Pure Cloud Solutions Hosted Voice Services
Client Advisories & Best Practices

This document provides advisory information to any Customers who wish to deploy the PCS Voice Services, covering mandatory configuration requirements, best practices and guidance on common pitfalls.

Due to the importance of Call Quality and Availability of Voice Services when deployed with an organisation, and bearing in mind the plethora of different phones, routers and IP connectivity options available, PCS are unable to support certain installations if they do not conform to the practices defined in this document.

PCS are able to provide a Total Voice Solution which provides supported Voice IP Lines with full Quality of Service controls, qualified phones, routers and switches. In the majority of cases we would recommend that this option is evaluated and undertaken by all customers to maximise the quality of service delivery. Where this option has been undertaken for a site, all the best practices in this document will already be engendered into the overall solution.

PCS do however recognise that in some cases it may not be cost effective to install a Total Voice Solution in all sites, such as where roaming or home based users require access. To this end the document provides information to support these installations, but also sets expectations accordingly.

All Customers considering deploying PCS Voice Services are strongly advised to read the following.

Use of Third Party IP Data Lines

The real-time nature of a telephone call requires an underlying transport infrastructure that is error free, un-congested and consistent in its delivery of voice packets. If one or more of these characteristics cannot be maintained, then the quality of voice call may be compromised. In terms of the infrastructure that is specific to a Customer, this covers the point from the phone to the PCS service, encompassing customer network, router, IP Data Line and the public internet route to the PCS service. Particularly problematic are:-

- Congested IP Data lines where Voice is shared with Data (e.g. Web traffic, streaming media)
- Congested customer networks where Voice is shared with Data (e.g. Web traffic, streaming media)
- ADSL Lines with contention ratios that "squeeze" data and provide low upstream bandwidth
- ADSL Lines which are data oriented and use techniques such as Interleaving
- Bonded ADSL Lines

More information on the topic of call quality and the issues to be aware of can be found at: http://natterbox.com/twiki/bin/view/Natterbox/WhatAffectsCallQuality

As part of the PCS Total Voice Solution, PCS recommend dedicated Voice IP Lines which can be supplied to each site where voice services are required. PCS also recommend the installation of a separate Voice network in the Customer premises to avoid data congestion.

Third Party IP Data Lines	
• **Interleaving:**	Optionally you may ask your provider to disable <u>if only</u> using the IP

Data Line for Voice. This may be inappropriate with some providers.

PCS cannot undertake investigations into reports of poor call quality, where a PCS user utilises a third party IP Data Line. The Customer should be aware that the ability to manage quality parameters of third party IP Data Lines is beyond the control of PCS, and PCS is therefore unable to make any guarantees on call quality for Voice Traffic which is utilising those third party IP Data Lines, including roaming users who may be utilising WIFI Hotspots, 3G, etc.

Firewalls

The PCS Voice Service uses the SIP & RTP Protocols for call handling between your phones and the service. This requires a number of firewall ports to be opened for outbound traffic in order to avoid such problems as one-way calls, or call setup issues in general.

Firewall Port Requirements

- **Outbound SIP Ports:** UDP port 5060 and 5061
- **Outbound STUN Ports:** UDP Port 3479
- **Outbound RTP Ports:** UDP Port 16384 to 32768
- **Outbound DNS:** UDP Port 53
- **Outbound HTTP:** TCP Port 80 and 443
- Customers must not lock down their firewalls to exchange SIP or RTP traffic with only the PCS IP ranges

Customers should be aware that some of the "Works With PCS" recommended phones will require access via port 80 and 443 for phone updates. If you are running an HTTP Proxy Server, you will need to allow the phones to pass through HTTP requests un-challenged!

Routers

NOTE ON DRAYTEK ROUTERS: Due to problems with their SIP implementation the PCS service does not support Draytek routers.

Routers may be provided as separate devices or maybe combined with the firewall. Some routers are sufficiently intelligent to understand the SIP protocol whilst others simply pass the traffic through untouched. In order to correctly route voice calls to and from phones, the PCS Service has to interact with your router's Network Address Translation tables to map IP addresses from the external network to your internal network. The Natterbox PCS Service expects the Router to pass all SIP traffic untouched.

Router Configuration Requirements

- Any SIP ALG (Application Level Gateway) support must be disabled in your router
- Any other SIP translation or modification parameters are reviewed and disabled
- QoS parameters for voice should be enabled.
- Within your network, you must ensure that the IP ranges used are one of the reserved non-routable IP ranges as defined by IANA.
    - 10.0.0.0 – 10.255.255.255
    - 172.16.0.0 – 172.31.255.255
    - 192.168.0.0 – 192.168.255.255

PCS are only able to support deployments using routers listed in the "Works With PCS Routers" section listed at:

http://natterbox.com/twiki/bin/view/Natterbox/WorksWithNatterbox#RouterS

Use of Third Party Phones

The PCS Service is designed to work with the majority of SIP based phones; however it should be borne in mind that each phone manufacturer supports different features and implementations of the core protocol. Whilst PCS endeavour to maintain compatibility with the Telephones and SoftPhones that our Customers choose, we cannot guarantee support of all features and capabilities.

Where a Customer requires use of a phone that is not supported, this will normally be addressed during the Deployment Stage, and if necessary, testing may be performed to qualify that phone.

Unless otherwise indicated, all phones must follow these basic guidelines.

Minimum Phone Requirements

- The Phone must support STUN for NAT traversal and must be configured to use:-
    - **STUN Server:** stun.natterbox.com
    - **Port:** 5060
- The Phone must be configured to send SIP Keep Alive packets
- The Phone must be configured to support SIP Session Timers
- The presented codec list must be configured to a maximum of two to avoid UDP fragmentation; typically G.711 and G.729 (if supported).

Please refer to: http://natterbox.com/twiki/bin/view/Natterbox/WhatIsSTUN for more details on STUN

PCS are only able to support devices listed in the "Works With PCS" Phones section listed at:

http://natterbox.com/twiki/bin/view/Natterbox/WorksWithNatterbox#PhoneS