REDCENTRIC UNITY IP VOICE SERVICE DEFINITION

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1. SERVICE OVERVIEW

Unity IP Voice is Redcentric's hosted enterprise voice solution for organisations needing to update their existing telephony system but who are also looking to control equipment, training and operational costs.

Unity IP Voice delivers:

- A carrier-grade IP telephony system with an extensive and innovative range of enterprise level user features
- Optional Unified Communications features including instant messaging, presence, video calling and desktop sharing
- Access to the service from desktop IP phones, desktop PCs, laptops and supported tablet PCs and smartphones
- A fully managed and hosted service with 24-hour technical support
- Call centre option with advanced call control and reporting
- Free calls between staff connected to Unity IP Voice
- Low setup costs with additional users & features charged on a pay-as-you-use basis
- Improved productivity for mobile staff by providing them with full access to all voice functionality regardless of where they are working even on external phones such as in a hotel
- A powerful but easy to use provisioning interface for the system administrator - no need for specialised in-house or contracted skills to manage the telephone system
- Project managed migration from an existing PBX environment on a per-user basis to maximise existing investment and avoid the need for a large, simultaneous upgrade
- Rapid recovery in the event of major IT or office facilities failure with the ability to quickly and seamlessly transfer all telephony services to a temporary office location
- A future-proof and highly scalable communications platform which can be easily integrated with your desktop computing environment

2. SERVICE DESCRIPTION

2.1 THE UNITY IP VOICE SWITCHES

The Unity IP Voice service is built on two centrally managed, software based, IP Voice Switches hosted in Redcentric's highly secure and resilient Data Centre environments.

This completely replaces the need for any Customer site PBX equipment and enables Customers to deploy IP Voice services with minimal upfront capital investment or in-house skills.

The Unity IP Voice Switch technology is based on SIP (Session Initiation Protocol) which provides a highly, scalable, cost effective and future proof platform for the provision of IP voice services.

Redcentric's Unity IP Voice platform has been deployed in a highly available configuration across Redcentric Data Centre locations in Reading and Harrogate. The core Unity IP Voice Switches are maintained and supported by Redcentric 24 / 7 and access to the Unity IP Voice Service is delivered across Redcentric's national IP Network (Managed IP-VPN) and via a range of connection technologies including ADSL, EFM, leased lines and 10Mbps, 100Mbps and 1Gbps Ethernet as required.

2.2 REDCENTRIC UK NETWORK – MANAGED IP-VPN

Unity IP Voice is provisioned on a dedicated, secure VPN across Redcentric's Managed IP-VPN MPLS network*. In order to assure call quality and IP Telephony performance, Redcentric have implemented Quality of Service (QoS) systems throughout the network.

A direct connection from the Customer site to the nearest Managed IP-VPN point of presence (POP) is required to provide service. These connections are available in bandwidths from 1Mbps to 1Gbps.

*Unity IP Voice may also be delivered across certain 3rd party Internet connections by using the "Unity Off-Net" service for Polycom phones or by using Unity UC software clients.

2.3 PSTN CONNECTIVITY

For external calls, Redcentric offers a low-cost PSTN breakout service directly from within its core network, which removes the need for PSTN lines to be installed at each office. This can be a very significant saving particularly when compared with the cost of installing ISDN30 lines. Redcentric's high speed PSTN interconnect enables IP callers to make and receive external calls, for example other providers' landline telephones, mobile telephones and international destinations.

2.4 AUDIO CODEC PREFERENCE

Unity IP Voice can be specified to prefer either the G.729a or the G.711 audio codec.

Where a Customer has G.729a preferred, all outbound calls made will attempt to negotiate in G.729a. This includes calls to both other sites within the organisation using the Unity service and external calls to the PSTN. All inbound calls from the PSTN will also attempt to negotiate in G.729a. Inbound calls from other sites within the organisation will attempt to negotiate based on the codec preference for the calling site.

Where a Customer has G.711 preferred, all outbound calls made will attempt to negotiate in G.711. This includes calls to both other sites within the organisation using the Unity service and external calls to the PSTN. All inbound calls from the PSTN will also attempt to negotiate in G.711. Inbound calls from other sites within the organisation will attempt to negotiate based on the codec preference for the calling site.

The G.729a codec uses approximately 50kbps of bandwidth per concurrent call (dependent on the network technology).

The G.711 codec uses approximately 100kbps of bandwidth per concurrent call (dependent on the network technology).

2.5 MOVES, ADDS AND CHANGES

A key feature of Unity IP Voice is it's easy to use and highly intuitive administration interface. Administration of traditional, PBX based systems require highly complex and time-consuming procedures often requiring specialist skills.

The Unity IP Voice secure web portal provides the Customer self-administration service management and enables end users to configure and manage their own services, significantly lowering the cost of Customer service and operations.

The portal is a hierarchical, web browser based, management interface which allows both system administrator and individual users to configure the Unity IP Voice Service to meet highly specific corporate practice and personal requirements.

Group Administrator

The Systems Administrator has access to a web-based management tool which gives them control over the Unity IP Voice service at a company-wide level.

End User

Viewed through a browser on the user's own PC, each individual Unity IP Voice user has their own restricted access to the management web portal. In addition to this, each user can choose to install the Unity basic desktop assistant, which is a small application installed onto the user's desktop providing easy and immediate control over all major features available to their licence type.

Example features include:

1. Click to dial from corporate, group or personal directories
2. Transfer a call to a colleague or make a 3-way call
3. Enable / Disable Call waiting, three way calling, do not disturb
4. Manage Voice mail options including delivery options and greeting messages
5. Set call diversion options
6. Quickly see call logs – missed calls, received calls, placed calls – and return any missed calls with a single click
7. Set remote working options, find me / follow me, redirect to alternative number e.g. mobile

2.6 EMERGENCY CALLS

Calls to the emergency services can be made using Unity IP Voice, subject to the Unity IP Voice service being available. Please see Section A of the Schedule to the MSA for further information about emergency calls.

Site location information is held by Redcentric, which defines the physical location address associated with each telephone number. This address information may be provided to the emergency services in the event that they are contacted to direct them to the source of the call. As such the accuracy of this information is critical. The Customer is responsible for notifying Redcentric of any change address where voice services are provisioned. Moving services to a new location without providing updated site location information is in breach of the General Condition 4 of the General conditions of Entitlement enforced by OFCOM.

2.7 DEPLOYMENT PROCESS

Redcentric's project managed migration to the Unity IP Voice service is a five-phase technical and business consultative process.

Phase 1: Technical Voice Audit

The objective of this audit is to document how the existing voice system is configured, and how any remote offices are inter-connected.

Phase 2: Technical Infrastructure Audit

The objective of this audit is to determine the readiness of the installed LAN to support voice. It looks at bandwidth capacity, Quality-of-Service, resilience, Power-over-Ethernet capability to support phones, and UPS backup in the event of a power failure.

Phase 3: Business Workshop

The objective of this workshop is to fully understand the Customer's telephony requirements, both now and in the future, and to design an IP telephony system to match them.

Phase 4: Training

Administrator training is provided at Redcentric's training facilities to provide an appointed Customer resource with the relevant skills to manage the Unity IP Voice Service at company level. In addition, end user training can be provided to help end users understand the Unity IP Voice Service and get the most from it.

Phase 5: Service Provisioning

Once the phone system has been installed, the installation team can remain on hand where necessary to ensure that any problems are quickly resolved. Where required, Redcentric may also provide on-site support staff for the first day to handle any queries or issues from end-users.

2.8 NUMBER PORTING

The porting of existing telephone number ranges, where possible, forms part of Redcentric's service delivery process. Redcentric currently has porting agreements with BT and Vodafone (formerly Cable and Wireless) that facilitate direct porting of their number ranges onto our service. This means the Customer can retain their existing inbound numbers as part of the Unity IP Voice service. If a Customer has previously ported numbers to another service provider, Redcentric need to confirm that the relevant commercial agreement is in place with that service provider as well as the OFCOM nominated range holder. Redcentric currently have these agreements with BT, Vodafone (Cable and Wireless, Thus and Energis).

Redcentric presales consultants will gather the necessary information to complete the number port but will need the Customer's assistance to identify the current Service Provider and Range Holder of the numbers, the site information including postcode and any associated direct dial inwards (DDI) numbers attached to main billing number. This information will then be included in the management summary of the contract so both parties have full visibility of the porting scope. Redcentric will in addition, request the Customer provides on letter headed paper a standard letter of authority (LOA). The LOA together with the porting request form allows Redcentric's porting desk to talk directly to the losing operator's helpdesk on any number discrepancies.

Once it has been established that a Customer's numbers can be ported, Redcentric will complete the relevant industry documentation and submit the porting request to the losing operator (and range holder where different) with the letter of authority. Porting lead times are regulated and are a critical milestone in the overall delivery of the Unity IP Voice Service. It is the Customer's responsibility to provide accurate data. Inaccurate data can result in the porting request being rejected. Once the porting order is accepted by the losing operator, the lead time commences between the range holder and gaining service provider. This typically takes 22 working days.

The Redcentric project manager assigned to the delivery will advise on the delivery date once confirmation is received back from the losing operator, who ultimately controls the date that the port will happen. Until the porting date confirmation is received back from the range holder and service provider any dates quoted will be on an indicative basis.

The installation price a Customer is quoted includes the number porting as long as the port is scheduled to take place between Monday to Friday and between the hours of 9 am and 4 pm. Any porting requests outside of the stated hours will incur additional out of hours costs. Bank and public holidays are treated as an out of hours.

On the day of the port there will be a minor disruption to service whilst the numbers move between networks. It's not possible to provide an exact time for the port, but once requested it will happen within a 3-hour window. I.e. if a port is request at 3pm, it will happen by 6pm on that day. In the unlikely event of problems Redcentric are able to invoke an emergency restoration within 24 hour which will see the numbers returned to the original provider and service is restored to its former state. The restoration can take up to 1 hour.

Porting of numbers is subject to paragraph 6.7 of Schedule 2 to the MSA.

2.9 BT DIRECTORY ENQUIRIES

New telephone numbers provided as part of the Unity IP Voice service are not as standard published into BT Directory Enquiries. Numbers ported to the Unity IP Voice service from other providers are adopted as they currently exist.

2.10 SERVICE COMPONENTS

Office Connectivity

Unity IP Voice is delivered as a service over a Redcentric connection. If a Managed IP-VPN connection is already installed, it's usually possible to supply voice over the existing connection without significant modification provided that there is sufficient bandwidth capacity (see below for more details). Managed IP-VPN connections are available for small and home offices as an ADSL connection, through to PPC and Ethernet connections for larger offices.

Unity IP Voice can also be delivered over certain 3rd party Internet connections using the Unity Off-Net service.

2.11 TECHNICAL DATA

Unity IP Voice uses the G.729a codec which consumes approximately 50kb/s per call or the G.711 codec which consumes approximately 100kb/s per call, depending on the network technology being used. This is how voice can be reliably delivered over Redcentric's managed ADSL connections.

Unity IP Voice is not supported over bonded xDSL or rate adaptive ADSL Max / 2+ services.

Unity IP Voice across Managed IP-VPN

Each Customer is provisioned with a private voice network over Managed IP-VPN. This ensures business grade call quality between sites. Managed IP-VPN is Redcentric's dedicated 10Gbps national MPLS network and is specifically designed to carry high volumes of both voice and data traffic without causing degradation to the voice service.

With the Managed IP-VPN connection, Redcentric provides a fully managed multi-service router for each Customer site to shape traffic and to ensure that voice traffic is not negatively impacted by other traffic (such as Internet traffic) also using the Managed IP-VPN connection.

Redcentric ADSL circuits are installed with a 20:1 contention ratio. Quality of Service (QoS) over DSL and LES/WES connections is implemented using ingress and egress traffic control at both ends of the Managed IP-VPN connection.

2.12 UNITY IP VOICE ACROSS THE INTERNET

Unity IP Voice can be delivered across certain Internet connections using the Unity Off-Net service. Unity UC software clients also work via the Internet.

2.13 UNITY IP VOICE BANDWIDTH REQUIREMENTS

If all available voice channels to a site are in use, the Unity IP Voice system will prevent an additional call being setup to protect the other active calls from call degradation. An incoming call which is unable to complete because the circuit is congested will be treated as if the person they were calling was on the phone. This may include for example, diverting the call to voice mail, placing it in a queue, or diverting it to someone else. A caller on hold does not consume a channel, it is only when the call is active (i.e. a two-way conversation) that a channel is needed.

Note that a call made between two phones on the same site does not consume a channel because once the call is established the voice path is routed directly between the two phones and not over the Managed IP-VPN circuit.

Notes:

1. You need to determine how many concurrent phone calls the site needs at any one time (number of channels). As a guideline, a normal office would need one channel – equivalent to a phone line - per 5 employees. However, some offices such as call centres require one channel per employee because of the high call demand. A good starting point is to look at how many phone lines or ISDN channels are already installed as a channel is equivalent to a phone line. The following table is provided as a rough guide:

Managed IP-VPN Circuit (and applied bandwidth)	Voice Channels (G.729a)	Voice Channels (G.711)
ADSL 1Mb/s	5	2*
ADSL 2Mb/s	5	2*
FTTC	15 (recommended)	15 (recommended)
EFM 2Mb/s	40	20
Ethernet 10Mb/s	200	100
Ethernet 100Mb/s	2000	1000

*2 channels if using a dedicated ADSL circuit, 1 channel if supporting voice and another service (e.g. Internet) across the same ADSL circuit.

2. It is important to also consider which other services will be using the connection in addition to voice such as Internet access, VPN, and Server Backup. These all require bandwidth and need to be included in the decision of which circuit is necessary.
3. If voice is being provisioned over a managed Redcentric ADSL circuit, the circuit can only support one additional service (i.e. Internet, Backup or VPN) in addition to voice. If three or more services are necessary, then an Ethernet (LES / WES) circuit will be required.
4. Voice cannot be provisioned over a Redcentric Bonded ADSL/SDSL connection. It is not supported over such connections.
5. Voice services are not supported over ADSL services other than for ADSL 1000 and ADSL 2000 fixed profile. On other ADSL services, the connection is subject to Dynamic Line Management (DLM) which can cause intermittent service interruption. Please see Managed IP-VPN service definitions for more detail.
6. Whilst FTTC services still rely on DLM, the effects are likely to be much lower than on ADSL services. As such FTTC service may still be subject to occasional intermittent service interruptions.
7. Total number of channels available on FTTC is determined by the sync speed achieved on the line following installation. This is established at point of installation. The exact number of channels that will be available over FTTC cannot established beforehand.
8. Redcentric strongly recommend that FTTC services be used to deliver a maximum of 15 channels. This is due to the SLA available on FTTC and the impact of a service outage. For more than 15 channels or for sites where the FTTC SLA is not satisfactory, the customer should use an EFM or Ethernet service which are delivered with a superior SLA.

Video Calling

Customers may optionally enable video calling to and from their Polycom VVX 501 and VVX 601 IP phones. Please note that this will require additional bandwidth on their voice network. In order to make a video call from a VVX 501 or VVX 601, customers will also need to purchase an optional Polycom USB camera. Video calling to / from Polycom VVX phones is disabled as default and a request to support must be made in order to enable it.

Please note that video calls consume approximately 1Mbps of bandwidth per call across the network.

Where video calling is being deployed, the Customer should always engage their Redcentric presales consultant to ensure that sufficient network capacity is provisioned.

2.14 PHONES / HANDSETS

Overview

The Customer will use phones provided by Redcentric in connection with the Unity IP Voice Service. Unless otherwise agreed other related equipment, e.g. headsets, are not provided.

Redcentric use select models from the Polycom VVX range of desktop phones and SoundStation IP range of conference phones. The phones may either be purchased or rented by the Customer and are either supplied with the installation of the Unity IP Voice Service or may be sent to the Customer site for self-installation.

Please note that when purchased, phones, cables and related accessories become the property of the Customer and as such the Customer is responsible for their on-going maintenance. They remain the ownership of the Customer at the end of the contract.

Unless otherwise stated, phones do not come with mains power supplies. If you are not intending on using 802.3af compliant Power over Ethernet, you will require power supplies which are available for an additional fee.

Please note: the terms of phone rental differ to that of purchasing phones. Phone rental is detailed in the Unity Phone Rental service definition.

The following phones are supported for use with the Unity IP Voice service:

 Polycom VVX 101	• Single line • Graphical LCD (132 x 64) resolution • 10/100 Ethernet port • 2 line keys with bi-colour (red/green) LED • 4 context-sensitive “soft” keys • Integrated IEEE 802.3af Power over Ethernet support • 4-way navigation key cluster with centre “select” key • Dedicated RJ-9 headset port

 Polycom VVX 201	• Two-Line Backlit LCD • Two 10/100 Ethernet ports (132 x64) resolution • • Two 10/100 Ethernet ports • 2 line keys with bi-colour (red/green) LED • 4 context-sensitive "soft" keys • Integrated IEEE 802.3af Power over Ethernet support • 4-way navigation key cluster with center "select" key • Dedicated RJ-9 headset port
 Polycom VVX 301	• 6 lines • 4-way navigation cluster with centre "select" key • Backlit grayscale graphical LCD (208 x 104) • Full-duplex speakerphone • Integrated IEEE 802.3af Power over Ethernet support • 2 port 10/100 Ethernet switch • Dedicated RJ9 headset port
 Polycom VVX 311	• 6 lines • 4-way navigation cluster with centre "select" key • Backlit grayscale graphical LCD (208 x 104) • Full-duplex speakerphone • Integrated IEEE 802.3af Power over Ethernet support • 2 port Gigabit Ethernet switch • Dedicated RJ9 headset port
 Polycom VVX 401	• 12 lines • 4-way navigation cluster with centre "select" key • 3.5" colour TFT display • Full-duplex speakerphone • Integrated IEEE 802.3af Power over Ethernet support • 2 port 10/100 Ethernet switch • Dedicated RJ9 headset port

 Polycom VVX 411	• 12 lines • 4-way navigation cluster with centre “select” key • 3.5” colour TFT display • Full-duplex speakerphone • Integrated IEEE 802.3af Power over Ethernet support • 2 port Gigabit Ethernet switch • Dedicated RJ9 headset port
 Polycom VVX 501	• 12 lines • Touch screen operation • 3.5” (9-cm) TFT (320 x 240) colour display • Full duplex speakerphone • Two port Gigabit Ethernet switch • Future support for optional USB video camera • Integrated IEEE 802.3af PoE support • Dedicated RJ9 headset port
 Polycom VVX 601	• 16 lines • Touch screen operation • 4.3” TFT (480 x 272) capacitive touch-screen • Full duplex speakerphone • Two port Gigabit Ethernet switch • Future support for optional USB video camera • Integrated IEEE 802.3af PoE support • Dedicated RJ9 headset port
 Polycom Soundstation IP 5000	• Advanced IP conference phone • Designed for executive offices and small conference rooms with up to 6 participants • 7 foot microphone pickup • Backlit 320 x 160 pixel graphical greyscale LCD • Integrated IEEE 802.3af PoE support • Note: Does not include power supply as standard

 Polycom Soundstation IP 6000	• Ideal for small- to medium-sized conference rooms • 12 feet of microphone pickup which can be expanded with optional extension microphones • High resolution pixel backlit graphical LCD • Integrated IEEE 802.3af PoE support • Includes 240v mains power supply
 Polycom Soundstation IP 7000	• Ideal for medium-sized conference rooms • 20 feet of microphone pickup which can be expanded with optional extension microphones • Large, high resolution pixel backlit graphical LCD • Integrated IEEE 802.3af PoE support • Includes 240v mains power supply
 "Polycom Trio" 8800	• Includes 240v mains power supply • Gesture-based, multitouch-capable capacitive touch screen • 5-inch colour LCD (720 x 1280 pixel),9:16 aspect ratio • Two-port gigabit Ethernet switch –10/100/1000Base-TX across LAN and 2nd port. 2nd port supports IEEE 802.3af PSE • 802.11 a/b/g/n (Wi-Fi) network connectivity • Integrated Bluetooth 4.0 and NFC

Polycom VVX Branding

As standard, Polycom VVX phones provided with the Unity service are branded with a Redcentric logo. This comprises of a screensaver on VVX 301, VVX 311, VVX 401 and VVX 411 models and an idle image on VVX 501 and VVX 601 models.

Customers may request that their Polycom VVX phones are de-branded, back to the factory default. Customers may also, in addition to the de-branding, request that their Polycom VVX phones are branded with their own custom logo.

The screensaver appears after the phone has been idle for a default period 5 minutes. The value for this period can be changed from 1 to 9999 minutes. It can also be disable on a per user basis.

The idle image appears when the phone is idle. It can be disabled on a per user basis.

Custom logo images must conform to the following specifications. The Customer is responsible for providing Redcentric with their images in the correct format.

Image Requirements

Image requirements for custom-branded logos:

Phone Model	Image Requirements
VVX 301	Screensaver, 208x104 pixels, JPEG or PNG
VVX 311	Screensaver, 208x104 pixels, JPEG or PNG
VVX 401	Screensaver, 320x240 pixels, JPEG or PNG
VVX 411	Screensaver, 320x240 pixels, JPEG or PNG
VVX 501	Idle image, 320x158 pixels, JPEG or PNG
VVX 601	Idle image, 480x190 pixels, JPEG or PNG

Please note that the Polycom VVX phones do not support progressive or multi-scan JPEG images.

2.15 CORDLESS (DECT) HANDSETS

Overview

Redcentric offer the Panasonic TGP600 cordless DECT IP phone for use with the Unity service. The phones may either be purchased or rented by the Customer and are either supplied with the installation of the Unity IP Voice Service or may be sent to the Customer site for self-installation.

Please note that when purchased, phones, batteries and related accessories become the property of the Customer and as such the Customer is responsible for their on-going maintenance. They remain the ownership of the Customer at the end of the contract.

Panasonic KX-TGP600

- Cordless DECT base station and 1 x handset bundle
- Up to 8 handsets per base station
- Up to 8 simultaneous calls per base station
- DECT radio technology
- Base station and each cordless handset docking station requires mains power supply

Additional handsets can also be purchased and registered with the base stations (up to a total maximum of 6 handsets):

 Panasonic KX-TPA60	• 1.8" colour LCD with backlight • Soft keys • 500 phone book address/20 last dialled number memory

Feature Support

The DECT phones support the following features:

- Transfer, blind
- Transfer, consulted
- Call Forwarding
- Do not Disturb
- Hold
- Conference
- Anonymous Call Rejection
- Block Caller ID

2.16 HEADSETS

Redcentric offers a range of wired, wireless and USB headsets. Wired and wireless headsets may be used with supporting Polycom phones. USB headsets may be used with a compatible PC using the Unity UC client for Windows.

Please note that headsets become the property of the Customer and as such the Customer is responsible for their on-going maintenance. They remain the ownership of the Customer at the end of the contract.

Headsets that develop a fault with in the manufacturer’s standard warranty may be returned to Redcentric in exchange for a replacement. This excludes headsets that have been damaged through misuse, through accidental damage or through operation outside of the manufacturer’s recommended operating conditions.

Consumables including cables, ear pads, and microphone tubes cords are not included in the hardware replacement policy. Headsets may be purchased only and are not available to rent.

Wired Headsets

Wired headsets for use with compatible and supported Polycom VVX and SoundPoint IP phones.

 **Plantronics SupraPlus H251A**	• Over-the head monaural design for maximum comfort. • Volume and mute audio controls at your fingertips
 **Plantronics SupraPlus H251N**	• Over-the head monaural design for maximum comfort. • Volume and mute audio controls at your fingertips • Noise-cancelling microphone filters out background noise.
 **Plantronics SupraPlus H261N**	• Over-the head Binaural design for maximum comfort. • Volume and mute audio controls at your fingertips • Noise-cancelling microphone filters out background noise.

Wireless Headsets

Wireless headsets for use with compatible Polycom VVX IP phones. Please note that use of the Electronic Hookswitch (EHS) feature available on wireless headsets is subject to support by the model of phone, and the phone's software version. Please consult Redcentric's support team for more details.

 **Plantronics CS540A**	• Convertible design – over the ear or headband • Wireless range of up to 120 metres. • Compatible with Plantronics Electronic Hookswitch (EHS) cable to remotely answer and terminate calls. • Noise-cancelling microphone. • Internal rechargeable battery providing up to 7 hours of wireless talk time. • Lightest DECT headset at just 21g
 **Plantronics CS510A**	• Over-the head Monaural design. • Wireless range of up to 120 metres. • Compatible with Plantronics Electronic Hookswitch (EHS) cable to remotely answer and terminate calls. • Volume and mute audio controls. • Noise-cancelling microphone. • Internal rechargeable battery providing up to 9 hours of wireless talk time.
 **Plantronics Savi W740-M**	• Convertible between multiple designs including over-the-ear, over-the-head and neckband design. • Wireless range of up to 120 metres. • Compatible with Plantronics Electronic Hookswitch (EHS) cable to remotely answer and terminate calls. • Volume and mute audio controls. • Noise-cancelling microphone. • Internal rechargeable battery providing up to 9 hours of wireless talk time • Anti-static shock protection • Strong and robust construction • Can also connect to supported mobile phones via Bluetooth. • USB connectivity for connection to a compatible PC.

USB Headsets

 Plantronics Blackwire C310	• Over-the head single earpiece design for maximum comfort. • Simple plug-and-play USB connectivity. • Noise cancelling microphone. • Intuitive inline controls to answer/end calls, control volume, and mute
 Plantronics Blackwire C510	• Over-the head single earpiece design for maximum comfort. • Simple plug-and-play USB connectivity. • Noise cancelling microphone. • Intuitive inline controls to answer/end calls, control volume, and mute • Ultra-soft leatherette ear cushion and lightweight headband mean all-day comfort
 Plantronics Blackwire C720	• Over-the head double earpiece design for maximum comfort. • Simple plug-and-play USB connectivity. • Noise cancelling microphone. • Volume and mute audio controls. • USB and Bluetooth connectivity • Ultra-soft leatherette ear cushions and lightweight headband mean all-day comfort

Bluetooth Earpiece

 Plantronics Voyager Edge	• Slim, sophisticated design for a comfortable fit • Signature Plantronics audio technology eliminates disruptive background noise and keeps calls clear • Charging case enables up to 16 hours talk time (up to 6 hours without case) • Automatically updates Lync status to available, away & busy using responsive sensors

Electronic Hook-switch Compatibility

Some wireless headsets enable you to answer and end phone calls using controls located on the headset. This feature is called Electronic Hookswitch (EHS). EHS functionality is only available with the Plantronics wireless range of headsets and is not with the Blackwire USB headsets or the H251 / H251N wired headsets.

The table below shows EHS compatibility for wireless headsets that Redcentric offers with the associated Unity service, and any adapters that may be required as part of this functionality. The Plantronics headsets that Redcentric provides are only supported with the range of Polycom SoundPoint phones outlined in the following table and are not supported with any other Polycom devices.

Polycom / Plantronics Accessory Matrix

The following table shows what connectors are required in order for each model of Polycom phone to work with each respective headset.

Handset	Plantronics					
	CS500 Range	**Savi 700 Range**	**Supraplus mon voicetube HW251/A**	**Supraplus bi voicetube HW261/A**	**Supraplus W/band mon N/cancel HW251N**	**Supraplus W/band bi N/cancel HW261N/A**
SoundPoint IP						
IP321	APP51 & Polycom adaptor	APP51 & Polycom adaptor	QD to 2.5	QD to 2.5	QD to 2.5	QD to 2.5
IP331	APP51 & Polycom adaptor	APP51 & Polycom adaptor	QD to 2.5	QD to 2.5	QD to 2.5	QD to 2.5
IP450	APP51	APP51	U10P	U10P	U10P	U10P
IP550	APP51	APP51	U10P	U10P	U10P	U10P
IP560	APP51	APP51	U10P	U10P	U10P	U10P
IP650	APP51	APP51	U10P	U10P	U10P	U10P
IP670	APP51	APP51	U10P	U10P	U10P	U10P
VVX						
VVX300	APP51	APP51	U10P	U10P	U10P	U10P
VVX310	APP51	APP51	U10P	U10P	U10P	U10P
VVX400	APP51	APP51	U10P	U10P	U10P	U10P
VVX410	APP51	APP51	U10P	U10P	U10P	U10P
VVX500	APP51	APP51	U10P	U10P	U10P	U10P
VVX600	APP51	APP51	U10P	U10P	U10P	U10P

Product	Manufacturer Partcode
APP51 Cable	38439-11
Polycom Headset Interface Adaptor	2200-11095-002
QD to 2.5	70765-01
U10P Connection lead	32145-01

*Please note that the Electronic Hookswitch Adapter (APP-51 Cable) is required if the Electronic Hookswitch functionality is required. Other connectors are required. Electronic Hookswitch feature is also subject to a minimum version of software on your Polycom phones. Please consult Redcentric's support team for details.

Headset Warranty

Headsets that develop a fault within the manufacturer's warranty period may be returned to Redcentric for a replacement.

This excludes headsets damaged through misuse, accidental damage, or through operation outside of the manufacturer's recommended operating conditions or if used in conjunction with a non-manufacturer approved device.

Headsets and headset accessories provided by Redcentric automatically become property of the Customer upon delivery and the Customer becomes responsible for the ongoing maintenance of this equipment and any failures that occur outside of the manufacturer's provided warranty.

We can offer some support with the Bluetooth headset when using Unity, however we cannot offer support when using other applications such as Skype.

2.17 ADAPTERS FOR ANALOGUE PHONES (IADS)

Overview

Redcentric can provide converters that allow standard analogue phones to connect to the Unity IP Voice service.

Note that these adapters are not suitable for Digital PBX handsets and can only be used with analogue phones.

Typical examples of where these adapters may be used include analogue wired phones and analogue line DECT phones.

2 port adapters are supplied with installation and initial configuration. However please note that they become the property of the Customer owns and as such the Customer is responsible for their on-going maintenance. They remain the ownership of the Customer at the end of the contract.

Note: The use of analogue telephone adapters for fax machines, modems or other analogue data devices is not supported.

2.18 UNITY UC SOFTWARE CLIENT

The Unity software client allows users to access the Unity service from a range of Internet-connected devices.

The Unity UC client provides users with access to a range of additional features as well as basic telephony functionality.

The client is available across a number of different device types. The following matrix shows which features are available on which device:

Feature	Windows desktop	Apple desktop	iPhone	Android phone	Android tablet
Contact directory	Yes	Yes	Yes	Yes	Yes
Unity call control settings	Yes	Yes	Yes	Yes	Yes
Presence	Yes	Yes	Yes	Yes	Yes
Instant messaging	Yes	Yes	Yes	Yes	Yes
Group chat	Yes	Yes	Yes	Yes	Yes
Voice calling	Yes	Yes	Yes	Yes	Yes
N-way voice calling	Yes	Yes	Yes	Yes	Yes
Video calling	Yes	Yes	Yes	Yes	Yes
Call & message history	Yes	Yes	Yes	Yes	Yes
My room	Yes	Yes	Yes	Yes	Yes
Desktop sharing	Yes	Yes	No	No	No

Please note that presence, instant messaging, group chat, video calling, my room and desktop sharing are only available to users assigned the Unity Collaborate service pack.

The Unity UC client requires a suitable Internet connection. It registers and connects with Redcentric's voice platforms via the Internet and is therefore subject to the same limitations as the Unity off-net service. Please see limitations section for details.

Unity UC client bandwidth requirements

The Unity client is intended to work over wireless or wired Internet access. Please note that the quality of the experience when using the Unity UC service is dependent on the availability of bandwidth and quality of the Internet connection available.

Please note that whilst the Unity UC service may work over mobile data networks, it is the customer's responsibility for ensuring that any such use of the service is permitted within the terms and conditions of their mobile provider. The customer is also responsible for any data charges incurred through such use of the service via mobile data networks.

Bandwidth requirements for the various Unity features are as follows:

- HD voice call 100kbps (G.722)
- Standard voice call 100kbps (G.711)
- Compressed voice call 50kbps (G.729)
- Video call including voice 1Mbps
- Desktop sharing session average 70 kbps, up to 500 kbps

Security

The Unity UC client allows users to access and make use of features of the Unity service from their mobile device. It also provides remote access to corporate telephone directories and the ability to initiate business telephone calls. As such, it should be ensured that the use of the application does not contravene any corporate security or compliancy rules or guidelines. It should also be ensured that users using the application use the timed lock and security PIN code feature on their mobile devices to prevent unauthorised access to the Unity service in the event of loss or theft.

2.19 UNITY BASIC DESKTOP ASSISTANT OVERVIEW

Unity desktop Assistant is a software application that allows Unity users to control their telephone service settings from their Windows desktop device. It works via a local wired/wireless Internet connection.

It is downloaded via the Redcentric downloads page.

Users wishing to use the application must be licensed in order to do so. This licence is applied as an additional service pack per user on top of their main service pack(s). Users attempting to user the application without the appropriate licence will receive an error message. Users will require their Unity username and password in order to sign in to the application.

Please note that functionality available via the application is subject to the user's assigned Service Pack and associated features.

Call Control Functionality

The application allows the user to control the following features of their Unity service:

- **Do Not Disturb** - Can be switched on or off. Ring Reminder plays a brief ring splash when an inbound call comes in to remind you that you have Do Not Disturb active.

- **Remote Office** - Can be switched on or off and the target telephone number can be changed. This feature must be active, and the telephone number must be that of your mobile phone to use the call back feature of the Unity Call Control app.
- **Simultaneous Ring Personal** - Can be switched on off. The called numbers can also be specified using the app.
- **Call Forwarding Always** - Can be switched on or off and the target telephone number can be specified. Ring Reminder plays a brief ring splash when an inbound call comes in to remind you that you have Call Forwarding Always active.
- **Call Forwarding No Answer** - Can be switched on or off and the target telephone number can be specified. You can also configure the number of rings after which the call is diverted to the target telephone number.
- **Call Forwarding Busy** - Can be switched on or off and the target telephone number can be specified.
- **Call Forwarding Busy** - Can be switched on or off and the target telephone number can be specified.
- **Calling Line ID Delivery Blocking** - Can be switched on or off
- **Voicemail/Unified Messaging** – Forward to email address, notification of new messages, rings before Voicemail.

Contact Directory Access

By pressing the Contacts tab, users are presented with the ability to view contacts on their local phone, on their personal Unity directory and on the enterprise Unity directory.

Call Logs

The app provides history of the latest 50 placed, received and missed calls for the user's Unity account. You can switch between All Calls or Missed Calls only. Missed Calls are displaced in red.

Making a Call (Call Back)

The app can be used to make a call, either by using the keypad, or by clicking on a contact via the Contacts tab. In order to do this, the user will require a Service Pack that contains the Remote Office feature. The Remote Office feature will also need to be active and have the telephone number of the user's mobile phone configured at the Remote Office phone number. The call then initiates a call-back from the Unity service to the user's mobile phone. When the user answers this call (as a call received to their mobile, appearing from their Unity DDI), the Unity service will then ring the called party. The called party will be presented with the user's Unity DDI, not their mobile telephone number.

It should be noted that using the Call Back service will incur a fixed to mobile charge for the leg of the call to the user's mobile phone. The second leg of the call from the Unity service to the called party may also be chargeable subject to the called destination.

Security

The Unity Call Control app allows users to access and make use of features of the Unity service from their mobile device. It also provides remote access to corporate telephone directories and the ability to initiate business telephone calls. As such, it should be ensured that the use of the application does not contravene any corporate security or compliancy rules or guidelines. It should also be ensured that users using the application use the timed lock and security PIN code feature on their mobile devices to prevent unauthorised access to the Unity service in the event of loss or theft.

2.20 USER SERVICE PACKS (USER LICENCES)

Unity users are provided functionality by assigning them with a service pack. This gives users a billing account and a phone number on the Unity system.

There is a range of core service packs designed around varying roles within an organisation. These are defined below.

Please note that phones / handsets are not included as part of the service pack and need to be ordered separately.

Standard	An entry level service pack providing general telephony functionality. Includes voice-mail and PC and Mac desktop integration.
Enterprise	For staff that need to work both in the office and remotely, such as working from home. Also includes voice-mail and PC and Mac desktop integration.
Collaborate	For staff requiring full telephony and unified communications features across desktop and mobile devices.
Receptionist (Enterprise Edition)	Advanced functionality for front office staff that need to handle incoming calls to the organisation. Includes the Receptionist application.
Receptionist (Office Edition)	A reduced-functionality version of Receptionist aimed at smaller offices. See Receptionist comparison table in the User Features section of this document for details.

As well as being provided with a service pack, users who need to be part of a Call Centre require additional licences. These licences are assigned to the user as an overlay, in addition to their core service pack.

Call Centre Standard (User Overlay)	This user overlay must be assigned to all users who need to be part of a Call Centre. This allows users to be a member of a single or multiple Call Centres. Please note that the Call Centre / Call Queue service is referred to as "Call Centre Standard (Group Service)".
Call Centre Agent (User Overlay)	This user overlay adds advanced Call Centre functionality to users, driven through a PC desktop application. It allows visibility and desktop control of call handling, agent status, supervisor escalations and access to group and enterprise contact directories. Call Centre Agents must have Call Centre Standard (User Overlay) as a prerequisite. Call Centre Agents may have any of the core service packs.

Call Centre Supervisor (User Overlay)	This user overlay adds advanced Call Centre Supervisor functionality to users, driven through a PC desktop application. It provides visibility and management of Call Centres (queues), visibility of agent status, barge-in and silent monitoring features and access to a range of real time and historic reports on queue and agent performance. Call Centre Supervisors must have Call Centre Standard (User Overlay) as a prerequisite. Call Centre Supervisors must have Office User or Mobile Worker as their core service pack.

Users who wish to be both a Supervisor but also an active Agent within the Call Centre(s) may be assigned both the Call Centre Agent (User Overlay) and Call Centre Supervisor (User Overlay) licences.

There are also two types of licences for phones that are not assigned to specific users. These are for public area type phones such as hot-desk areas and meeting rooms.

Please note that phones / handsets are not included as part of the service pack and need to be ordered separately.

Hot Desk Phone	For phones in hot-desk areas. Staff that have been assigned "Mobile Worker", "Hot Desk Receptionist" or "Hot Desk Receptionist – Office Edition" licences can log into a hot-desk phone and use it as their own phone. When logged in, calls to the user's DDI or extension number are routed to the associated hot-desk phone. When they make calls from the hot-desk phone, their name / DDI (where available) is presented. The message waiting indicator on the hot desk phone also flashes when the associated user has a new voice message waiting. When a user is not associated with the hot-desk phone, it may still be used to make and receive calls. When used in this disassociated status, the phone will use its own underlying DDI / name that it can be called using, and that will be presented (where available) on outbound calls made from the phone. Users associate and disassociate with the hot-desk phone through their voice portal from the desired hot-desk phone or via the user level web admin portal. Note: when a hot-desk phone is associated with a user (a user is logged in on the phone), there is no visible notification on the phone's screen. The extension number visible on the phone's screen remains that of the underlying hot-desk phone regardless of any user association.
Meeting Room Phone	General purpose licence for public area phones with minimal functionality. Intended for meeting rooms and conference rooms.

There is also the following 'virtual' user licence, which is not assigned to a specific person or a phone. Note that this licence is not intended to be used with a phone.

Call Park Station	Call Park Station allows a user to transfer a call to a 'call parking bay' where it is placed on hold. It can then be retrieved by another user within the group by simply calling the extension number of the Call Park Station.

USER FEATURES

The following table indicates the individual service features included in each user account. Service features are not available on their own or individually outside of a user account.

User Features	Standard	Enterprise	Collaborate	Receptionist Enterprise Edition	Receptionist Office Edition	Call Centre Standard (User Overlay)	Call Centre Premium (User Overlay)	Call Centre C Agent (User Overlay)	Call Centre Supervisor (User Overlay)	Hot Desk Phone	Meeting Room	Call Park Station
Aliases	✓	✓	✓	✓	✓							
Alternate Numbers		✓	✓	✓	✓							
Anonymous Call Rejection	✓	✓	✓	✓	✓							
Authentication	✓	✓	✓	✓	✓					✓	✓	✓
Automatic Callback	✓	✓	✓	✓	✓							
Automatic Hold / Retrieve												✓
Barge-in Exempt		✓	✓	✓	✓							
Basic Call Logs	✓	✓	✓	✓	✓					✓	✓	
Busy Lamp Field		✓	✓	✓	✓							
Call Centre Standard						✓						
Call Centre Premium							✓					
Call Centre Agent Functionality								✓				
Call Centre Supervisor Functionality									✓			
Call Forwarding Always	✓	✓	✓	✓	✓							
Call Forwarding Busy	✓	✓	✓	✓	✓							
Call Forwarding No Answer	✓	✓	✓	✓	✓							
Call Forwarding Selective		✓	✓	✓	✓							
Call Notify		✓	✓	✓	✓							
Call Return	✓	✓	✓	✓	✓							
Call Transfer	✓	✓	✓	✓	✓					✓	✓	
Call Waiting	✓	✓	✓	✓	✓					✓	✓	
Calling Line ID Blocking Override	✓	✓	✓	✓	✓					✓	✓	
Calling Line ID Delivery Blocking	✓	✓	✓	✓	✓					✓	✓	
Calling Name Retrieval	✓	✓	✓	✓	✓					✓	✓	
Client Call Control	✓	✓	✓	✓	✓					✓	✓	
CommPilot Express		✓	✓	✓	✓							
Connected Line Identification Presentation	✓	✓	✓	✓	✓					✓	✓	

User Features	Standard	Enterprise	Collaborate	Receptionist Enterprise Edition	Receptionist Office Edition	Call Centre Standard (User Overlay)	Call Centre Premium (User Overlay)	Call Centre C Agent (User Overlay)	Call Centre Supervisor (User Overlay)	Hot Desk Phone	Meeting Room	Call Park Station
Connected Line Identification Restriction	✓	✓	✓	✓	✓					✓	✓	
Desktop Sharing			✓									
Directed Call Pickup		✓	✓	✓	✓							
Directed Call Pickup with Barge-in		✓	✓	✓	✓							
Distribution Lists	✓	✓	✓	✓	✓							
Do Not Disturb	✓	✓	✓	✓	✓							
External Calling Line ID Delivery	✓	✓	✓	✓	✓					✓	✓	
Greetings	✓	✓	✓	✓	✓							
Hoteling Guest	✓	✓	✓	✓	✓							
Hoteling Host										✓		
Instant Messaging			✓									
Internal Calling Line ID Delivery	✓	✓	✓	✓	✓					✓	✓	
Last Number Redial	✓	✓	✓	✓	✓						✓	
Line ID Blocking												
Multiple Call Arrangement		✓	✓	✓	✓							
Outlook Integration		✓	✓	✓	✓							
Pre-alerting Announcement		✓	✓	✓	✓							
Personal Phone List												
Presence			✓									
Remote Office		✓	✓									
Selective Call Acceptance		✓	✓	✓	✓							
Selective Call Rejection		✓	✓	✓	✓							
Sequential Ring		✓	✓	✓	✓							
Shared Call Appearance		✓	✓	✓	✓							
Simultaneous Ring		✓	✓	✓	✓							
Speed Dial 8	✓	✓	✓	✓	✓							
Speed Dial 100	✓	✓	✓	✓	✓							
Three-Way Call	✓	✓	✓	✓	✓						✓	
Unity Call Centre Web App								✓	✓			
Unity Software Client – Desktop (Call Control Only)	✓	✓										

User Features	Standard	Enterprise	Collaborate	Receptionist Enterprise Edition	Receptionist Office Edition	Call Centre Standard (User Overlay)	Call Centre Premium (User Overlay)	Call Centre C Agent (User Overlay)	Call Centre Supervisor (User Overlay)	Hot Desk Phone	Meeting Room	Call Park Station
Unity Software Client – Desktop (Call Control, Voice, Video)			✓									
Unity Software Client – Mobile (Call Control, Voice, Video)			✓									
Unity Receptionist Application [1]				✓	✓							
Video Calling (Unity client to Unity client)			✓									
Voicemail Management	✓	✓	✓	✓	✓							
Voicemail Portal	✓	✓		✓	✓							

[1] Available functionality within the Unity Receptionist Application is based on the Receptionist licence type assigned to the user logged in. A comparison of features between the variants of Receptionist licence is defined in the following table.

2.21 RECEPTION USER ACCOUNTS

The below table shows the difference between the types of Receptionist user account:

User Features	Receptionist	Receptionist Office Edition
Call Control		
Call Control – Dial Other	✓	✓
Call Control – Accept	✓	✓
Call Control – Hold	✓	✓
Call Control – End	✓	✓
Call Control – Blind Transfer	✓	✓
Call Control – Consult Transfer	✓	✓
Call Control – 3 Way Call Conferencing	✓	✓
Call Control – Voice Mail	✓	✓
Handle Multiple Call Simultaneously	✓	✓
Last Redirected Support	✓	
Directed Call Pickup Support	✓	✓
Camp On	✓	✓

User Features	Receptionist	Receptionist Office Edition
Operator Barge In	✓	✓
Day / Night Mode	✓	
Queuing	✓	
ACD / Call Queue Support	✓	
ACD / Call Queue States Support	✓	
ACD / Call Queue Monitoring	✓	
ACD / Call Queue Manipulation	✓	
ACD / Call Queue Logs	✓	
ACD / Call Queue Integration	✓	
Client / Server		
Automatic Sign-In Option	✓	✓
Requires Java Installation	✓	✓
Multi-tenanted Support	✓	
Usability		
Hot Desking Capable (Hoteling Guest)	✓	✓
Caller ID	✓	✓
Switchboard	✓	✓
Company Notes	✓	
Company Profile	✓	
Dial from Outlook Contacts	✓	
Keyboard Shortcuts	✓	✓
Customisable Panel Sizes	✓	✓
Repositories		
Personal Directory	✓	✓
Group / Enterprise Directory	✓	✓
Monitored Contacts	200 static, 300 dynamic, enterprise wide	8 group wide
Speed Dial 8/100	✓	✓
LDAP Directory Lookup	✓	
Outlook Directory Lookup	✓	
Call History / Logs	✓	✓
Call Stats	✓	
Contact Status	✓	✓
Mass Contact Support (40,000)	✓	
Multiple Directory Views	✓	
Keywords Contact Search	✓	✓
Custom Search Filter	✓	
Integration		
Outlook Integration	✓	
CLI Delivery via Outlook	✓	
Features		
Call Forwarding Always (Day/Night Mode)	✓	
Voice Messaging User (Day/Night Mode)	✓	

Please note that ACD / Call Queue support requires Receptionist user to be part of a Call Centre and will require the user to be assigned Call Centre Standard (user overlay) licence.

2.22 UNIFIED COMMUNICATIONS FEATURES

Overview

As well as voice, Unity enables users to communicate in multiple other ways too, including:

- Instant messaging
- Presence
- Video calling
- Desktop sharing

Users can access the Unity service as well as these ways of communicating via a Unity UC software client which is available on Windows desktop, Apple OS X desktop, iPhone, Android smartphone and Android tablet PCs.

Feature List

Feature	Description
Instant messaging	Enables Unity UC users to chat to other Unity UC users via instant message.
Presence	User presence including on a call and in a meeting via Outlook integration. Users can also manually configure their presence via the Unity client.
Voice calling	Users can make and receive voice calls to and from any telephone number permissible under their calling plan permissions.
Multi-way voice calling	Users can partake in multi-way voice calls in line with their calling plan permissions. The maximum number of participants in multi-way voice calls is 6.
Video calling	Users can make and receive video calls with other Unity UC users.
My Room	Combines multiple communications tools into a single virtual room. Enables a Unity UC user to host multi-way instant message, voice and desktop sharing sessions.
Desktop sharing	Enables Unity UC users to see and share presentations, spreadsheets, and other media from the Unity client.

License packs and features

The Unity Collaborate service pack provides the assigned user with the full range of telephony and UC features as well as clients for both desktop and mobile devices.

Existing Unity customers can upgrade to the Unity Collaborate Pack and should contact their account manager for pricing. Upgrading in this way means that the Unity Collaborate Pack will replace users' existing service packs.

A Unity Collaborate pack is also available as an overlay to provide UC functionality to Receptionist users. When upgrading existing users, all of their current functionality will be retained including their existing DDI. The upgrade will simply add the additional UC features onto their user account.

Unity software client

The Unity client supports consistent set of features across different devices. The following matrix shows which features are available on which client:

Feature	Windows desktop	Apple desktop	iPhone	Android phone	Android tablet
Contact directory	Yes	Yes	Yes	Yes	Yes
Unity call control settings	Yes	Yes	Yes	Yes	Yes
Presence	Yes	Yes	Yes	Yes	Yes
Instant messaging	Yes	Yes	Yes	Yes	Yes
Group chat	Yes	Yes	Yes	Yes	Yes
Voice calling	Yes	Yes	Yes	Yes	Yes
N-way voice calling	Yes	Yes	Yes	Yes	Yes
N- way Video calling	Yes	Yes	Yes	Yes	Yes
Call & message history	Yes	Yes	Yes	Yes	Yes
My Room	Yes	Yes	Yes	Yes	Yes
Desktop sharing	Yes	Yes	No	No	No

Please note that the Unity client is included for desktop and smartphones as part of the Unity Collaborate service Pack.

2.23 GROUP SERVICE LICENCES

Group licences provide a group (typically associated with a Customer site) with features that are not specific to individual users. These features are group wide and bring added functionality, as required.

A group service licence is required for each instance of each group service required.

The follow table describes the range of group licences:

Call Pickup Group	Call Pickup allows users to answer any ringing line within their call-pickup group. A call-pickup group is defined by the administrator and is a subset of the users in the group that can pick up each other's calls. A single group can have multiple Call Pickup groups defined simultaneously, but a given user can only belong to a single Call Pickup group.

Hunt Group	The Hunt Group service allows incoming calls to a central phone number to be distributed among the members of that group according to a hunting policy. Available hunting policies are as follows: • Regular (linear) – The incoming calls to the group start hunting on the first user in the list and hunt all the provisioned users sequentially, until an idle user is found or the end of the list is reached. • Circular – The incoming calls to the group start hunting with the user following the last user to receive a call. When the end of the list is reached, the hunting circles back to the first user in the list. The hunting ends when an idle user is found or all the users have been visited. • Uniform – The incoming calls to the group are presented with the user that has been idle for the longest time. • Simultaneous – The incoming calls alert all idle users in the group. The call is connected to the first user to answer the call. • Weighted – The incoming calls alert agents in a pseudo-random fashion according to their relative weight. Agents with a higher weight are assigned more incoming calls than agents with lower weights. In all cases, if all users in the hunt group are busy, the incoming call is provided with the busy processing that applies to the hunt group.
Call Centre Standard (Group Feature)	Call Centre Standard (Group Feature) allows incoming calls to a central phone number to be distributed among associated users or agents according to a definable hunting policy (as per Hunt Group). Call Centre Standard (Group Feature) also includes administrator-configurable options for queuing calls when all agents are busy, caller announcements for queue position / wait time, comfort messages and a number of routing policies for managing overflows and busy periods. Users must be assigned to the Call Centre Standard (Group Feature) in order for them to receive calls from it. All users that need to be assigned to a Call Centre Standard (Group Feature) must be assigned the Call Centre Standard (User Overlay) licence. Users who are licensed with Call Centre Standard (User Overlay) may also be licensed with Call Centre Agent (User Overlay) or Call Centre Supervisor (User Overlay), providing enhanced Call Centre functionality and advanced supervisor reports. Note that the Music on Hold feature of Call Queue (ACD) does not require a separate Music On Hold licence. Music on Hold as part of Call Queue (ACD) may be configured separately to that of the rest of the group or department. (It is the Customer's responsibility to ensure that they possess the required broadcasting rights to any media they choose to upload.)

Auto Attendant (IVR)	Auto Attendant (IVR) provides effective fielding and delivery of inbound calls and to the intended destination through interactions with the caller. It is reached by dialling an associated telephone number or extension. Once connected, the caller is played a customisable greeting that provides a menu of options to route their call. The menu, which is configured by the group administrator, can provide the caller with a number of options as follows: • One-Key Dialling – The caller presses a pre-defined DTMF key to reach a particular phone number or extension within the group. This option is also used to build multi-level IVR menus. • Operator Dialling – The caller presses a pre-defined DTMF key to reach an operator. • Name Dialling – The caller spells the name of the intended party, using the numerical DTMF keypad. Upon identifying a unique match, the caller is played the name of the called party and is then transferred. • Extension Dialling – The caller enters the extension of the intended party through the numerical DTMF keypad. Upon collecting the full extension, the caller is played the name of the called party, and is then transferred. • The group administrator may define a holiday schedule that can be associated with an Auto Attendant. This allows the Auto Attendant to automatically adjust the presented options in line with working business hours.
Music on Hold	This service allows a group administrator to upload an audio file that can be broadcast to held parties during Call Hold and Call Park. A system default audio file is provided which may be used freely as part of the Unity IP Voice Service. Music on hold may be specified down to department level, allowing different music on hold files to be used for different departments. It is the Customer's responsibility to ensure that they possess the required broadcasting rights to any media they choose to upload.
Authorisation Codes	The Authorisation Codes feature allows the group administrator to select specific users who must enter a valid authorization code, when making a call to a party outside of the group. Users assigned the Authorisation Codes feature are prompted to enter a valid authorisation code when making a call outside of the group. The code entered by a user must match one of the valid codes previously configured by the group administrator. Emergency calls are never prompted for an authorisation code.

2.24 GROUP SERVICE ENHANCEMENTS

Certain group services can be enhanced with individual user type features.

There are two optional enhancements for Group Services:

- Group Service Voicemail
- Group Service Enhancement Pack

Group Service Voicemail allows the Auto Attendant (IVR), Call Centre Standard (Group Feature) and Hunt Group services to be overlaid with voicemail.

Group Service Enhancement Pack allows Auto Attendant (IVR), Call Centre Standard (Group Feature) and Hunt Group services to be overlaid with any of the following features. Multiple features from the following list may be applied as appropriate.

- Call Forwarding Busy
- Call Forwarding Always
- Call Forwarding Selective
- Alternate Numbers
- Selective Call Acceptance
- Selective Call Rejection
- Pre-alerting Announcement

One instance of Group Service Enhancement Pack is required per group service needing any of the above features.

2.25 LICENCE REALLOCATION FEE

Where licences require reallocating from one Unity Group to another, Redcentric will charge an administration fee.

2.26 MOVES, ADDS AND CHANGES ADMINISTRATION FEE

Adds, moves and changes defined in this service definition as "Customer Managed Adds, Moves and Changes" are to be undertaken by the Customer via the Unity web administration portal.

Where the Customer requests that Redcentric undertake any of these Customer Managed Adds, Moves and Changes, this will incur an administration fee, per change.

2.27 UNITY CALL CENTRE

Overview

Unity Call Centre is an optional enhancement to Unity IP Voice that is designed for help desk and call centre. The system is flexible enough to be used for small teams where only a few agents are needed, to larger complex environments with many agents supporting multiple call queues. Unity Call Centre requires no additional hardware and is delivered by simply overlaying additional licences on a per user basis.

Call Centres and Call Centre Users

In order to join call centres, users must be assigned a call centre user overlay licence. This overlay is added in addition to their primary service pack.

Once a user has been licensed for call centre, they may join single or multiple call centres. Each call centre works in a similar fashion to a Hunt Group by distributing inbound calls to a group of predefined users. A configurable hunting policy defines the pattern in which these users are rung. There is also the ability to queue callers in the event that all users assigned to the call centre are unavailable. Calls that have been held for a definable period of time can be overflowed to other call centres or other destinations. Callers into the call centre can be presented with a welcome announcement, music on hold, call position announcements and intermittent comfort messages.

There are two variants of call centre user overlay licence, Call Centre Standard and Call Centre Premium. The table below compares features between the two overlay licences:

User Features	Call Centre Standard	Call Centre Premium
Call Distribution Features		
Maximum calls in queue	50	525
Regular call distribution	✓	✓
Circular call distribution	✓	✓
Uniform call distribution	✓	✓
Simultaneous call distribution	✓	✓
Weighted call distribution	✓	✓
Announcements		
Entrance	✓	✓
Estimated Wait Message	✓	✓
Comfort Message	✓	✓
Music on Hold	✓	✓
Comfort Bypass		✓
Whisper		✓
Call Processing Features		
Overflow	✓	✓
Stranded calls	✓	✓
Bounced calls	✓	✓
Holiday services		✓
Night service		✓
Forced forwarding		✓
Reporting *		
Basic Statistics	✓	✓
Advanced Reporting *	✓	✓
Call Centre Monitoring		
Silent Monitoring (Supervisors) **		✓

* Please note that reporting functionality (with the exception of Basic Statistics) requires additional Call Centre Agent and Call Centre Supervisor overlay licences to be assigned to call centre users (see Advanced Call Centre Functionality).

** Please note that only agents licenced with Call Centre Premium can be monitored.

Call Distribution Features

A call received by a call centre can be distributed to the various agents staffing the queue using the following different policies:

- **Regular call distribution** – Incoming calls hunt through agents in the order they appear on the list, starting from the top each time.
- **Circular call distribution** – Incoming calls hunt through agents in the order they appear on the list, starting with the agent who follows the agent who received the previous call. When the search reaches the end of the list, it loops back to the top and continues until it has tried all users.
- **Uniform call distribution** – Incoming calls hunt through all agents, starting with the agent who has been idle the longest and ending with the agent who most recently answered a call.
- **Simultaneous call distribution** – Incoming calls alert all agents at the same time. The first agent to answer handles the call.
- **Weighted call distribution** – Incoming calls are assigned to idle agents based on percentages assigned to the agents in the call centre's profile. This feature supports an element of skills-based routing, since a higher percentage of calls can be routed to more highly skilled agents within the call centre.

Call Centre Announcements

When a call is received by the call centre, audio announcements can be configured to be played to the caller. Default audio announcement files are provided for each announcement / treatment.

- **Entrance Message -** the entrance message is the first message played to the caller when the call reaches the call centre. The entrance message is optional. If the entrance message option is enabled, it is played under the following conditions:
    - There are no agents available to accept the call –or- Agents are available, and the *Play ringing when offering call* option is not enabled. Once the entrance message has finished playing, Music On Hold and comfort messaging are provided to the caller, if enabled.

- **Mandatory Entrance Message -** An option is provided to force the playback of the entrance message on each inbound call. When enabled, the entrance message is mandatory and is played to completion before any attempt is made to offer the call. The caller cannot escape out of the message using the escape digit and a supervisor cannot manipulate the call during the mandatory entrance message. The message is also played before processing the call out of the queue by a Call Centre policy, such as Stranded Calls, or Overflow based on time.

- **Estimated Wait Message -** The Estimated Wait Message (EWM) provides queue information to the caller. When a new call is added to the call queue, the EWM is played after the entrance message and before any other announcement. If the entrance message is disabled and the EWM is enabled, the EWM is played. This announcement has two modes of operation:
    - Queue Position: In this case, the caller is informed of their current position in the call queue (for example, "You are caller number 12 in the queue").
    - Estimated Waiting Time: In this case, the caller is given an estimated number of minutes before the call will be answered by an agent (for example, "Your estimated wait time is 5 minutes").

  The estimated wait message announcements are localised according to the language of the queue. The EWM is only played once when a call first enters the call centre. If the caller overflows to another queue, they hear a new EWM, if the message is enabled.

- **Comfort and Music-On-Hold Messages -** These messages are played to the caller after the entrance message (if configured to play). The Music-On-Hold (MOH) message is played before the comfort message, when configured. The administrator can specify the use of the default file or a custom file. The comfort and Music-On-Hold messages keep playing to the caller in a loop until the call is answered by an agent or until action is taken by a Call Centre policy (for example, Overflow).

- **Comfort Message Bypass -** An alternate comfort message can be enabled for calls that are expected to be answered quickly instead of the usual comfort/Music-On-Hold treatments. This policy applies after the entrance message has finished playing (if applicable). The time threshold that triggers the comfort message bypass is configurable. When a new incoming call is received by the queue, if the longest waiting time for a call in the queue is less than or equal to this threshold, then the alternate Comfort Message service is triggered. The comfort message bypass options include playing ringback and/or playing a specific comfort message bypass announcement.

- **Call Whisper Message** - The call whisper message is a message that is played to the agent immediately before the inbound call is connected. The calling party hears ringback, announcements, or Music On Hold during the whisper message. The message typically announces which call centre the call is coming from. This allows the agent to identify which call centre the caller has dialled without the need to look at their phone or Call Centre Agent client. For instance, ten separate numbers can all be routing their inbound calls to a single call centre. Each of the ten numbers can have their own customised call whisper message so the agent knows which number was dialled and can provide the appropriate greeting.

2.27 Call Processing Policies

The call centre has various policies that affect the processing of calls received by the call centre. The following section describes how these policies work.

- **Overflow Policy -** There are two types of overflow scenarios that can occur in a call centre queue:
    - Based on size – These overflow scenarios occur when an incoming call cannot be queued because the queue has reached the configured maximum quantity.
    - Based on time – These overflow scenarios occur when a queued call is not handled (either by an agent or by another Queue policy) within a specified amount of time. The call is removed from the queue and handled according to the related Overflow policy actions.

Options are available to configure the threshold used to determine when a call overflows based on time as well as to determine the size of the queue.

The following actions may be performed on a call that has triggered overflow:

- Busy: Overflow calls are provided with Busy treatment. If the queue is configured with the Call Forwarding Busy or the Voice Messaging service, then the call is handled accordingly.
- Transfer: Overflow calls are transferred to the configured destination.
- Ringing: Overflow calls are provided with ringing until the caller releases the call.

For all the types of actions, the policy can be configured to play an announcement prior to proceeding with the action. In this case, the announcement is played once to completion before the action is processed.

- **Stranded Policy** - This policy allows for the configuration of the processing of stranded calls. A stranded call is a call that is being processed by a queue that has no agents currently staffed. (An agent is said to be staffing a queue if the agent has joined the queue and is not in the *Sign-out* state.) If the last agent staffing a queue "unjoins" the queue or signs out, then all calls in the queue become stranded and handled as described. If an incoming call is received by a queue with no agents staffing the call centre, then the call is initially put in the queue. Once the queued call is ready to be offered to an agent, if there are no agents staffing the queue, then the call is processed as a stranded call. In particular, if the mandatory *Entrance Message* option is enabled, then the entrance message is played to completion before the call is handled as a stranded call. There are multiple options for handling stranded calls. The following "actions" may be configured:
    - None: Calls remain in the queue.
    - Busy: Calls are removed from the queue and are provided with Busy treatment. If the queue is configured with the Call Forwarding Busy or the Voice Messaging service, then the call is handled accordingly.
    - Transfer: Calls are removed from the queue and are transferred to the configured destination.

  - Night Service (*Premium* only): Calls are handled according to the Night Service configuration. If the Night Service action is set to "none", then this is equivalent to this policy being set to "none" (that is, calls remain in the queue).
  - Ringing (*Premium* only): Calls are removed from the queue and are provided with ringing until the caller releases the call. The ringback tone played to the caller is localized according to the country code of the caller.
  - Announcement (*Premium* only): Calls are removed from the queue and are provided with an announcement that is played in a loop until the caller releases the call.

- **Bounced Policy** - This policy handles processing of bounced calls. A bounced call is a call that is being routed to the agent but for some reason (agent does not answer the call, they change to unavailable, their device is not registered, and so on) the call is not answered. Options are configurable to flag a call as a bounced call if the agent fails to answer a call within the specified amount of time (as determined by the number of rings and the applicable ring cycle). A bounced call is treated with the highest importance and is placed ahead of the rest of the non-bounced queued calls in the queue. An option is available to indicate whether a call should be flagged as a bounced call if the agent receiving the call changes to the *Unavailable* state while the call is being presented to them.

  The call can be transferred to a new destination upon being bounced.
  The Supervisor client application is also notified and shows a visual indicator that a queue entry is a bounced call.

  An option is configurable to alert an agent if a call is kept on hold by the agent for a specified duration. In addition, an option is also provided to bounce the call back to the queue if the call is kept on hold for longer than a specified duration. When the held call centre call is the only call present on the agent's device, this triggers a hold reminder to the agent in the form of a ring splash.

- **Holiday Service Policy** - This policy allows calls to be processed differently during holiday periods. The Queue policy refers to a particular schedule and allows configuration of a specific routing action when a call is received during this schedule. The following actions may be configured:
  - None: This is equivalent to not having a holiday schedule. The call is processed as if it was received during a non-holiday period.
  - Busy: The incoming call is provided with Busy treatment. If the queue is configured with the Call Forwarding Busy or the Voice Messaging service, then the call is handled accordingly.
  - Transfer: The incoming call is transferred to the configured destination.

  In the case of the *busy* and *transfer* actions, the policy can be configured to play an announcement prior to proceeding with the action. In this case, the announcement is played once to completion before the action is processed.

- **Night Service Policy -** This policy allows calls to be processed differently during non-business hours. Business hours are defined as a time schedule at the group level. The Queue policy refers to this and allows the configuration of a specific routing action when a call is received outside of business hours. By default, an "Every Day, All Day" business hour schedule is defined for the queue. The following actions can be configured:

  - None: This is equivalent to having an "Every Day, All Day" business hour schedule.
  - Busy: The incoming call is provided with Busy treatment. If the queue is configured with the Call Forwarding Busy or the Voice Messaging service, then the call is handled accordingly.
  - Transfer: The incoming call is transferred to the configured destination.

  For the *busy* and *transfer* actions, the policy can be configured to play an announcement prior to proceeding with the action. In this case, the announcement is played once to completion before the action is processed.

Advanced Call Centre Functionality

Once users have been assigned the Call Centre Standard (user overlay) or Call Centre Premium (user overlay) licence and are part of a call centre (or multiple call centres), advanced call centre functionality can be delivered by overlaying additional licences to the users. These licences are summarised below:

Call Centre Agent licence upgrades the user, providing them with a desktop PC application (or thin web client), inbound call visibility, contact directory access with click to dial, call control and call centre availability control. Call Centre Agents also have the ability to escalate calls to predefined Supervisors.

Call Centre Supervisor licence upgrades the user, providing them with a desktop PC application (or thin web client) and supervisor functionality including visibility of the queue, the ability to manipulate the queue, and real time visibility of agent status. Supervisors also get access to a range of advanced real time and historic reports on the performance of the queue and the agents.

2.28 CALL CENTRE REPORTING

Basic Statistics

Each call centre includes access to basic statistics on queue and user performance. These statistics are accessible to Group Administrators (with read/write access) via the CommPilot web administration portal.

Statistics can be viewed by the administrator by specifying a time period across which the report is to be run. A reporting function is also available that that automatically e-mails reports out to up to two recipients on a daily basis.

The agent statistics are provided for the 24-hour period ending at midnight when the report is generated.

Basic statistics are stored for a minimum of 48 hours.

The following details statistics available in the basic statistics.

Queue Statistics

The following table shows the queue statistics. These statistics are listed in the order in which they are displayed in the files in a statistics report as well as on the web interface.

Statistics	Description
Number of busy overflows	This is the number of calls that came in after the queue limit was exceeded. Such calls are likely forwarded to voice mail.
Number of calls answered	This is the total number of calls answered handled by an agent.
Number of calls abandoned	This is the total number of calls for which the caller has hung up or selected to leave a message before an agent became available.
Number of calls transferred	This is the total number of calls that are transferred out of the Call Centre queue. Typically, a call is transferred from a given Call Centre queue to another Call Centre queue using a client application (for example, by the Supervisor using their desktop application).
Number of calls timed out	This is the total number of calls that remain unanswered and that are forwarded out of the Call Centre queue upon timeout.
Average wait time	This is the average amount of time that callers spend waiting for the next available agent to answer the call.
Average abandonment time	This is the average time that callers spend waiting for an agent before hanging up or selecting the option to leave a message.

Average number of agents staffed	This is the average number of agents staffed during the period for this Call Centre instance. An agent who has joined the Call Centre and who is not in the *sign-out* state is considered as staffed.
Average number of agents talking	This is the average number of agents who were in the *talking* state during the period for this Call Centre instance.

User (Agent) Statistics

The following table shows the agent statistics. In basic statistics, an Agent is defined as a user who is licensed to join a Call Centre and is assigned to the respective one. Agent statistics are available on a per-queue basis. If an agent is assigned to multiple queues, then agent statistics are independently maintained and provided for each queue. These statistics are listed in the order in which they are displayed in the files in a statistics report as well as on the web interface.

Statistics	Description
Number of calls handled	This is the total number of calls that the agent has handled. This statistic accounts for all Call Centre calls that are released by the agent during the specified period.
Average call time	This is the average time that an agent spends on calls from the Call Centre instance. This statistic accounts for all Call Centre calls that are released or transferred by the agent during the specified period. If the agent transfers a call (for example, to another queue), then the call time only accounts for the time spent on the call by the agent prior to the call transfer. In previous releases, the call time after the call transfer would be allocated to both the redirecting agent and the agent answering the call from the other queue. This behaviour is changed upon upgrade and is not activatable.
Number of calls unanswered	This is the total number of calls extended to the agent that are not answered (for any reason other than because the agent is busy). Note that for a single call to a Call Centre instance, an agent may be rung multiple times as the call can be placed in the queue and presented to the agent again. Therefore, this statistic may be incremented more than once for a given call to the Call Centre instance.
Total Talk Time	The amount of time that the agent was busy handling calls for this Call Centre instance.

Total staffed time	The amount of time that the agent has joined the Call Centre instance and was not in the *sign-out* state.

Advanced Reporting – Description

The Advanced Reporting capability enables Call Centre supervisors to view historic and real time reports on the performance of the Call Centre Queue(s) and assigned Agents.

Advanced Reporting requires that users assigned to the Call Centre(s) have the Call Centre Agent overlay licence, as well as Call Centre Standard user overlay. It also requires at least one user to have the Call Centre Supervisor user overlay, in order for them to view the reports via the Supervisor level Call Centre desktop application.

Dashboard

The *Dashboard* tab shows a summary of real-time measurements on the monitored call Centres and agents.

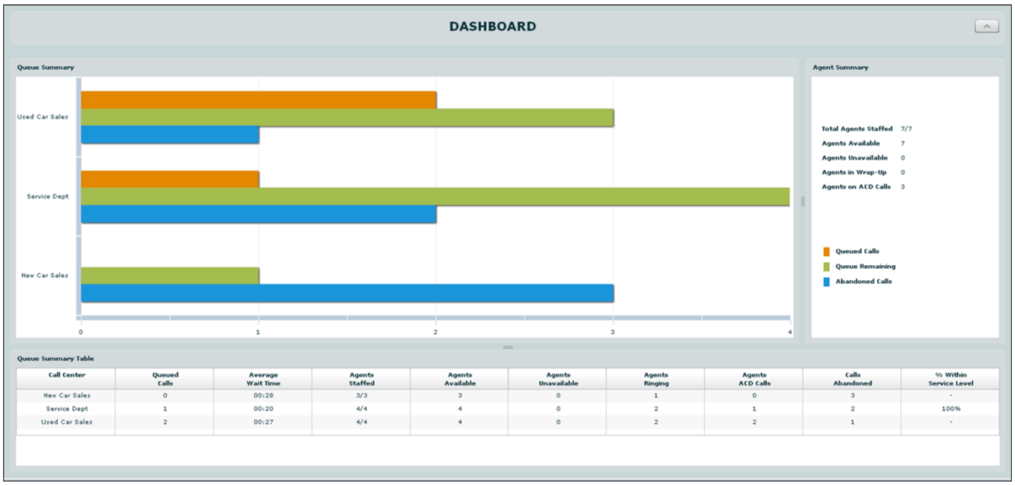


Chart Information for Dashboard

The Dashboard contains the components in the following subsections.

The ***Agent Summary*** shows the real-time key performance indicators for the supervised team of agents. The statistical measures are shown in the following table:

NOTE: It is recommended that all the agents for the Call Centre are assigned to the supervisor so the Queue Summary table is more accurate. Only metrics for agents assigned to the supervisor are shown in the table.

Statistic	Description	Calculation
Total Agents Staffed	The ratio of signed-in agents managed by the supervisor compared to the total number of agents managed by the supervisor for this call Centre.	Count of all agents managed by the supervisor that are in *Sign-In*, *Available*, *Unavailable*, or *Wrap-Up* ACD state, over the count of all agents managed by the supervisor.
Agents Available	The number of agents currently showing *Available* as their ACD state.	Count of all agents managed by the supervisor that are in *Available* ACD state.
Agents Unavailable	The number of agents currently showing *Unavailable* as their ACD state.	Count of all agents managed by the supervisor that are in *Unavailable* ACD state.
Agents in Wrap-Up	The number of agents currently showing *Wrap-Up* as their ACD state.	Count of all agents managed by the supervisor that are in *Wrap-Up* ACD state.
Agents on ACD Calls	The number of agents in *Talking phone* state on calls distributed from the ACD.	Count of all agents managed by the supervisor that are in *Talking* state on ACD calls.

The ***Queue Summary*** provides the following information:

Statistic	Description	Calculation
Call Centre	The names of the call centres the supervisor has access to.	Not applicable
Queued Calls	The number of calls that are queued in that call Centre.	Count of calls that are currently queued, waiting for agents.
Average Wait Time	The average time a caller has been waiting in the queue.	Calculation of the average amount of time a caller spends in the queue before the call is offered to an agent.
Agent Staffed	The ratio of signed-in agents managed by the supervisor, compared to the total number of agents managed by the supervisor for this call Centre.	Count of all agents managed by the supervisor that are in *Sign-In*, *Available*, *Unavailable*, or *Wrap-up* ACD state, over the count of all agents managed by the supervisor.
Agents Available	The number of agents in *Available* ACD state.	Count of all agents managed by the supervisor that are in *Available* ACD state.
Agents Unavailable	The number of agents currently in *Unavailable* ACD state.	Count of all agents managed by the supervisor that are in *Unavailable* ACD state.
Agents Ringing	The number of agents in *Ringing phone* state.	Count of all agents managed by the supervisor that are currently in *Ringing* state (call is being offered to an agent).
Agents ACD Calls	The number of agents in *Talking phone* state on calls distributed from the ACD.	Count of all agents managed by the supervisor that are currently in *Talking* state on an ACD call. Agents on direct dialed inbound or outbound calls are not included.
Calls Abandoned	The number of callers who have terminated the call while waiting in the queue.	Count of the number of calls that were abandoned while waiting in the queue. This measurement starts at zero when the supervisor signs in to the Call Centre client.

Statistic	Description	Calculation
% Within Service Level	The percentage of calls that were answered within a pre-defined period of time. This is shown in minutes and seconds (mm:ss). The period of time is defined using the slide bar at the top of the tab. A typical *Service Level* use case is as follows: Let us suppose that the goal is to answer 80% of calls within 20 seconds. The supervisor defines the period of time at 00:20 (20 seconds) on the slide bar. This measurement then reflects the percentage of calls that were answered within 20 seconds. Therefore, if the measurement is less than 80%, the supervisor knows they are outside of their target *Service Level*.	Calculation of the percentage of calls that are answered within the designated *Service Level* time. % Within Service Level = Number of calls answered within service level / Number of calls answered in the interval + number of calls abandoned in the interval.

Agent Activity Report (Supervisor and Agent)

NOTE: Agents can use the Agent Activity Report to view their own statistics only.

The Agent Activity Report displays metrics related to a selected agent's call handling activity for all call centres for which the agent is a member. The heading for this report is Agent Activity – Agent Full Name – Interval, for example, *Agent Activity – Agent X – Half Hourly Report*. This report can also be set to view summary data for all agents.

To view an Interval report (half-hourly, hourly, and so on), the report must be requested for an individual agent.

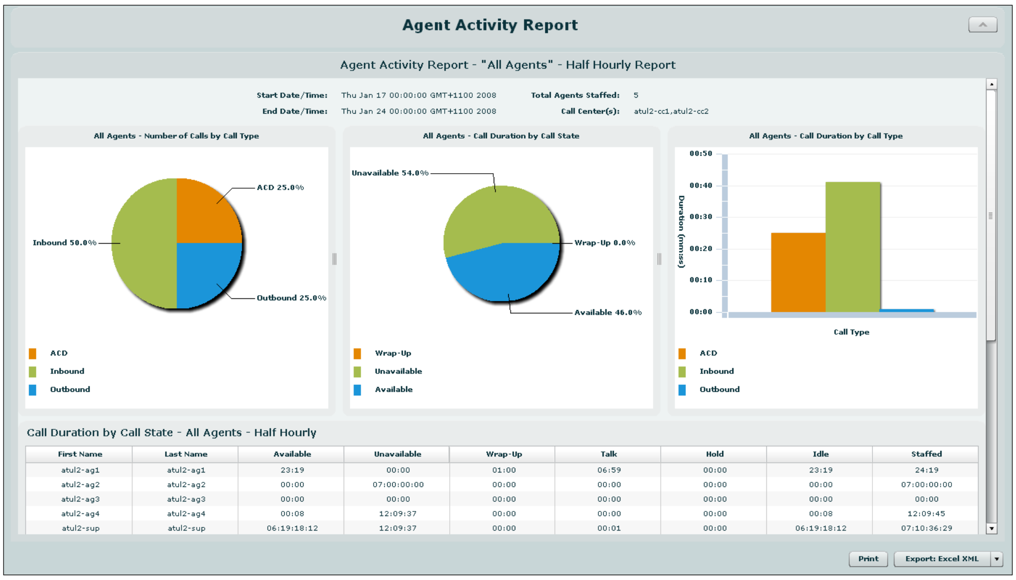


Chart Information for Agent Activity Report

The following table describes the graph information on the Agent Activity Report, from left to right:

Statistic	Description
Number of Calls by Call Type (Pie Chart)	
ACD	The percentage of ACD calls that an agent has answered during the specified interval.
Inbound	The percentage of direct non-ACD calls that an agent answered during the specified interval.
Outbound	The percentage of answered outbound calls that an agent made during the specified interval. **NOTE**: This includes calls that were made while performing Consultative Transfer, Escalate to Supervisor, and Conference. However, if the agent transfers the call before the third party answers, then the call is not included. In addition, calls resulting from a blind transfer initiated by the agent are not included.
Call Duration by Call State (Pie Chart)	
Wrap-Up	The percentage of time that an agent spent in *Wrap-Up* state during the specified interval.
Unavailable	The percentage of time that an agent spent in *Unavailable* state during the specified interval.
Available	The percentage of time that an agent spent in *Available* state during the specified interval.
Call Duration by Call Type (Vertical Bar Chart)	
ACD	The duration in minutes and seconds that an agent spent on ACD calls during the specified interval. This includes talk time and hold time.
Inbound	The duration of time in minutes and seconds that an agent spent answering direct non-ACD calls during the specified interval.
Outbound	The duration of time in minutes and seconds that an agent spent making outbound calls during the specified interval. **NOTE**: This includes calls that were made while performing Consultative Transfer, Escalate to Supervisor, and Conference. However, if the agent transfers the call before the third party answers, then the call is not included. In the case where the agent transfers the call after the third party answers, then the duration does not account for the call time of the other parties after transfer. Calls resulting from a blind transfer initiated by the agent are not included.

Table Information for Agent Activity Report

The following table describes the columns on the Agent Activity Report, from left to right. Where applicable, calculations are provided for statistics that are derived from other fields:

Statistic	Description
Call Duration by Call State	
First Name	The first name of an agent. This is not the CLID name.
Last Name	The last name of an agent. This is not the CLID name.
Available	The time during the specified interval that an agent was in the *Available* state.
Unavailable	The time during the specified interval that an agent was in the *Unavailable* state.
Wrap-Up	The time during the specified interval that an agent was in the *Wrap-Up* state.
Talk	The time during the specified interval that an agent spent on ACD calls in the *Talking* state.
Hold	The time during the specified interval that an agent spent on ACD calls in the *Hold* state.
Idle	The time during the specified interval that an agent spent idle in the *Available* state. The agent is considered idle if they are not on an ACD call.
Staffed	The time during the specified interval that an agent spent not in the *Signed-Out* state. This includes *Signed-In*, *Available*, *Unavailable*, and *Wrap-up* states.
Number of Calls by Call Type	
First Name	The first name of an agent. This is not the CLID name.
Last Name	The last name of an agent. This is not the CLID name.
ACD	The number of ACD calls that were answered by an agent during the specified interval.
Inbound	The number of direct non-ACD calls answered by an agent.
Outbound	The number of answered outbound calls an agent made during the specified interval. **NOTE**: This includes calls that were made while performing Consultative Transfer, Escalate to Supervisor, and Conference. However, if the agent transfers the call before the third party answers, then the call is not included. In addition, calls resulting from a blind transfer initiated by the agent are not included.
Call Duration by Call Type	
First Name	The first name of an agent. This is not the CLID name.
Last Name	The last name of an agent. This is not the CLID name.
ACD	The duration of ACD calls answered by an agent during the specified interval.
Inbound	The duration of direct non-ACD calls an agent answered during the specified interval.
Outbound	The duration of outbound calls an agent spent making during the specified interval. **NOTE**: This includes calls that were made while performing Consultative Transfer, Escalate to Supervisor, and Conference. However, if the agent transfers the call before the third party answers, then the call is not included. In the case where the agent transfers the call after the third party answers, then the duration does not account for the call time of the other parties after transfer. Calls resulting from a blind transfer initiated by the agent are not included.

Agent Utilisation Report

The Agent Utilisation Report displays metrics related to a selected agent's call handling activity for all call centres for which the agent is a member. The heading for this report is Agent Utilisation – Agent Full Name – Interval, for example, Agent Utilisation – Agent X – Half Hourly Report. This can also be set to view all agents.

Chart Information for Agent Utilisation Report

The following table describes the graph information in the Agent Utilisation Report, from left to right:

Statistic	**Description**
Agent Call Summary (Pie Chart)	
ACD Calls	The percentage of ACD calls that were answered by an agent during the specified interval.
Inbound Calls	The percentage of direct non-ACD calls that were answered by an agent during the specified interval.
Outbound Calls	The percentage of answered outbound calls that an agent made during the specified interval. **NOTE**: This includes calls that were made while performing Consultative Transfer, Escalate to Supervisor, and Conference. However, if the agent transfers the call before the third party answers, then the call is not included. In addition, calls resulting from a blind transfer initiated by the agent are not included.
Held Calls	The percentage of ACD calls that an agent placed on hold during the specified interval. **NOTE**: This measurement captures the number of times an agent placed a call on hold. A single call placed on hold 3 times counts as 3 held call events.
Agent Call Summary by Call Type (Pie Chart)	
Avg ACD Duration	The duration percentage that an agent spent on ACD calls during the specified interval.
Avg Inbound Duration	The duration percentage that an agent spent on direct non-ACD calls.
Avg Outbound Duration	The duration percentage that an agent spent on outbound calls. **NOTE**: This includes calls that were made while performing Consultative Transfer, Escalate to Supervisor, and Conference. However, if the agent transfers the call before the third party answers, then the call is not included. In the case where the agent transfers the call after the third party answers, then the duration does not account for the call time of the other parties after transfer. Calls resulting from a blind transfer initiated by the agent are not included.

Statistic	Description
Agent Performance (Line Chart)	
Avg ACD Time	The average time in minutes and seconds that an agent spent on ACD calls during the specified interval.
Avg Wrap-Up Time	The average time in minutes and seconds that an agent took to wrap up ACD calls.
Avg Talk Time	The average time in minutes and seconds that an agent spent talking on ACD calls.
Avg Hold Time	The average time in minutes and seconds that an agent spent on hold on ACD calls.
Avg Handle Time	The average time in minutes and seconds that an agent spent handling a call. Handle time = Talk time + Hold time +Wrap-up time *Wrap-up* time is associated with the agent's previous ACD call.

Table Information for Agent Utilisation Report

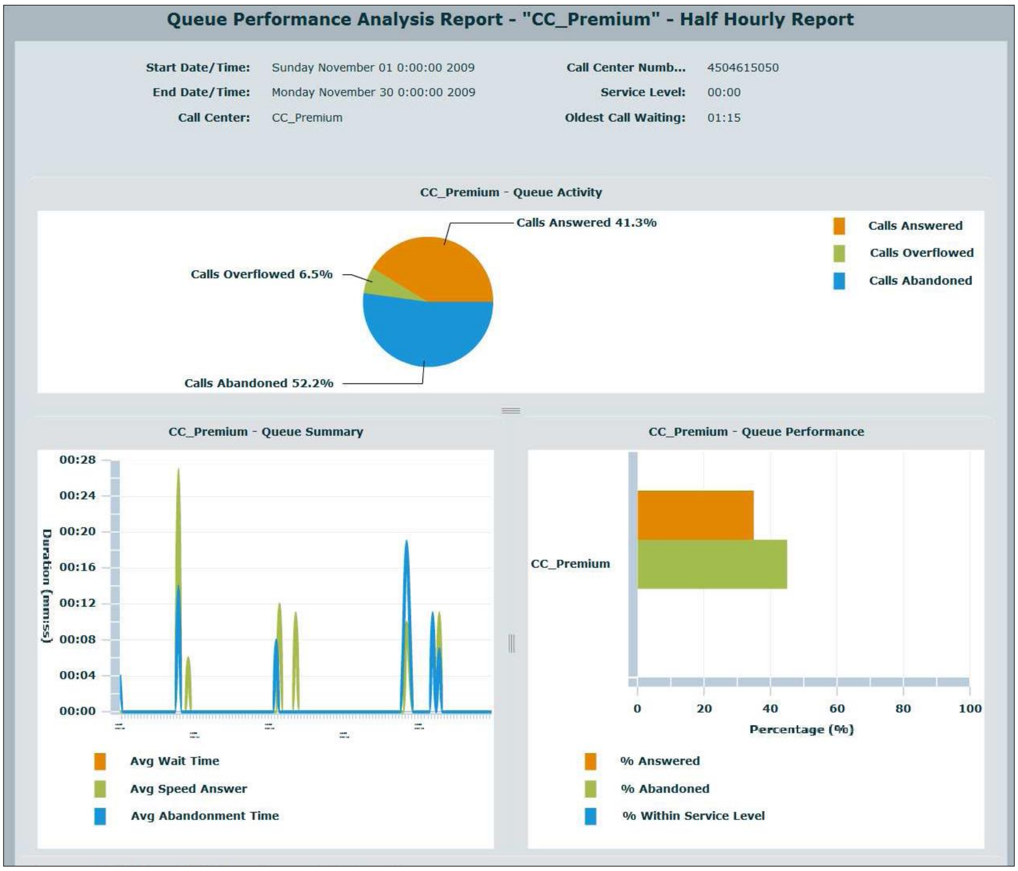
The following table describes the columns in the Agent Utilisation Report, from left to right. Where applicable, calculations are provided for statistics that are derived from other fields:

Statistic	Description
Agent Call Summary	
First Name	The first name of an agent. This is not the CLID name.
Last Name	The last name of an agent. This is not the CLID name.
ACD Calls	The number of ACD calls that an agent answered during the specified interval.
Inbound Calls	The number of direct non-ACD calls that an agent answered during the specified interval.
Outbound Calls	The number of answered outbound calls that an agent made during the specified interval. **NOTE**: This includes calls that were made while performing Consultative Transfer, Escalate to Supervisor, and Conference. However, if the agent transfers the call before the third party answers, then the call is not included. In addition, calls resulting from a blind transfer initiated by the agent are not included.
Held Calls	The number of times an agent placed an ACD call on hold during the specified interval.
Agent Call Summary by Call Type	
First Name	The first name of an agent. This is not the CLID name.
Last Name	The last name of an agent. This is not the CLID name.
Avg ACD Time	The average length of ACD calls during the specified interval. Avg ACD Time = Total ACD Call Time/ACD Calls
Avg Inbound Time	The average length of direct non-ACD during the specified interval. Avg Inbound Time = Total Inbound Call Time/Inbound Calls

Statistic	Description
Avg Outbound Time	The average length of answered outbound calls that an agent made during the specified interval. Avg Outbound Time = Total Outbound Call Time/Outbound Calls **NOTE**: This includes calls that were made while performing Consultative Transfer, Escalate to Supervisor, and Conference. However, if the agent transfers the call before the third party answers, then the call is not included. In the case where the agent transfers the call after the third party answers, then the duration does not account for the call time of the other parties after transfer. Calls resulting from a blind transfer initiated by the agent are not included.
Agent Performance	
First Name	The first name of an agent. This is not the CLID name.
Last Name	The last name of an agent. This is not the CLID name.
Avg ACD Time	The average duration of an agent's ACD calls that includes the ring time, talk time, and hold time of each ACD call during the specified interval. Avg ACD Time = Total ACD Call Time/ACD Calls
Avg Sign-In Time	The average time in minutes and seconds that an agent spent in *Sign-in*, *Available*, *Unavailable,* and *Wrap-Up* ACD states during the specified interval. Avg Signed-In Time = Staffed Time/Number of Sign-Ins by that agent in that interval. **NOTE 1**: The Staffed Time is defined as follows: Sum of time in [Sign-In + Available + Unavailable + Wrap-up]. **NOTE 2**: When ACD states are used as recommended, this value reflects the amount of time the agent spent on their "shift".
Avg Wrap-Up Time	The average time in minutes and seconds it took an agent to wrap up during the specified interval. Avg Wrap-Up Time = Total Wrap-Up Time/Num Wrap-Up
Avg Talk Time	The average of an agent's talk time in minutes and seconds that excludes ring time during the specified interval. Avg Talk Time = Total Talk Time/ACD Calls
Avg Hold Time	The average of an agent's hold time in minutes and seconds during the specified interval. Avg Hold Time = Total Hold Time/ACD Calls
Avg Handle Time	This is the average of an agent's handle time, in minutes and seconds, during the specified interval. Avg Handle Time Handle = Total Talk Time + Total Hold Time + Total Wrap-Up Time)/ACD Calls

Queue Performance Analysis Report

The Queue Performance Analysis Report displays metrics related to the performance of a call centre ACD groups. The heading for this report is Queue Performance Analysis – Call Centre Name – Interval Report, for example, *Queue Performance Analysis – Support – Half Hourly Report*. This can be changed to view "All Call Centres".



Queue Performance - CC_Premium - Half Hourly

Date and Time	% Answered	% Calls in Queue	% Abandoned	% Within Service Level
Nov 10 2009, 14:30	0%	-	100%	0%
Nov 10 2009, 15:00	-	-	-	-
Nov 10 2009, 15:30	-	-	-	-
Nov 10 2009, 16:00	-	-	-	-
Nov 10 2009, 16:30	-	-	-	-
Nov 10 2009, 17:00	-	-	-	-
Nov 10 2009, 17:30	-	-	-	-
Nov 10 2009, 18:00	-	-	-	-
Nov 10 2009, 18:30	-	-	-	-
Nov 10 2009, 19:00	-	-	-	-
Nov 10 2009, 19:30	-	-	-	-
Nov 10 2009, 20:00	-	-	-	-
Nov 10 2009, 20:30	-	-	-	-
Nov 10 2009, 21:00	-	-	-	-
Nov 10 2009, 21:30	-	-	-	-
Nov 10 2009, 22:00	-	-	-	-
Nov 10 2009, 22:30	-	-	-	-
Nov 10 2009, 23:00	-	-	-	-
Nov 11 2009, 9:30	60%	-	40%	0%
Nov 11 2009, 10:00	-	-	-	-
Nov 11 2009, 10:30	-	-	-	-
Nov 11 2009, 11:00	100%	-	0%	0%
Nov 11 2009, 11:30	-	-	-	-

Chart Information for Queue Performance Report

The following table describes the graph information on the Queue Performance Analysis Report, from left to right:

Statistic	Description
Queue Activity (Pie Chart)	
Calls Answered	The percentage of calls that were answered by the call centre.
Calls Overflowed	The percentage of calls that exceeded the configured maximum queue length of time of the call centre.
Calls Abandoned	The percentage of calls that were abandoned while waiting in the call centre queue.
Queue Summary (Line Chart)	
Avg Wait Time	The average time in minutes and seconds that answered calls where in the queue of the call centre.
Avg Speed Answer	The average time in minutes and seconds that it took for calls to be answered in the call centre. *Avg Speed of Answer* includes queue time and alerting time.
Avg Abandonment Time	The average time in minutes and seconds before a call was abandoned while waiting in the call centre queue. Abandonment time does not include transferred or overflowed calls.
Queue Performance (Horizontal Bar Chart)	
% Answered	The percentage of calls that was answered by the call centre.
% Abandoned	The percentage of calls that was abandoned while waiting in the call centre queue.
% Within Service Level	The percentage of calls that was within the designated service level of the call centre.

Table Information for Queue Performance Report

The following table describes the columns in the Queue Performance Analysis Report:

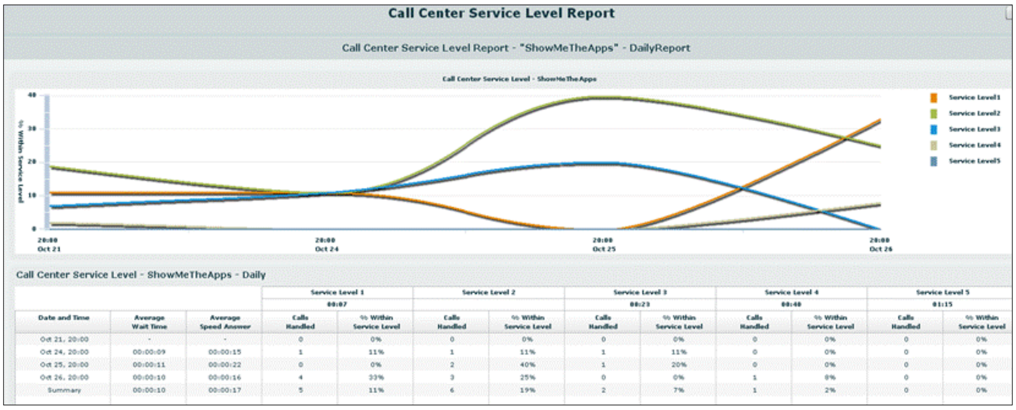
Statistic	Description
Queue Activity	
Call Centre	The list of call centres that the supervisor is monitoring and managing.
Calls Received	The number of calls that were delivered to the call centre. **NOTE**: Calls Received = Calls Answered + Calls In Queue* + Calls Abandoned. **Calls in Queue* is a transient state, so a real-time report increments both the Calls Received and the Calls in Queue until the call is answered, abandoned, or transferred, at which point the Calls in Queue decrement and the other values increment. Overflowed calls are not included in Calls Received since the calls are immediately diverted to another location. Calls transferred into the queue are counted as Calls Received, even when the call is initially answered by an agent in the queue and transferred by the agent back into the same queue.
Calls Answered	The total calls answered by agents.

Statistic	Description
Calls In Queue	The number of calls that are in the queue. ▪ Historical report – This value reflects the number of calls in the queue when the interval changed. ▪ Real-time report – The last interval listed in the report reflects the current number of calls in the queue, subject to the refresh rate. The summary row always shows "–". **NOTE**: When viewing "All Call Centres", this value reflects the sum of the calls left in queue at every interval over the specified time period. This value should only be used to identify which call centres warrant additional analysis, to determine which intervals had the high number of calls left in queue.
Calls Abandoned	The number of calls that are abandoned by the caller when calls are in queue or when calls are ringing for agent. **NOTE**: Calls Abandoned includes escaped calls where the caller decides to press "0" to leave a message (as configured for the Call Centre, Voice Messaging User settings) or gets diverted to a Call Forwarding Busy destination.
Calls Overflowed	The number of calls that exceeded the maximum queue length or the maximum time threshold in the Overflow settings. These calls may be transferred or diverted to voice mail. This value is not included in the number of Calls Received since the calls are immediately diverted to another location.
Queue Summary	
Call Centre	The list of call centres that the supervisor is monitoring and managing.
Avg Time In Queue	The average waiting time or delay in queue excluding ring time displayed in minutes and seconds. Avg Time In Queue = Total Queue Time/Calls Answered
Avg Speed Answer	The average time in queue including ring time displayed in minutes and seconds. Avg Speed = (Total Queue Time + Total Ring Time)/Calls Answered
Avg Abandonment Time	The average time a caller is in the queue before hanging up displayed in minutes and seconds. Avg Abandonment Time = Total Abandonment Time/Calls Abandoned
Avg Staff	The average number of agents who are in *Sign-In*, *Available*, *Unavailable*, or *Wrap-up* ACD state for the queue during the reporting interval. Avg Staff = Staff Time/Specified Interval
Queue Performance	
Call Centre	The list of call centres that the supervisor is monitoring and managing.
% Answered	The percentage of calls that were answered by the call centre. % Answered = Calls Answered/Calls Received
% Calls in Queue	The percentage of calls that is in the queue. % Calls in Queue = Calls In Queue/Calls Received The summary row is not applicable to this column and always shows "–".
% Abandoned	The percentage of calls that was abandoned while waiting in the call centre queue. % Abandoned = Calls Abandoned/Calls Received

Statistic	Description
% Within Service Level	The percentage of calls that were answered within the designated Service Level of the call centre. % Within Service Level = Number of calls answered within service level / Number of calls answered in the interval + number of calls abandoned in the interval.

Service Level Report

The Service Level Report is also known as a Grade of Service Report and displays metrics related to how long callers wait before being connected to an agent. The heading for this report is Service Level – Call Centre Name – Interval Report, for example, Service *Level – Support – Half Hourly Report*. This can be changed to view All Call Centres.



			Service Level 1 00:07		Service Level 2		Service Level 3 00:23		Service Level 4 00:40		Service Level 5 01:15	
Date and Time	Average Wait Time	Average Speed Answer	Calls Handled	% Within Service Level	Calls Handled	% Within Service Level	Calls Handled	% Within Service Level	Calls Handled	% Within Service Level	Calls Handled	% Within Service Level
Oct 21, 20:00	-	-	0	0%	0	0%	0	0%	0	0%	0	0%
Oct 24, 20:00	00:00:09	00:00:15	1	11%	1	11%	1	11%	0	0%	0	0%
Oct 25, 20:00	00:00:11	00:00:22	0	0%	2	40%	1	20%	0	0%	0	0%
Oct 26, 20:00	00:00:10	00:00:16	4	33%	3	25%	0	0%	1	8%	0	0%
Summary	00:00:10	00:00:17	5	11%	6	19%	2	7%	1	2%	0	0%

Chart Information for Service Level Report

The following list describes the graph information in the Service Level Report, from left to right:

Statistic	Description
Service Level (Line Chart)	
% Within Service Level	The percentage of ACD calls that were answered by an agent within each service interval (this is the acceptable service level).

Table Information for Service Level Report

The following table describes the columns in the Service Level Report. The columns are repeated according to the number of service levels selected:

Statistic	Description
Service Level	
Call Centre	The list of call centres that the supervisor is monitoring and managing.
Average Wait Time	The average wait time in queue during the specified interval. This is repeated for each service level. Average Wait Time = Total Queue Time/Calls Answered
Average Speed Answer	The average speed to answer during the specified interval. This is repeated for each service level. Average Speed Answer = (Total Queue Time + Total Ring Time)/Calls Answered
Calls Answered	The number of ACD calls answered by an agent during the specified interval. This is repeated for each service level. The number increments when the call is answered.

Statistic	Description
% Within Service Level	The percentage of ACD calls that were answered by an agent within each defined service level. % Within Service Level = Number of calls answered within service level / Number of calls answered in the interval + number of calls abandoned in the interval. **NOTE**: The service level time starts when a call enters the queue, including time listening to a greeting.

2.29 TECHNICAL DATA

Unity requirements and prerequisites:

Please see the latest version of the Unity Requirements document for details of:

- Network requirements and prerequisites
- Software / operating system requirements
- PC hardware requirements
- Firewall requirements

This document can be found at:

http://www.redcentricplc.com/downloads

Call centre capacities:

Maximum number of agents in a queue	50
Maximum number of queues per Supervisor	15
Maximum number of agents per Supervisor	20
Maximum number of Supervisors per agent	1 or 2
Queue (ACD) Guard Timer	5 to 10 seconds
Maximum queue (ACD) size	50 (Call Centre Standard) 525 (Call Centre Premium)

Call Centre limitations:

It should be noted that the following limitations apply to the Call Centre Premium features:

Wrap up state	▪ Agents do not receive personal phone calls when in Wrap up state

	▪ Agents do not receive phone calls when in Wrap up state in a High-volume call centre ▪ Agents can cause Wrap-up issues if they set their Do not Disturb status manually
Reports and statistics	▪ Basic Reporting statistics are only stored for 48 hours. ▪ Basic Reporting statistics are only visible to the Group Administrator. ▪ Advanced Reporting statistics is stored in full for 180 days. After 180 days, half hourly statistics are then compacted into hourly statistics. Hourly statistics are then stored for a further 365 days at which point hourly statistics are further compacted into daily statistics. These daily statistics are then deleted after a further 365 days. ▪ Advanced Reporting statistics are only visible to a licensed Call Centre Supervisor via the Call Centre PC desktop or web application.
Remote Office	▪ Call centre cannot be used with the Remote Office feature
Call Forwarding	▪ Call centre cannot be used in conjunction with Call Forwarding
IP Handsets	▪ The application will not function correctly unless the agent or supervisor logged in is also registered to an active handset

Music on Hold file format and specification

Group Music on Hold	**Required File Format and Specification** The audio file format and specification must adhere to the following: • CCITT u-Law • 8.000 kHz • 8 bit mono • .WAV file type • Maximum recording duration 10 minutes • Maximum file size 4.6 MB
Music on Hold as part of Call Queue (ACD)	**Required File Format and Specification** The audio file format and specification must adhere to the following: • CCITT u-Law • 8.000 kHz • 8 bit mono • .WAV file type • Maximum recording duration 5 minutes • Maximum file size 2.4 MB

2.30 UNITY CLIENT – KNOWN LIMITATIONS

- Video calling will not currently work for users who have Unity call recording enabled (please note that this also includes Polycom VVX)

- Call forward not reachable is not usable if mobile data is enabled as devices stay registered over the mobile data network (applicable to mobile versions of the client only)
- File transfer is not currently supported (applicable to desktop versions of the client only)
- Under certain call scenarios, service tones (for example ringing tone) are US tones and not British tones

2.31 UNITY CLIENT – DEPLOYMENT IN A VDI ENVIRONMENT

The Unity application is only supported in a VDI environment with the application installed on the thin client which must be running Windows and configured in the VDI environment as a local access application.

Requirements if running in a VDI environment

- Citrix XenApp 6.5 feature pack 2 or Citrix XenDesktop 7
- Citrix Platinum level license
- Terminal operating system must be Windows

Limitations if running in a VDI environment

- Outlook integration is not available.
- File transfer is not available.
- Hardware capacity on the terminal will determine quantity of supported video and implementing support for cameras with hardware encoding will alleviate this issue.

2.32 TRADITIONAL LINES & CALLS

Overview

Redcentric recognise that the transition to IP may span a period of time, which means that a Customer would otherwise need to deal with two or more service providers as their telecom estate transitions from traditional technology to IP. To simplify the migration, Redcentric offer both IP and traditional services, enabling the provision of an entire, immediate solution followed by a project managed transition to IP.

Customers can migrate the majority of existing analogue and ISDN lines to Redcentric as well as order the installation of new lines. Upon migrating lines, DDI numbers can also be ported to Redcentric so there is no change needed existing business telephone numbers.

Please see Lines and Calls service definition for details of this service.

Note: Redcentric may be unable to migrate lines from certain providers.

2.33 UNITY OFF-NET SERVICE

Unity Off-Net enables Unity IP Voice to be delivered to supported Polycom IP phones across a non-Redcentric network connection.

There are two variants of the Unity Off-Net service:

- Unity Off-Net (automated phone provisioning)
- Unity Off-Net (manual phone provisioning)

These are described below:

Unity Off-Net (automated phone provisioning):

- Unity IP Voice phones connect to the Customer's existing WAN connection via their LAN
- The Customer must provide certain information in their DHCP scope to provide the phones with details of Redcentric's voice platform to facilitate automated provisioning
- Redcentric do not provide any pre-configuration of the phones
- Connection from the Customer site to Redcentric's SBCs, typically over the Internet

Technical Requirements – Unity Off-Net (automated phone provisioning):

- DHCP must be provided on the LAN to which the phones are connected
- Where Redcentric do not pre-configure phones, additional information must be provided in the DHCP scope
- The point of egress to the internet must be capable of running NAT or PAT
- Phones must be allowed:
    - Access to the internet for the purpose of resolving FQDNs via DNS
    - Access to the internet for the purpose of accessing Redcentric's HTTP and HTTPS servers for auto-configuration of the handsets
    - Access to the internet for the purpose of reaching Redcentric's SBCs via the SIP and RTP protocols. (ports are not relevant as there will be PAT in place)
    - Hold a TCP and/or UDP session open at all times. The port will be re-initialised every 60 seconds via means of a re-registration with our SBCs

    - Connect directly to the LAN via CAT5e cable.
    - There should only be wired LAN infrastructure between the handset and the Internet (no wireless links or repeaters)
- Phones must be able to resolve to:
    - portal.ipt.intechnology.co.uk
    - boot.ipt.intechnology.co.uk
    - gateway.ipt. intechnology.co.uk
    - 213.146.130.196 - TCP & UDP 5060, UDP 49152-65535
    - 178.250.96.68 - TCP & UDP 5060, UDP 49152-65535
    - 213.146.130.200 - TCP & UDP 5060, UDP 49152-65535
    - 178.250.96.74 - TCP & UDP 5060, UDP 49152-65535
    - 213.146.135.47 (dms.ipt.intechnology.co.uk) - TCP 80, TCP 443
    - 178.250.100.86 (dms.ipt.intechnology.co.uk) - TCP 80, TCP 443
    - 213.146.135.15 (Auto Prov) - TCP 80, TCP 443

It is recommended / may be necessary for the Customer to disable any form of SIP Inspection on their firewall.

Limitations – Unity Off-Net (automated phone provisioning)

- The maximum number of simultaneous calls that can be made is dependent on the symmetric bandwidth available on the Customer's Internet connection
- Availability of the Unity IP Voice service is dependent on the availability and correct operation of the Customer's Internet and related equipment
- No call quality assurances can be made by Redcentric when the Unity IP Voice Service is delivered across a non-Redcentric network, such as the Internet

Unity Off-Net (manual phone provisioning):

- Redcentric manually pre-configure the phones with details of the voice platform prior to despatch or installation
- Unity IP Voice phones connect to the Customer's Internet connection via their LAN
- The Customer need not provide additional information in their DHCP scope
- Connection from the Customer site to Redcentric's SBCs, typically over the Internet

Technical Requirements – Unity Off-Net (manual phone provisioning):

- DHCP must be provided on the LAN to which the phones are connected
- Where Redcentric do not pre-configure phones, additional information must be provided in the DHCP scope
- The point of egress to the Internet must be capable of running NAT or PAT
- Phones must be allowed:
    - Access to the internet for the purpose of resolving FQDNs via DNS
    - Access to the internet for the purpose of accessing Redcentric's HTTP and HTTPS servers for auto-configuration of the handsets
    - Access to the internet for the purpose of reaching Redcentric's SBCs via the SIP and RTP protocols. (ports are not relevant as there will be PAT in place)
    - Hold a TCP and/or UDP session open at all times. The port will be re-initialised every 60 seconds via means of a re-registration with our SBCs
    - Connect directly to the LAN via CAT5e cable.
    - There should only be wired LAN infrastructure between the handset and the Internet (no wireless links or repeaters)
- Phones must be able to resolve to:
    - portal.ipt. intechnology.co.uk
    - boot.ipt. intechnology.co.uk
    - gateway.ipt. intechnology.co.uk

- 213.146.130.196 - TCP & UDP 5060, UDP 49152-65535
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- 213.146.130.200 - TCP & UDP 5060, UDP 49152-65535
- 178.250.96.74 - TCP & UDP 5060, UDP 49152-65535
- 213.146.135.47 (dms.ipt. intechnology.co.uk) - TCP 80, TCP 443
- 178.250.100.86 (dms.ipt. intechnology.co.uk) - TCP 80, TCP 443
- 213.146.135.15 (Auto Prov) - TCP 80, TCP 443

It is recommended / may be necessary for the Customer to disable any form of SIP Inspection on their firewall.

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- No call quality assurances can be made by Redcentric when the Unity IP Voice Service is delivered across a non-Redcentric network, such as the Internet

2.34 PROFESSIONAL SERVICES

Project Definition Workshop

The project definition workshop is a design session during which a Redcentric technical consultant will design the voice solution with the Customer.

Project definition workshops may be undertaken over the phone, over a web conference, at a Redcentric office or at the Customer site subject to requirements.

Voice Technical Consultancy

Typical examples of voice technical consultations may include:

- Post-installation workshops to change the solution design
- Office moves
- Disaster recovery planning

Voice Technical Consultancy may be delivered over the phone, over a web conference, at a Redcentric office or at the Customer site subject to requirements.

Voice Project Management

Project management of voice service implementation. May also include project management of Customer specific requests such as site relocations for example.

Administrator Training

This mandatory course is to familiarise system administrators with the steps required to carry out Customer-managed adds, moves and changes via the web administration portal.

The course objectives are:

- To be able to explain the roles of a group administrator and a user.
- To be able to set up Unity IP Voice for a user using the 'CommPilot' web administrator portal.
- To be able to manage group services.
- To be able to manage user services.

Unity IP Voice Administration Training is delivered at one of Redcentric's training facilities or on the Customer site.

On-site Training

This optional consultancy service places a Redcentric voice training specialist onsite. Typical requirements for onsite training include:

- **Floor-walker training** – provides users with hands-on training, typically on or near to the day the new service goes live. It ensures staff are comfortable and proficient with the key operations and features of the Unity service and provides an on hand resource to answer any user queries.
- **Receptionist training** – focused training on ensuring receptionists are comfortable and proficient handling and fielding inbound calls using the Receptionist Console application.

- **Call Centre training** – for call centre deployments, provides hands-on training to agents and supervisors on the use of the Call Centre solution and applications.
- **Train the trainer** – a popular approach whereby the Redcentric trainer provides detailed end-user training to a number of Customer-designated 'power users'. These power users then provide onward training and assistance to end users.

Engineer Installation – During Business Hours

Provides physical installation and testing of phones during business hours.

Business hours are defined as Monday to Friday 0900 to 1730 excluding public holidays.

Engineer Installation – Outside of Business Hours

Provides physical installation and testing of phones outside of business hours.

Out of business hours is defined as 0500 – 0900 and 1730 – 2130 and is subject to a minimum of four hours' work.

Failed Installation Appointment Fee

A charge made where an installation appointment is cancelled with less than 24 hours' notice, where an engineer is unable to gain access to site or where an engineer is unable to carry out the installation due to a lack of site readiness.

Adds, Moves & Changes Administration Fee

A charge made for each administrative request undertaken by Redcentric's support team where the request is defined as a Customer manageable add / move / change.

Travel and Expenses for Professional and Installation Services

Pricing for professional services and installation services does not include travel and expenses which are chargeable separately.

Within mainland UK, travel and expenses costs are limited to 15% of the total professional services charges. Outside of mainland UK, travel and expenses costs are priced on application.

2.35 SYSTEM PROVISIONING

Within the Unity IP Voice core platform Redcentric will create a dedicated and secure Administration Zone for the Customer. This will allow an authorised administrator to log onto the Unity IP Voice system in order to configure, manage and view all aspects of the service.

Subject to the Administrator having attended the recommended Unity IP Voice- System Administrator training course, Redcentric will provide initial help in setting up and establishing the key System Administration parameters plus help with any problems / issues relating to the set up.

2.36 SYSTEM MAINTENANCE

Note: Redcentric may perform remote handset security checks and firmware upgrades during the hours of 8pm – 3am Monday-Sunday. The checks/upgrades will force a reboot of the end user's handset. In the unlike event of a user not being able to make a call the affected user should in the first instance power cycle the handset and if this doesn't clear the issue follow normal support process and raise a ticket with the Redcentric support desk.

2.37 CUSTOMER MANAGED ADDS, MOVES AND CHANGES

It is expected that all of the following adds, moves and changes will be undertaken by the Customer via the Unity web administration portal. Where the Customer requests that Redcentric undertake any of the following, this will incur an administration fee, per change.

Configuration and administration of the following Group Services and Features:

- Administrator passwords
- Hunt groups
- Call Queues (Call Centres)
- Auto-attendants (IVRs)
- Call pickup groups
- Incoming and outgoing call plans (inline with Redcentric's voice security policy unless otherwise agreed in writing)
- Password rules plans (inline with Redcentric's voice security policy)
- Voice portal pass-code rules (inline with Redcentric's voice security policy)
- Outgoing digit plan
- Custom contact directories
- Common phone lists
- Departments
- Holiday schedules
- Time schedules
- Authorisation codes
- Music on hold
- Group / departmental CLI (telephone number) presentation options
- Group voicemail (where applicable)
- Feature access codes
- Voice portal options

Configuration and administration of the following user services and features

- User passwords and voice portal pass-codes
- User voicemail
- User services and features
- User PC desktop applications

Management and administration of the following resources:

- Phones (devices)
- Numbers and their allocation (telephone numbers)
- User service pack allocation and availability

- Group service licence allocation and availability
- Recording of and uploading of voice prompts and announcements

2.38 STANDARD DELIVERABLES

The following are deliverable as part of the Unity IP Voice service delivery:

- Presales technical consultation to design solution
- Completion of PST document (which defines the designed solution)
- Provision of the Unity service as per agreed PST
- Installation and testing of the phones within normal business hours (as part of engineer installation)
- User and phone documentation in soft copy (in English language only)

2.39 CHARGEABLE SERVICES AND EXTRANEOUS CHARGES

The following are not included as part of the standard Unity IP Voice service delivery, but are available as optional, chargeable services / items:

- Administrator training - mandatory for at least one person
- On-site training
- Voice technical consultancy beyond the consultation included to design solution
- Excess engineer time (incurred through additional onsite works required, Customer delays or lack of site readiness)
- Out of business hours installations
- Failed appointments
- Fixed cabling services
- Patch cables (other than those included with phones)
- Cabling accessories
- Adds, moves and changes undertaken by Redcentric (where such adds, moves and changes are defined as Customer managed) – see "Customer Managed Adds, Moves and Changes" section
- Follow up technical consultations such as redesign workshops
- Recording and provisioning of recordings and prompts – a dial-in facility can be provided for Customers to record their own prompts and announcements which will then be uploaded by Redcentric's support team. This is chargeable as an Adds, Moves and Changes administration fee (N-IPVC-142).

2.40 RESPONSIBILITIES

Customer Responsibilities

- Under the OFCOM Condition 4 of the General conditions of Entitlement relating to site location information for emergency services:
    - Provision of accurate site location information for each telephone number provided as part of the Unity IP Voice Service
    - Immediate notification of any changes of address associated where voice services are provisioned
    - Advising where users are nomadic and work from no fixed location
    - Notification where users change from working between a fixed location and being nomadic, or change from being nomadic, to working at a fixed location

- LAN Design, capacity, performance and availability
- LAN port availability, cabling to desk, UTP patch cable supply
- LAN IP addressing, DHCP server availability and configuration
- Provision of required Quality of Service settings on LAN
- Provision of suitable computer hardware, software, equipment and services necessary to connect to and to use the Unity IP Voice service, and for ensuring that all equipment connected conforms to the specifications and routing protocols specified by Redcentric;
- Configuration of any firewall or other Internet limiting device to provide Internet access on required ports to user PCs as specified in order to operate Unity IP Voice software applications
- Local dial plans, definition of extension numbers and ranges, CLI preferences
- ALL PBX systems configuration and ongoing maintenance
- Local (company) dial plans and numbering
- Ensuring Customer administrators are authorised and trained in the use and configuration of Unity IP Voice system
- Identification of any analogue telephone line requirements such as FAX machines, Franking machines, PCs with modems etc.
- Cabling / patching for any analogue devices such as FAX machines etc. to an analogue telephone line (which would usually be installed in the comms/server room).
- Provision, management, maintenance and availability of any non-Redcentric network used for the delivery of the Unity IP Voice service when using the Unity Off-Net service.
- Ongoing self-administration of system settings configurable within Group Admin configuration portal: access to the administration Interface by the Customer's system administrator enables control over certain functionality of the Unity IP Voice Service and the management of accounts including:
    - creation/configuration of authorised users (where required licences have been assigned by Redcentric)
    - enabling/disabling licensed features and;
    - entering and maintaining dial plans, DDIs, extension numbers and CLI settings;
    - controlling voice mail settings;
    - integrating company-wide directories for click to dial features; and
    - setup and configuration of licensed group services such as ACD/Call Centres and Auto Attendant/Name dialling.
- Ensuring the security and proper use of all user IDs and passwords (including those of the System Administrator) used in connection with the Services (including changing passwords on a regular basis) and must take all necessary steps to ensure that they are kept confidential, secure, used only for the proper purpose and not disclosed to unauthorised third parties.

Redcentric Responsibilities

- Unity IP Voice Core system operation, availability & Customer zone implementation
- Quality of Service across Managed IP-VPN VPN and across core network
- UK and international dialling dial plans
- Provision for the Customer's ability to make / receive calls to / from the PSTN
- Supply of any software application required to use the Unity IP Voice Service
- Redcentric reserves the right to access the administration Interface and the information stored by the Customer at any time for technical and operational reasons and to amend, modify and replace the Administration Interface as reasonably required from time to time.

2.41 SECURITY BEST PRACTICES

The purpose of this section is to detail the potential security risks associated with the operation of any phone system and to provide information and requirements on how to minimise these risks when using Unity IP Voice.

Security and privacy must focus on controlling unauthorised access or excessive access to features within the voice platform.

It is imperative that you ensure the correct level of security is implemented on your organisation's configuration to prevent fraudulent use of the Unity IP Voice Service. Fraudulent use can result in additional call costs to your organisation if security measures are not implemented.

It is your organisation's responsibility to ensure adequate security practises/processes are followed and that the appropriate security measures are implemented. The security requirements in the following document are subject to constant review and improvement.

Background: How Phone Systems are attacked

Inherent within any voice system is the potential for abuse. The methods through which a service can be abused include the following:

- Unauthorised remote access to a telephony system in order to:
    - Make expensive calls at zero cost to themselves
    - Make calls to premium rate numbers to fraudulently generate revenue
    - Unlawfully intercept confidential information such as voicemail messages
- Deliberate and direct misuse by staff including:
    - Making expensive calls whilst at work to save on their own bill
    - Fraudulent use of call forwarding – when users misuse their call forwarding services to make long distance or premium rate calls by dialling their own number from outside the office
    - Calling premium rate numbers to generate revenue for themselves / another
- Inadvertent direct misuse by staff including:
    - Being coerced or tricked into dialling premium rate numbers
    - Being coerced or tricked into providing a service for fraudsters (such as through call forwarding)
- Deliberate remote misuse by staff including:
    - Dialling into the phone system from outside the office to make non-business or fraudulent calls
    - Accessing the system using a voice service web portal or software application to make non-business or fraudulent calls

Hackers are constantly looking for ways to access telephony systems via a user's account with a view to scanning the options for a means of making outbound calls. The Customer ends up bearing the costs of these calls, which are often made to expensive premium rate or international numbers. Details of how to hack into telephone systems are even posted on the Internet.

Ensuring that each user's password is hard to deduce is the primary method to combating the fraudsters.

In the event that a hacker accesses a system, fraudulent abuse may take the form of:

- Accessing a user's account and establishing an onward call from the system.
- Accessing a user's account, setting the call forwarding to an external number (such as a premium rate dialling service) and then calling the user's number at a local rate.
- Interception of the user's confidential voicemails.

All of these attacks are preventable with the adoption of a simple voice security policy. It is imperative that adequate security is implemented on your organisation's telephony configuration to prevent this.

It is important that the following configuration elements including the definable policies are included within your security policy.

1. Group Administrator and end-user Passwords
2. End-user voice-mail passcodes
3. Outgoing Calling Plan
4. Making outgoing calls through the voice-mail portal
5. Housekeeping (managing new and redundant user accounts)

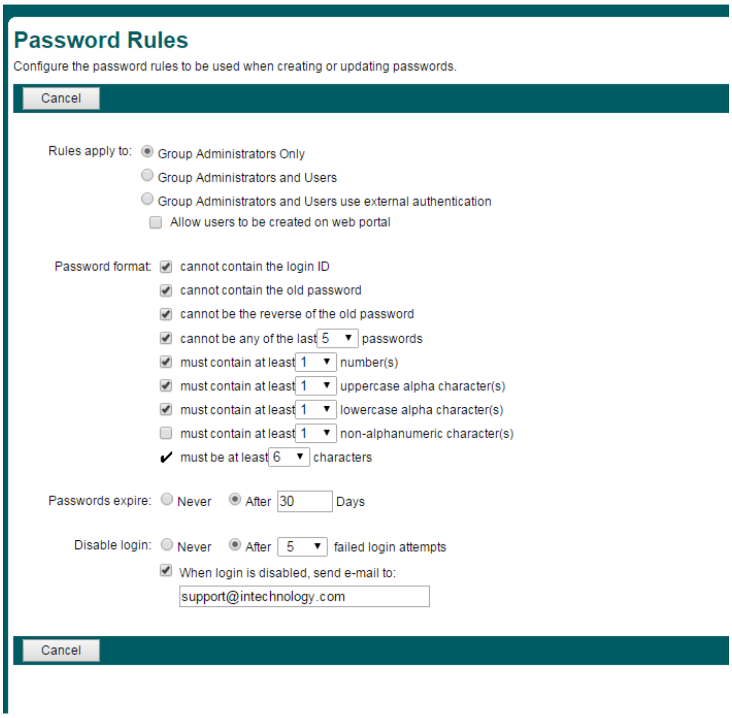
Group Administrator and End-User Passwords

The Group Administrator (responsible for management of the Unity IP Voice Service at Group level) and users require passwords for access to the web admin portal. This portal provides the group admin with administrator privileges, allowing them access to manage the Unity IP Voice Service and configure users and features. This portal provides users with access to configure and manage their own individual features. The user passwords are also used by users to log in to the voice service software applications.

Unauthorised access to the portal at either group or user level would have significant security consequences and as such Redcentric require the following:

1. Passwords must contain at least one capital letter and one number.
2. Password ageing is enabled by default at system level and will require a signed Customer waiver prior to Redcentric disabling this feature.
3. The limit for password retry lock-out is set to 3 or less to prevent multiple systematic retries.
4. The e-mail address to which account lock-out notifications are sent is constantly monitored.

A screenshot of the password rules screen in which the group administrator can manage password rules follows:

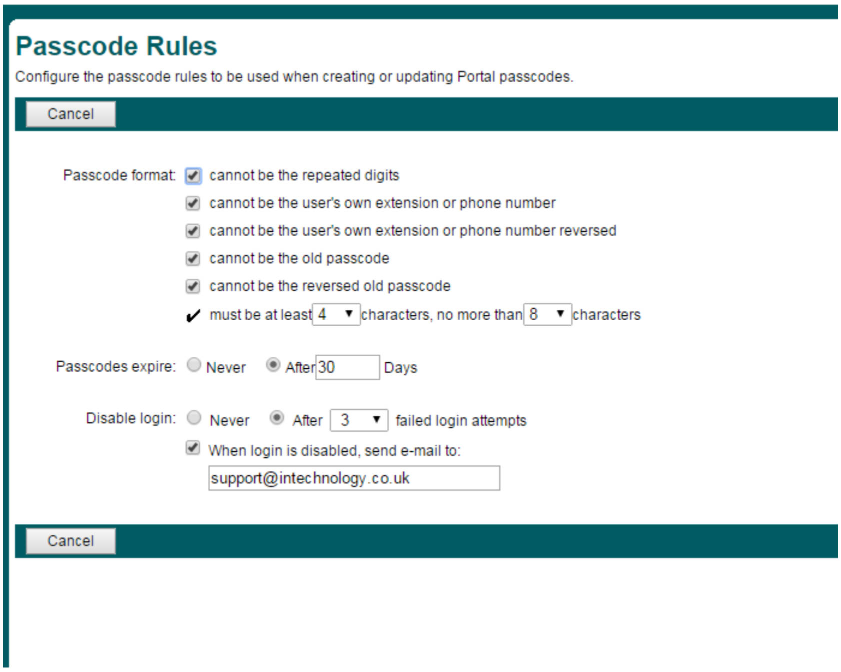


End-User Voice-Mail Passcodes

The easiest way for hackers to access a telephone system is by selecting a user number then trying 'easy' passcodes to gain access to the menus. Because of this, Redcentric require the following points are followed:

1. Passcodes must not be repeating digits such as "1111" or "2222" etc.
2. Passcodes must not use "1234" or use their extension number.
3. The limit for passcode retry lock-out is set to 3 or less to prevent multiple systematic retries.
4. Ensure the e-mail address to which account lock-out notifications are sent is constantly monitored.

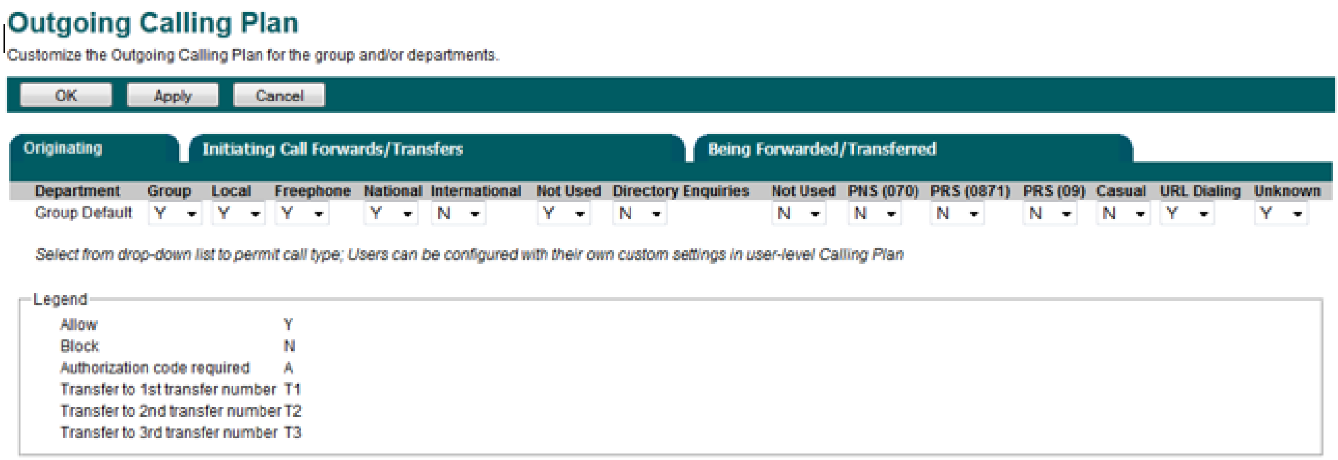
A screenshot of the passcode rules screen in which the group administrator can manage passcode rules follows:



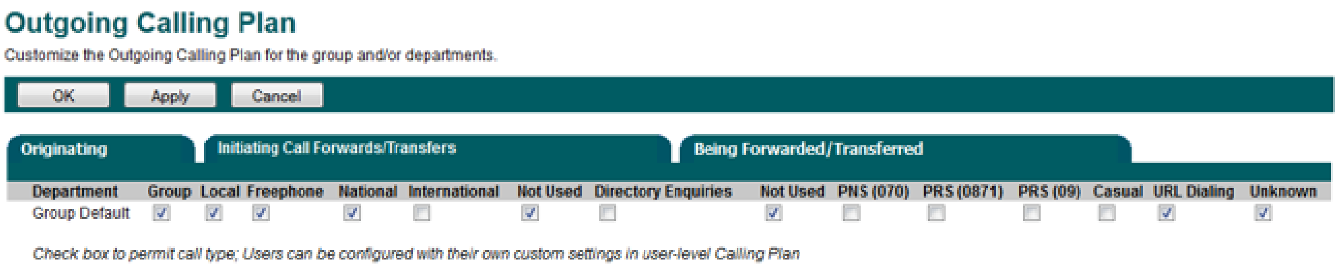
Outgoing Calling Plan

The Outgoing Calling Plans (OCP) dictate what types of telephone numbers can be dialled by users. This includes originating (making) calls, and what numbers users can forward or transfer calls to. The OCPs are set and managed by the Group Administrator via the web admin portal. This feature can specify calling plans for the entire group, or individual departments. Redcentric strongly recommend that only the minimum number of call types is permitted. **Redcentric also recommends that calls to premium rate numbers and international destinations are barred.**

Screenshot of the Outgoing Call Plan screen for originating calls below:



Screenshot of the Outgoing Call Plan screen for initiating call forwards / transfers below:



Please note that the Outgoing Calling Plan can be configured down to a specific user level. It is therefore important to ensure that Outgoing Calling Plans are carefully managed and maintained by the group administrator.

Default Outgoing Calling Plan

Redcentric will provision the Unity IP Voice Service with a default Outgoing Calling Plan. This is configured as below in the interests of security. Customers requiring the ability to call destinations prohibited by the default Outgoing Calling Plan will need their group administrator to enable the required destinations.

Group Calls (calls within the Unity Group) – PERMITTED

Local Calls – PERMITTED

National Calls – PERMITTED

Freephone Calls – PERMITTED

International Calls – NOT PERMITTED

Directory Enquiries – NOT PERMITTED

PNS 070 (Personal Number) – NOT PERMITTED

Premium Rate (0871) – NOT PERMITTED

Premium Rate (09) – NOT PERMITTED

Currently there is no option to bar mobile calls which are permitted.

Restricted International Destinations

In the interests of minimising the risk of fraudulent calls, Redcentric has barred access to a number of high-risk international destinations. The current list is defined below:

- Pakistan 0092
- Nigeria 00234
- El Salvador 00503
- Somalia 00252
- Lithuania 00370
- Guinea 00224
- Ghana 00233
- Egypt 0020
- Cuba 0053

Please note that these destinations are barred at a system level. Even with international calls permitted by the group administrator on the Outgoing Calling Plan, these destinations will still be barred.

If the Customer needs to make calls to any of these high-risk international destinations, barring can be removed by submitting a request to Redcentric support. Note that this will remove barring to all of the high-risk international destinations as defined above. It is not possible to permit / bar individual high-risk international destinations.

Making outgoing calls through the voice-mail portal

This feature allows staff to make phone calls through the same portal that they use to pick up their voice-mails. This can beneficial to certain types of employee but also represents a security risk because it allows outgoing calls to be made by anyone that has been able to guess an employee's voice-mail passcode.

Because of this, Redcentric do not enable this feature on any user accounts and it is Redcentric's strong recommendation that this service remains disabled. Due to the significant scope for fraudulent use, Redcentric will request a written waver of liability from the Customer before re-activating this service.

Authorisation Codes

The Authorisation Codes feature in Unity allows an administrator to enforce that a user enters a security code before they are permitted to make a call. This is primarily intended for public area phones but may also be used to prevent calls being made from any phone without first entering a security code.

Housekeeping

On new installations, new user accounts are normally created by the Redcentric service delivery team. Customers are also able to change passwords away from those initially set by Redcentric at any stage after deployment of the IP Unity Voice Service. Customers may also configure new user accounts following installation after purchasing the user licences. Passwords and passcodes that are generated must always be secure (i.e. not 0000, 1234 etc.).

A redundant account with an easy to deduce password is the ideal vehicle for a hacker to fraudulently use the system and could remain undetected for a long time. The former employee could also continue to use the account (or pass on the details) at the organisation's expense. Redcentric recommends:

- Disable all visitor accounts when not required.
- Ensure that test or demo accounts are disabled when not in use or that they follow the strict password regime mentioned above.
- Remove all leaver accounts immediately or change the password.

Supplemental Security Recommendations

- Remove or de-activate all unnecessary system functionality.
- Review your bills regularly to spot any increases in call volumes or calls to suspicious destinations.
- Disable all surplus user accounts until you have a user for them.
- Only give individuals the appropriate and minimum level of system access they need to carry out a specific task i.e. Service Packs.
- Restrict access to your core communications equipment, such as your main computer room to avoid physical access to network switches. Hackers can intercept calls and reroute them, capture the data for later playback of the conversation or listen in on the call to capture the conversation in real time.
- Ensure that network switches are deployed within secure locations as opposed to network hubs to remove the threat of eavesdropping from within the network using PC based devices.
- Ensure that telephony devices are segmented from data devices by using the appropriate VLAN configurations
- Continually review these security recommendations at regular intervals.

3. IMPLEMENTATION AND ACCEPTANCE

3.1 ACCEPTANCE CRITERIA

- Handset registers to the Unity IP Voice platform
- Internal calls can be made and received
- External calls can be made and received (in line with user's calling plan rights)
- User voicemail service can be accessed (where licensed and where deployed)
- Unity Desktop Assistant installed, and user can log in (where licenced and where deployed)
- User training completed (where purchased and where applicable)
- Customer administrator training completed
- Customer administrative access provided

4. SERVICE LEVELS AND SERVICE CREDITS

4.1 SERVICE LEVELS

The Service Levels applicable to the Unity IP Voice Service are as follows:

Service Level: Measurement Period: Month	
Measured element	Service Level
Availability of the core Unity IP Voice system	Not less than 99.99%

4.2 FLOOR SERVICE LEVEL

The Floor Service Level applicable to the Unity IP Voice Service in respect of Availability of the Core System is 85.0%.

4.3 SERVICE CREDITS

The Service Credits applicable to the Unity IP Voice Service shall be calculated as follows.

The formula for calculating the Service Credits shall be:

$$\text{Service Credit} = \frac{C \times S}{MS}$$

Where:

S = the number of seconds by which Redcentric fails to meet the Service Level for Availability in the relevant Month

C = total Charges payable in respect of the Unity IP Voice Service for the same Month

MS = total number of seconds in the same Month

5. DATA PROCESSING

5.1 SCOPE

- Redcentric does not access, alter or use any application data that is running on the Unity IP Voice Service except as specifically stated below.
- In terms of operating the Unity IP Voice service, API commands are passed into the IP Voice associated supporting servers to orchestrate the build/management of identified users that have subscribed to the service.
- Admin users that have the appropriate role/privileges assigned to them can access the service via a secure web portal to manage their company configuration.
- End users that have the appropriate role/privileges assigned to them can access the service via a secure web portal to manage their individual configuration.
- The agreed roles and responsibilities are provisioned based on documented Customer requirements.

5.2 DATA STORAGE AND UNENCRYPTED DATA

- All Unity IP Voice service associated data is stored within Redcentric's privately owned and managed data centre facilities.
- All access to data within the Unity IP Voice service is via secure portal.
- All access to data is restricted to Customer identified users.

5.3 DATA PROCESSING DECISIONS

- In the normal course of business Redcentric does not make any data processing decisions in relation to the Unity IP Voice Service. Processing is automated and instigated by the Customer.
- Redcentric Support can be asked by the Customer to intervene in the event of an issue with the Service. In such a case Redcentric may make decisions that can affect data processing, but such actions will only be undertaken at the request of and in conjunction with the Customer.

5.4 SERVICE CONFIGURATION WITH RESPECT TO DATA

- The initial service configuration is built using a combination of Redcentric and Customer provided information.
- As data controller, Redcentric holds the following information on Users on Redcentric's BroadWorks telephony service delivery platform:
    - Company Information: Company Name, Address,
    - Account Admin User Data: First name, Last Name, Email Address, Phone Number, Company Address
    - User Data: First Name, Last Name, BroadWorks Username, Optional: Email Address
    - Call Metadata: Party A, Party B, Call Length, and other SIP attributes or the CDR: Time and Date, BroadWorks Enterprise ID, GroupID and UserID, Call TrackingID and CallID.

5.5 DATA BACKUP

- Redcentric performs daily database data backups.
- Backup data is encrypted during transit.

- Backup data is encrypted whilst in storage.

5.6 SUB-PROCESSORS

- No other parties are involved in delivering the Unity IP Voice Service, and there are no sub-processors appointed by Redcentric.

5.7 CUSTOMER ACCESS TO DATA

- The Customer has login rights to the Call Reporting service via secure web portal.
- Access to the Unity IP Voice Service is based on roles and responsibilities defined by the Customer as part of the service setup.
- Redcentric can access reporting data, but would only do this after a formal support request by the Customer/authorised user.

5.8 SECURITY ARRANGEMENTS AND OPTIONS

- Customers have access via a secure portal to manage their own Unity IP Voice Customer level configuration, but they are unable to interact directly with the back-end systems to modify any service wide configurations.
- Customer access to the portal uses role-based access controls (RBAC), integrated with Redcentric core voice platform
- All locations meet physical security standard ISO27002 section 11.1 or equivalent.

HARROGATE (HEAD OFFICE)

Central House
Beckwith Knowle
Harrogate HG3 1UG

THEALE

2 Commerce Park
Brunel Road
Theale, Reading
Berkshire RG7 4AB

CAMBRIDGE

Newton House
Cambridge Business Park
Cowley Road
Cambridge CB4 0WZ

READING

3-5 Worton Drive
Reading
Berkshire RG2 0TG

LONDON

John Stow House
18 Bevis Marks
London EC3A 7JB

INDIA

405-408 & 410-412
Block II, 4th Floor, White House
Kundan Bagh, Begumpet
Hyderabad 500016

0800 983 2522
info@redcentricplc.com
www.redcentricplc.com

redcentric
business technology. managed.

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