describe technologies that use a variety of protocols to exchange voice, fax, and other forms of information, traditionally carried over the public Switched Telephone Network (PSTN). The call travels in the form of packets, over a Local Area Network (LAN), or the Internet, avoiding PSTN tolls.

Network: Our network configuration will be ip address 10.10.41.139 255.255.255.0 gateway 10.10.41.1, DNS server list 127.0.0.1, 10.10.100.21, 10.10.40.10

Firewall: We will use The AlliedWare firewall that include the functionality required to perform the update of SIP packet payloads.

The structure of the packets involved in a SIP exchange is relatively complex, and involves embedded IP addresses (IP addresses embedded into the payload of the packet not just IP addresses in the packet header).

Protocols that involve embedding IP addresses into packet payloads are typically rather challenged by NAT. SIP is no exception to this rule. In order for SIP to operate successfully through a NATing firewall, it is necessary for the firewall to be able to reach right into the payload of outgoing packets, and translate any embedded private addresses into the correct public addresses (and vice versa for incoming packets).

The agent within the firewall that performs this activity is referred to as the SIP ALG (Application Level Gateway).

2. A thorough explanation of how your solution will work - the exact process of how calls are made, received, interfaced with the PSTN

In our PBX, we have configured 3 extensions (Production, Department of Finances and Department of Marketing) and a General VoiceMail. In addition to Time Conditions and an open and closed business IVR. In addition, 3 trunks have been configured: Jaime.tafoya@pbx iax2 trunk (teammate PBX, with its outroute duly configured), admin@pbx iax2 trunk (to connect our PBX with PSTN, therefore we will have connectivity with any number outside the plan of internal dialing) and we also created PSTN trunk towards the SIP provider, to route any outgoing dialing from our PBX to the SIP provider, allowing the receiver to see an incoming call from their own assigned DID, instead of using the administrator's DID number.

We have also created outbound routes towards the Admin PBX, so that calls are routed from any extension through the Sip trunk, to the PSTN.

We have also configured an inbound route to Time Conditions, which means that any call that comes in both locally and from PSTN will be routed to the Time Condition application.

3. An explanation of how your solution meets the business requirements

One of the main requirements of our client is to be able to carry out everything mentioned at the lowest possible cost, therefore and first of all we must mention the solution of working with the Asterisk system for this new system, which is OpenSource therefore so much is free. The fact of working and configuring trunks for call routing will also make the monthly cost of the service much cheaper than implementing a normal and old-fashioned telephone service, since for a lower cost we can have a greater number of lines available in the same trunk. To connect other link sites, we will use the iax2 protocol and for handsets and softphones we will configure SIP protocols. Of course, taking into account the scalability of the system, the solution that we are proposing will have to pay us an extra amount, in future extensions of lines and connectivity.

4. A description of each component

IP telephony: (Internet Protocol Telephony) is the term used to describe data communications which exchange phone calls, voice, fax and other forms of communications over the internet. Where traditionally, voice calls would be carried via analogue PSTN line, IP Telephony transport this voice data in packets via LAN or the internet.

VoIP: (Voice over Internet Protocol). It refers to making phone calls that are made through the Internet, rather than through a regular landline or a mobile network. A VoIP system works by taking the analogue voice signals, converting them into digital signals, then sending them as data over the broadband line. It is a very useful way of making calls- for a start, once it is set up it is a lot cheaper than using normal phone lines. It means that, depending on the setup, it may not have to pay for phone calls based on distance, which country is called, or how much time spent chatting.

Asterisk: Is an open-source framework for building communications applications. Asterisk turns an ordinary computer into a communications server. It powers IP PBX systems, VoIP gateways, conference servers and other custom solutions. It is used by small businesses, large businesses, call centers, carriers and government agencies, worldwide. Asterisk is free and open source.

PBX: (Private branch exchange) is a telephone system within an enterprise that switches calls between users on local lines, while enabling all users to share a certain number of external phone lines. In contrast to a PSTN, the main purpose of a PBX is to save the cost of requiring a line for each user to the telephone company´s central office.

Softphone: Is a type of software-based phone. It allows you to make phone calls over an internet connection without needing designated physical hardware, and it can be installed on desktops and mobile devices. In short, softphones help you make telephone calls without an actual telephone.

SIP protocol: (Session Initiation Protocol) is an open signaling protocol standard developed by the Internet Engineering Task Force (IETF) for establishing, managing, and terminating real-time communications over large IP-based networks, such as the Internet. Communications via voice, video, or text (instant messaging) may take place using any combination of SIP -enabled devices, such as a softphone on a laptop, a wireless handheld device, a mobile phone, or an IP phone with videoconferencing capabilities.

Sip is an application layer peer-to-peer communications protocol for establishing, manipulating, and tearing down communication sessions. But the protocol is extensible. It is used to identify, locate, and enjoin parties who want to communicate using any peer-to-peer media type. However, SIP does not transport the media itself: That is handled by codecs within the communications sessions.

SIP trunks: Is a virtual connection between an Internet Protocol Private Branch Exchange (IP PBX) and a telephone service provider, providing SIP-based voice and UC services, connected over an enterprise´s data network connections.

ISTP: The SIP sessions (whether they are voice calls, video conferences, multimedia sessions, etc) need a public network provider to get to where they are going if they are no internal calls handled in the enterprise network. That is the role of a service provider, just as it was with traditional circuit-switched voice.

A service provider that provides transport and termination of SIP calls is the ITSP, sometimes also called a SIP service provider. It can really think of an ITSP as just a “phone company” with the difference being their interface with the network is a data connection using SIP to control the flow and routing of sessions.

PSTN: (Public switched telephone network) is the world´s collection of interconnected voice-oriented public telephone networks. PSTN is the traditional circuit-switched telephone network. PSTN comprises all the switched telephone networks around the world that are operated by local, national or international carriers. These networks provide the infrastructure and services for public telecommunication.

IAX2: Is a VoIP protocol native to Asterisk. It was created as a means of easily establishing trunks between Asterisk servers, hence the name, "Inter Asterisk eXchange".

The initial version of the protocol (IAX) is now deprecated. The current version is officially known as IAX2, but it is commonly called just "IAX".

IAX2 is attractive in a few ways: Connection Simplicity / NAT Traversal, trunking / bandwitdth conservation, Security / Encryption.

Outbound routes: Outbound routing is a set of rules that the PBX uses to decide which trunk to use for an outbound call. Having multiple trunks allows you to control cost by routing calls over the least costly trunk for a particular call. Outbound routes are used to specify what numbers are allowed to go out a particular route.

Inbound routes: Inbound routing is one of the jey pieces to a functional PBX. The inbound routes module is the mechanism used to tell the PBX where to route inbound calls based on the phone number or DID dialed. This module is used to handle SIP, PRI and analog inbound routing. Also are often used in conjunction with time conditions and IVRs.

IVR: (Interactive Voice Response) Allows callers to access information via a voice system which has been pre-recorded. An IVR is used automatic method for callers to utilize menu options via touch tone keypads selection or speech recognition to have their calls routes to the specific destination.

Call parking: Allows you to park a caller so that another extension can retrieve the caller held in the parking lot.

Voicemail: Is a feature which allows you to prompt a caller to leave a message in the event that no one answers an incoming call.

Ring-groups: It is a feature which will ring all the extensions simultaneously when an Incoming call is detected. If not answered it will go to a shared Voicemail box which should be accessed by any extension.

Time conditions: Is the communication feature in Hosted PBX that enables the businesses to set up distinct time patterns for call handling. It permits to route after-hour calls to various destinations at a different time like working hours, holidays or even lunchtime.

Handset: Is a component of a telephone that a user holds to the ear and mouth to receive audio through the receiver and speak to the remote party via the built-in transmitter.

5. The rational for the design decisions you made

Why Asterisk IP-PBX system?

We have chosen as a solution an IP-PBX based system and specifically Asterisk because it is becoming more available from smaller upstarts, changing forever the traditional proprietary PBX manufacturers stronghold on the business PBX market. In fact, Asterisk is an open-source PBX software and has captured the dominate position of PBX software. It has proved itself to be efficient, cost effective and stable, and something that the industry giants are having to compete with. Also includes advanced communication features, like voicemail to email, but also provides scalability and the ability to connect to traditional PSTN lines, so there is no need to change carriers. In addition, Asterisk has an excellent external platform (gui) called free PBX, which provides us in a friendly way the opportunity to configure and monitor everything that happens in our system.

It uses the Lan: An IP-PBX business phone system will reside on the network using existing LAN. The PBX server is only a short distance away, so signaling distance and time latency is very short and does not depend on traveling over the Internet and other networks.

Lower operational costs over time: In addition to taking advantage of lower cost VoIP routing, purchasing the IP-PBX lowers costs over time.

Easier to configure and install than proprietary phone systems: Propietary phone systems can be cumbersome and difficult to navigate around their software to configure and install. An IP-PBX system will be much more familiar to computer people, especially someone who has experience with networks.

Why SIP trunks?

Using SIP trunking with an in house IP-PBX can connect to lower cost VoIP providers; reducing phone bills, especially long distance and international calls.

Why Softphones?

Desktop softphones apps increase team´s flexibility and offer benefits such as video conferencing, messaging, contact integration, presence etc. Also offer several advantages for customers, including decreased costs from licensing fees, easier contact importation, and seamless integration with localized features

Geographic Flexibility, Work Anywhere: A desktop softphone allows users to make and take phone calls from any computer with an internet connection. This enables employees to stay connected away from the office, whether they are at a café, an airport, etc. It also saves on hardware costs-instead of a desk phone, a user needs only a headset for optimal call quality.

Contact Integration, save time and effort: Most desktop softphones can pull contacts from outside sources, such as a user´s email account,.csv file, or a computer´s contacts app.

Video and messaging, enhanced collaboration: Video and messaging are two key ways in which desktop softphones surpass most desk phones. Video capabilities allow employees to participate in one-to-one and multi-party video conferences.

Why IVR?

Intelligent Call routing: IVR phone system technology can intelligently route calls based on the caller´s telephone number. This means that calls can automatically be directed to the last consultant spoken to, VIP callers can be placed at the front of the calling queue, or calls can be directed to a specific consultant that speaks a particular language, the options are endless.

Support for busy times and disaster recovery: IVR contact center technology gives organizations options for their callers to self-serve and leave messages. In times of high demand due to emergencies or seasonal peak activity, the IVR self-service feature can support the contact center by easing pressure on the agents.

IVR technology can be invaluable during times of disaster recovery. A preprogrammed alternative route can be activated in the event of an emergency so that selected calls can be routed to an alternative call center or to staff working from home during severe weather or epidemics such as COVID-19.

Improving the customer journey: Traditionally inbound IVR systems (and outbound IVR calls) have a bad reputation. Poorly planned systems can frustrate callers with their complex and clunky designs. Today, IVRs are revolutionizing the customer journey with clear messages that reflect te organization´s brand, providing an effective self-service tool.

Task 3

Compile an inventory of hardware and software components. You must include the recommended hardware requirements for an Asterisk/FreePBX installation for an organization of this size.

Software

Quantity	*Item*
1	**Virtual machine 4gb ram/i3 cpu processor / 1Tb (raid 1)**
1	Free PBX Bistro
1	Asterisk
1	Sip Trunk
1	Zoiper (softphone)
1	Linphone (softphone)

Hardware

Quantity	*Item*
3	*IP PHONES Grandstream GXP1620*
1	Switch PoE
3	Headphones Jabra evolve 20 Stereo

Task 4

Prepare a work breakdown of your deployment plan.

Step 1

Product quotation: Software and Hardware, visit to the company and meeting with clients. SIP trunk contracting. Virtual design and diagramming. Inspection and approval of the VMWARE environment and the internet connection. Project possible conflicts and problems that may arise in the installation of the system. Modify budget if necessary.

Step 2

Free PBX Bistro suite software installation and hardware such as telephones and their corresponding cabling. Installation of both zoiper and lindphone softphones and protocol configuration. Poe Switch Installation

Step 3

Configuration of Extensions of the 3 departments. SIP Trunks configuration and connectivity with ISTP. SIP Trunks with IAX2 protocol, Outbound and inbound routes configuration.

Step 4

Configuration and setup of IVR, call parking, call queuing, voice mail, ring groups, outbound route to PSTN and also the configuration of Time Conditions.

Step 5

This step will be the testing area of the entire system, connectivity will be tested between the PBX extensions and with external extensions to other PBXs. The IVR will be tested, the correct call derivation process will be confirmed with the dialing process and that each recorded message corresponds. The softphone configuration will be observed and corrected if necessary

STEP 6

It will be coordinated with the working group in a future monitoring plan both weekly and monthly, possible problems or saturations that may emerge will be prevented, and solutions to these will be proposed in advance.

Task 5

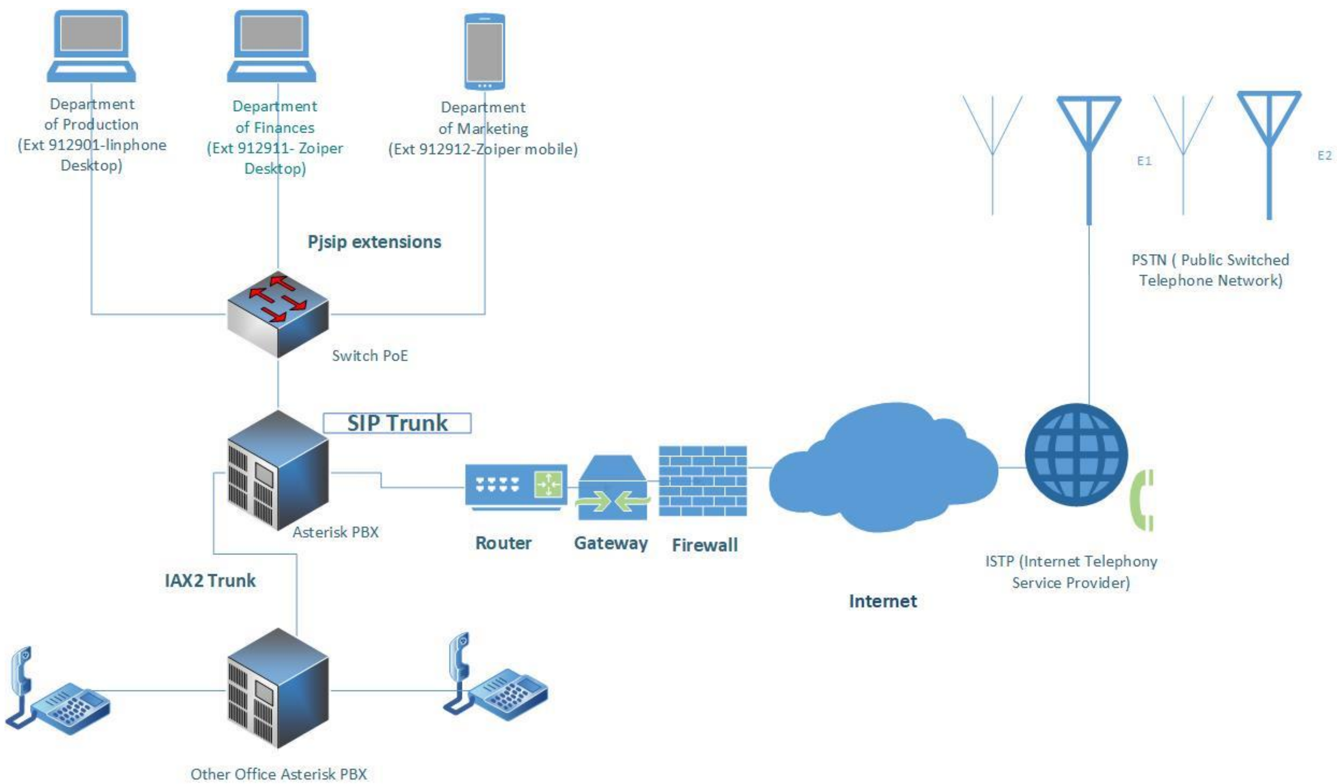
Write a post deployment test plan.

Test Plan
OTS Company

Test option and procedure	Results we should expect
Virtual Machine Test and Network Test	We should have access to our virtual machine environment without problems and have our network configurations both our Ip address 10.10.41.139 and our gateway 10.10.41.1 correctly configured; we will ping between servers to check connection.
We will make calls between the extensions of our PBX	Calls between the production department, finance and marketing should be directed where appropriate and the call activated to the appropriate softphones. For example, extension 912911 for the finance department receives calls on a laptop with Zoiper software.
Sip Trunks test between PBX	We will make a call to another PBX in this case to a classmate's PBX, what should happen is that we should receive and make calls to other extensions in other PBXs without problems and with stability.
Test connection to PSTN	We will make calls to any mobile outside of the extensions that we have created, in this case I will make both outbound and inbound calls to my own mobile, this will verify that the sip trunk routes are correctly configured to the outside.
IVR Test and Procedure	We will make a call from a mobile phone from abroad to our company and we will verify that the instructions and options created in our recording system are correct, then in different calls we will go to the derivation of the different departments and their extensions. We will dial 912911 to go to the finance department and we will check that the music on hold is correctly recorded with your waiting message. After this we will check that the holiday option of the time condition leads us to the corresponding announcement and therefore to the General Voicemail.
Call quality test	In each of the aforementioned calls we will go on to check the quality of these, that is to say that there are no delays, or audio losses etc.

Task 6

Create a drawing of the solution using Visio. Include all physical and virtual components



Submission requirements

You are required to submit the following as evidence for this assessment:

1. Written tasks should be completed on a word processor
2. Drawing tasks should be completed on Visio
3. You must click the submit button