COMMANDER SIP

PRODUCT GUIDE

February 2020

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WHAT IS COMMANDER SIP?

Commander SIP is the clever way to take advantage of VOIP technology to reduce the cost associated with phone systems. Connecting Commander SIP direct to your IP enabled PBX or connecting Commander SIP Plus to PBX systems that are not IP enabled, saves money on older more expensive PSTN and ISDN phone lines.

Commander SIP is offered as an "Over the Top" (OTT) SIP Trunking service delivered over the Internet without QoS (Quality of Service). Commander SIP is available over any suitable Internet connection.

An optional Dedicated Access service is available where the customer's existing internet service may not be sufficient.

WHO IS THIS DOCUMENT FOR?

This guide is intended to assist customers in the setup, configuration and support of this service. This document is designed for IT professionals who:

- Have suitable expertise in the design, configuration and use of the services provided.
- Have the required security access and administrative control to be able to configure settings as required.
- Are responsible for and able to install and operate various software applications which may be required to enable the customer to enjoy the benefit of the service.

Please note that all content and information provided in this guide is solely for the purposes of illustration and that each individual customer's requirements will vary for each site. It should not be relied upon by any person as being complete or accurate.

BENEFITS OF COMMANDER SIP

The key benefits of SIP Trunking to the end user will be around the additional feature set, flexibility and scalability when compared to traditional fixed line services.

Commander SIP is a scalable, flexible solution. Capacity can be added quickly if required in single or multiple channels from a minimum of 2 channels. (Note: channels are the VOIP term for telephone lines.)

- Cost effective solution on a per channel basis compared to PSTN / ISDN2 / ISDN10/20/30 services
- Can be delivered quickly without the lead times typically associated with new PSTN/ISDN services - (excludes time to complete porting of existing service numbers)
- Business Continuity with disaster recovery options included in all trunks
- Flexible Indial range capability (10, 50 and 100 ranges) along with LNP agreements with major carriers allows you to keep your service number in most cases, allowing you to keep your current extension setup or save on costs if you don't need a 100 range.
- As it's delivered over the Internet it adds flexibility to service delivery. Thereby avoiding expensive, time consuming relocations of all your fixed line services.
- PAYG, included local and national calls or included local, national and mobile call options available dependant on your call usage.
- Customer selectable simultaneous channels to suite their bandwidth availability (Minimum of 2 channels)

- Integrated with Commander Phone with the ability to share number ranges across SIP and Commander Phone
- Single Online Self-management portal through Command Central for SIP and Commander Phone
- Overlay features available to extend the features of the PBX
- Commander Phone extensions to extend capacity of PBX or cater for remote workers or additional sites
- Facilitates self-paced migration from PBX to Commander Phone (Hosted PBX)

BENEFITS OF COMMANDER SIP PLUS

Commander SIP Plus includes an Integrated Access Device (IAD) that provides an ISDN handoff. SIP Plus is useful for older PBX's that are not IP compatible, or where it is preferable to keep the ISDN interface to the PBX.

- Take advantage of all the benefits of SIP
- No need to replace existing PBX hardware

AVAILABILITY

SIP is available over any suitable Internet connection from any provider. However, it is preferred that the broadband connection and any associated fixed line is transferred to or provided by Commander.

CALLING PLANS

Customers have a choice of calling plans. Plans come with either PAYG call rates or included local and national calls or a local, national and mobile call option. Discounts are available for plans on longer contract terms.

 See Product Sheet for plan details:

- Commander SIP = Regular SIP Trunks
- Commander SIP Plus = SIP Trunk with an IAD that provides ISDN hand-off (ISDN2 or ISDN10/20/30)

Three Plan Types

- SIP Connect = Pay as you go (PAYG) rates
- SIP Calling = included standard Local and National Calls
- SIP Premium = included standard local, national and mobile calls

Timed calls billed per second. All rates inclusive of GST. Monthly Access Fee is per channel - minimum of 2 channels applies.

COMMANDER SIP OPTIONAL SMART EXTRAS

FEATURE	INCLUDES
Enhanced SIP user (Required to purchase any other user level Smart Extra's)	Features equivalent to Commander Phone Executive
Mobility (per user)	Smart UC Mobile
CloudQ	2 Call Centre Basic Users + 1 queue
CloudQ Access	1 Call Centre Basic Users
Reception Central	Receptionist Thin Client
Auto Receptionist	Auto Attendant
Hunt Group (Per Group)	Hunt Group
Fax2Email	Fax to Email
10 Number Range	Block of 10 Numbers
50 Number Range	Block of 50 Numbers
100 Number Range	Block of 100 Numbers
Channels (per line)	Concurrent calls

COMMANDER SIP PLUS

The following hardware is used to provide the ISDN handoff to non-IP PBX equipment. Please note that hardware used will be based on business requirements and may differ from the models listed here.

SIP PLUS PRI IAD

CPE Type	Image
AudioCodes Mediant 500 – 1BRI	
AudioCodes Mediant 500L-4BRI	
AudioCodes Mediant 800B-2PRI	
AudioCodes Mediant 800C-4PRI.	

All hardware is available to purchase outright or on an MRO over 24 months.

CONTRACT TERMS OPTIONS

Services are available on 0, 24, 36, 48 or 60 month terms.

EARLY TERMINATION

Early Termination Fee (ETF) applies per channel. The fee is reduced pro-rata per month over the life of the contract. Any MRO's will also need to be paid out.

OTHER FEES AND CHARGES

Setup fees will be waived for customers choosing a contracted term.

Adds, moves and changes will attract a fee per request.

Channel upgrades (increasing number of channels) will be provided at no charge but channel downgrades (reduction in numbers of channels) will attract a fee.

CONCURRENT CALL SETTING

Commander SIP is a Voice over Internet Protocol (VoIP) service. Like all VoIP telephony services, the voice quality is dependent on the bandwidth available of the data access service that is carrying the voice traffic. Each concurrent call will require 100kbps of data in both up and down streams. Commander SIP controls the number of concurrent calls through the number of channels specified for the trunk. The number of channels should be determined by the available bandwidth at the customer's site(s) during the peak busy time. Customers will notice voice quality deteriorate if the available bandwidth is being consumed whether by voice or other data applications.

Customers affected may:

- Reduce the number of SIP channels (concurrent calls)
- Increase the capacity (speed) of their access service
- Purchase a Dedicated Access service from us specifically to carry their voice traffic only

PREREQUISITIES?

Customers must have a suitable Internet connection to carry Commander SIP trunks and the associated voice traffic. A suitable Internet connection will require 100kbps of both up and down stream bandwidth per SIP channel. The Internet connection can be provided by Commander or a 3rd party.

WHAT TO EXPECT

There are a few steps involved in getting a SIP Trunk service up and running. With some planning, we can make the process a smooth one. Here is what to expect at each stage.

QUOTING AND ORDERING

What You Do	What We Do

Quoting

Discuss with us:

- If you have checked that your PBX is SIP Trunk capable / upgradeable (confirm with vendor) or you intend to use SIP Plus.

- How many SIP Trunk channels are required, call plans, any extra features.

- Do you need to port any phone numbers?

- How long you would like the service for initially

- Any networking or hardware requirements.

We will:

- Listen to your needs and provide you with a quote for the most suitable service.

- If the solution is complex or over 30 channels one of our UC Specialists will assist in reviewing your requirements.

If you would like to proceed with the quote, then we may need some further details:

- Company entity, address & ABN

- Contact details:
  - o Your authorised representative
  - o Billing Contact (the person to send invoices to).
  - o Technical Contact

We will:

- Liaise with you to collect any other Customer Requirements.

- Send you an order form & porting form (if required)

**Order Processing
1- days from receipt**

You should:

- Ensure the order form is correct and all the details are as per your requirements.

- Sign and return the order form.

- Provide information for porting.

We will:

- Confirm we have all the details needed to start delivery of your service.

- Confirm order acceptance within 2 business days of receipt of order.

- Begin delivery of the service.

SERVICE DELIVERY

	What You Do	**What We Do**
Onboarding **2- Days from order**	• Provide any missing technical information from the initial order details.	• Send you confirmation of order receipt • Send you SIP Trunking details for you to configure your trunk or provide to your PBX technician • Configure & deliver required hardware. • Arrange number porting cutover (generally done after the service has been delivered). • Arrange any access service to be delivered (Access can take 5 to
Number Porting **30- days from order**	• Advise of the best time and method for porting and if any changes are required. • Engage your PBX technician to be involved in the number porting.	• The onboarding and porting teams will make contact to organize the best time for porting cutover and method. • Divert numbers (If required). • Confirm SIP Trunks are configured and working as per requirements.

Your Service is Now Ready to Use (Billing will now commence)

	What You Do	**What We Do**
Setup and Operation	Now you can: • Setup & configure your SIP Trunk Service. • If porting isn't complete, you can arrange for diverting numbers to your new SIP Trunk.	We will: • Provide you with support you require in accordance with our Service Schedule. • Bill the service. Your first bill may include pro rata charges plus a month in advance. • If you have other services, they may be invoiced separately.

Note: Times are indicative only and may change without notice depending on various factors.

EMAIL

Regular updates are provided by email. Example of a SIP Account notification email
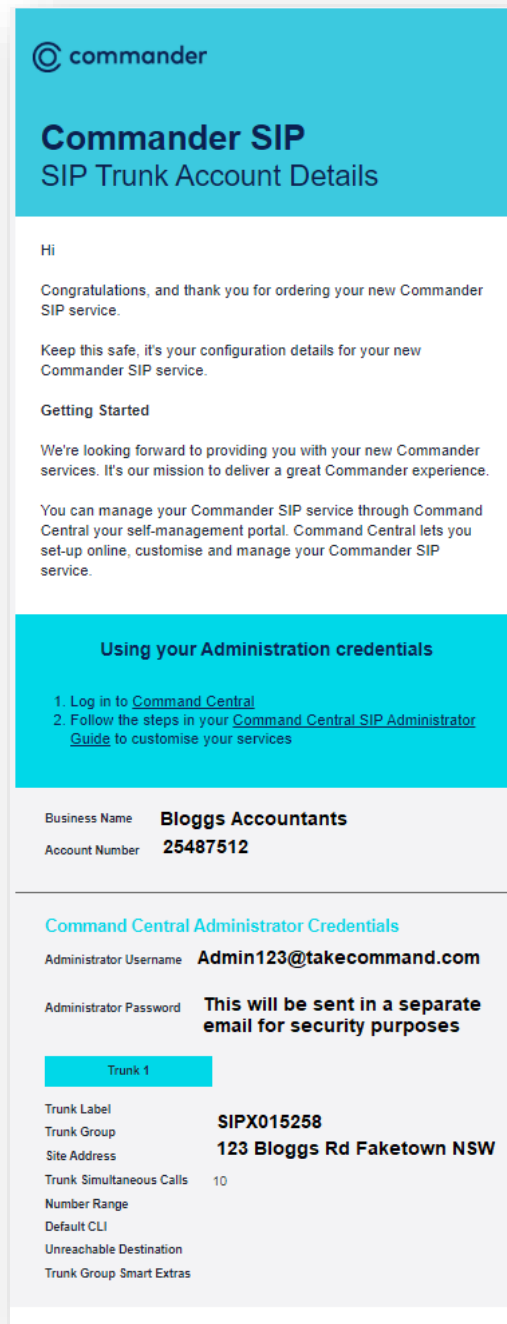
READY FOR SERVICE

When the service has been delivered and tested we will send the technical contact, nominated on the order form, a Ready for Service email.

This email will contain information specific to your customer account and provides your server with the credentials required to make and receive calls across the SIP Trunk.

Please consult your PBX user guide(s) for instructions on how to configure your PBX for SIP using these credentials. If you are implementing SIP Plus, please follow the installation instructions in the Quick-start Guide delivered with the equipment.

SETTING UP YOUR SIP TRUNK SERVICE

This area describes the generic configuration procedures required for an IP PBX to enable voice communications with the Commander voice network.

The IP PBX server uses the Session Initiation Protocol (SIP) to communicate with the SIP Servers in the Commander network to make and receive calls via a SIP Trunk line.

There are many different types of IP PBX and this guide describes the general configuration items that are important for use with the Commander voice network. It does not describe the configuration items specific to your model of IP PBX, for those details please refer to the user guide(s) for your IP PBX.

SOLUTION CONFIGURATION
IP PBX WITH SIP

The following diagram shows a typical installation of an IP PBX. The IP PBX is placed at the customer site with phones associated to it. The IP PBX is configured to register the number range/s for the SIP Trunk service with the SIP Servers in the Commander network.
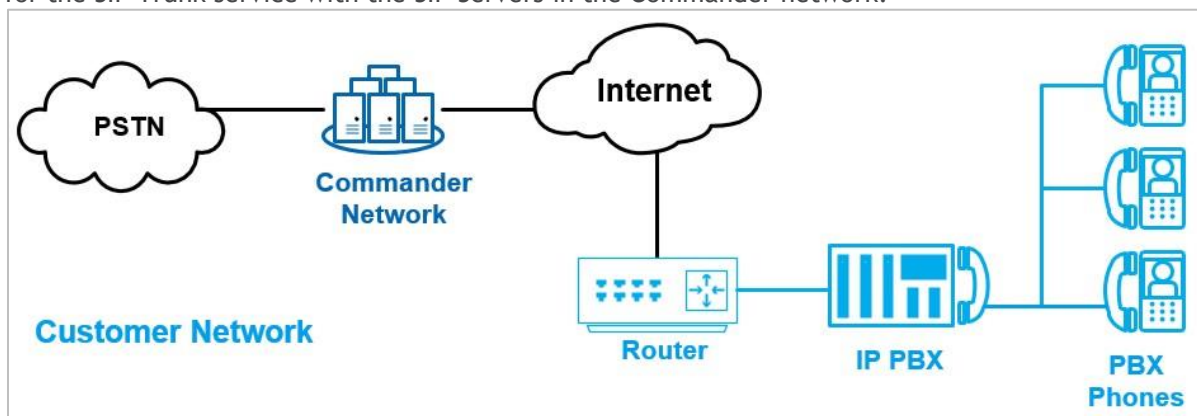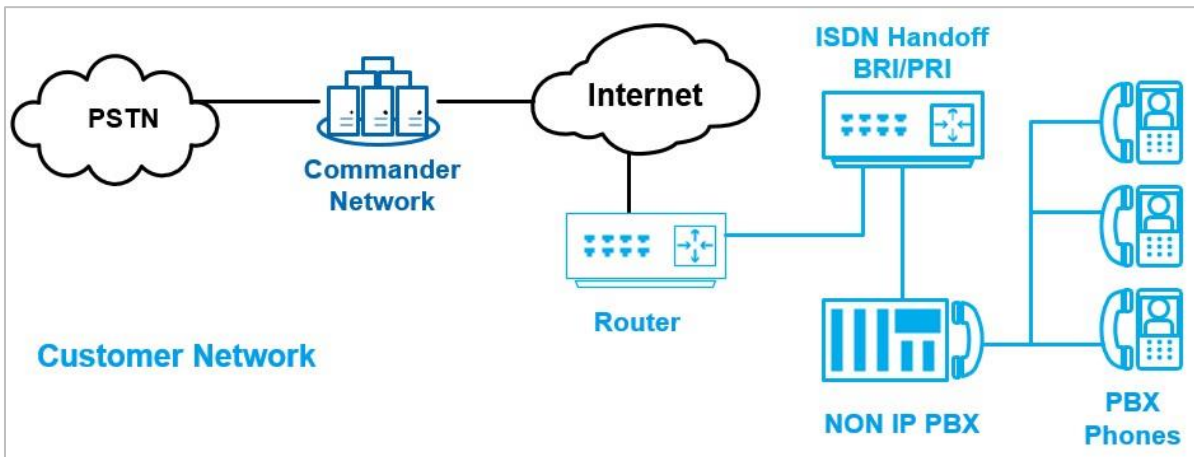

IP PBX WITH SIP

The router shown in the above diagram, could already be in place from an existing service. No router is provided as a standard part of the SIP.

It is recommended that the IP PBX be configured with a dedicated Ethernet interface and a separate Ethernet interface be used to connect to the IP phones and other servers on your private network.

NON-IP (ISDN OR TRADITIONAL) PBX WITH SIP PLUS

For non-IP PBX's an IAD with an ISDN interface is supplied by Commander when SIP Plus is ordered.

NON-IP PBX WITH SIP

For installation of the hardware, please follow the installation instructions on the Quick-start Guide/s delivered with the equipment.

ENTERPRISE TRUNK GROUPS

Enterprise trunk groups are supported where interaction is required between two or more trunks.

Using this methodology SIP Trunk groups can be profiled to suit multiple locations or configured for a failover situation.
Example of a SIP Trunk service for 2 locations:
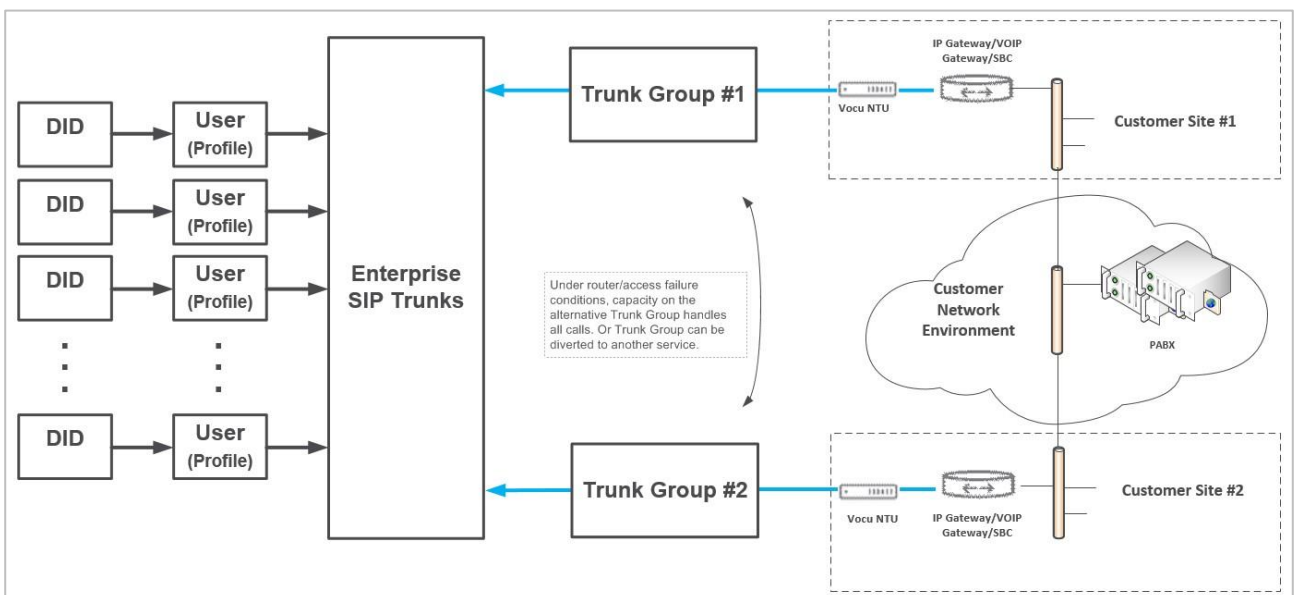
DOMAIN NAME SERVERS

As this product is an over-the-top service, your network should derive up-stream DNS Servers from your ISP.
These servers should then be offered to network hosts (including PABX's) through DHCP.

In the event you are using a Dedicated Access or one of our Internet services, this should be automated. If you need to statically assign DNS servers, they can be obtained from the information here:

STATE	PRIMARY	SECONDARY
Sydney	203.134.64.66	203.134.65.66
Melbourne	203.134.24.70	203.134.26.70
Brisbane	203.134.12.90	203.134.102.90
Perth	203.134.17.90	211.26.25.90

IP ROUTING

If you would prefer your default router on your IP PBX to go via an alternative network connection rather than the Commander voice network, you will need to configure static routes please contact Commander for further information.

PBX DETAILS

Proxy/Registrar:	sipconnect.commander.net.au
Domain Name:	sipconnect.commander.net.au
UDP Port:	5060 or 15060

Notes:
Some versions and some PBX do not support outbound proxy. All our traffic must go through the outbound proxy but in most PBXs that do not support outbound proxy you can manipulate proxy fields to get this same behaviour. (usually by proxy=outbound proxy and domain=sipconnect.commander.net.au).

CODECS

The following is the Codec information required for your PBX.

MANDATORY	SUPPORTED
G.711A	G729
G.711U	

Supported Fax: T.38 (preferred) and G.711.
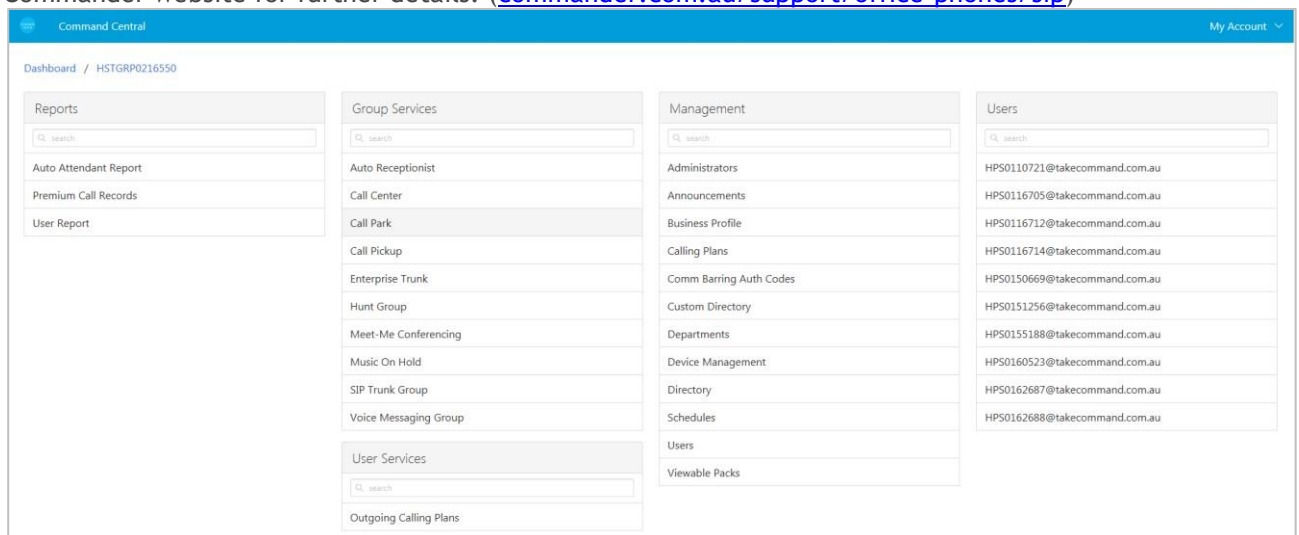
CUSTOMER PORTALS

COMMAND CENTRAL

Command Central is the Commander customer portal for managing all Commander Voice services including SIP, Commander Phone and Commander Key Phone. The one portal makes it easier to manage user and group features across all product types. It can be used on PCs, Tablets and mobile phones so you can manage your IP Voice services on any compatible device. It is a single portal for users and Group administrators granting levels of access depending on the credentials entered.

The portal allows customers to manage their SIP services online including access to:

- View and configure SIP settings
- Manage overflow settings
- Manage Unreachable settings
- Manage Enterprise Trunking
- Forward the trunk to another destination

See the Command Central Administrator Guide available from the support section of the Commander website for further details. (commander.com.au/support/office-phones/sip)

COMMAND CENTRAL DASH   BOARD

TAKE COMMAND

Take Command is the Commander customer portal for managing your Commander Voice account and billing for Commander SIP, Commander Phone and Commander Key Phone.

Take Command allows customers to manage their account online including access to:

- View Commander Voice Services
- Manage Account Details
- See Invoices
- Make Payments
- Setup Usage Alerts
- Create and Manage Cost Centres
- See Reports

Visit takecommand.com.au/#/login/
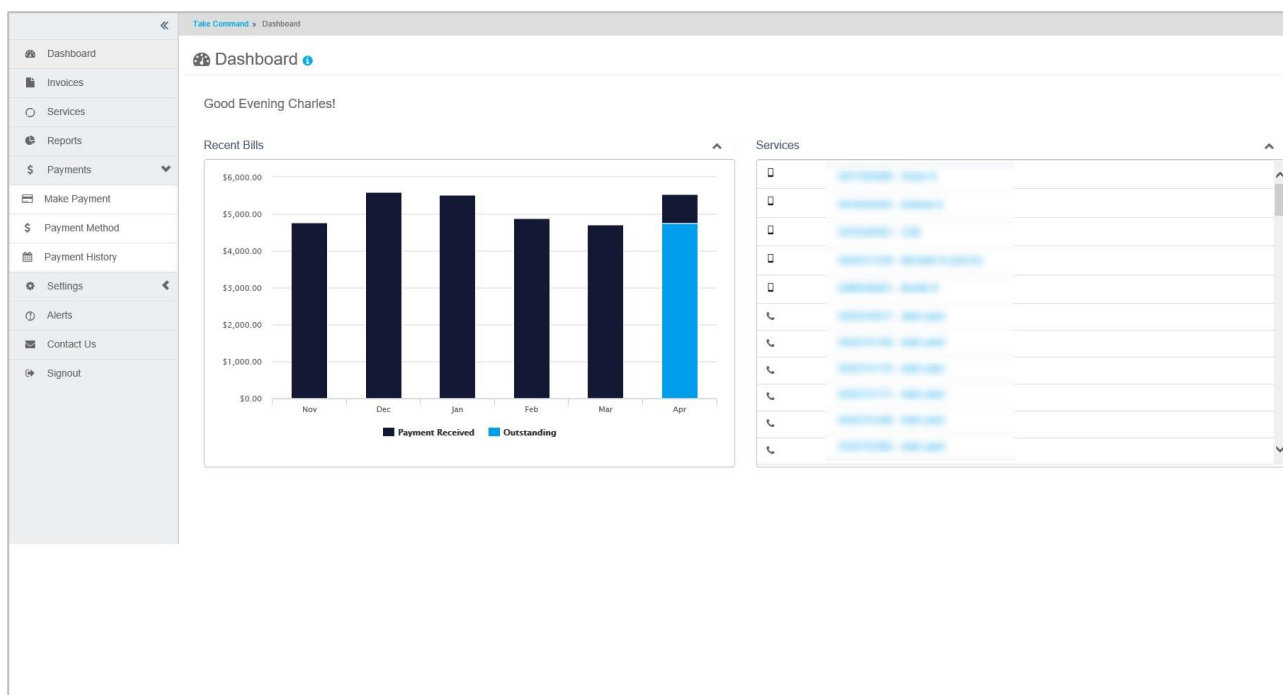
SERVICE SUPPORT

Basic troubleshooting information is provided in this document to assist in fault finding your Commander SIP service.

A couple of items worth considering before starting troubleshooting are:

- Has any work been performed on this service in the last 12 to 24 hours? (Something might have been disconnected)
- Do you have any other services down currently? With Commander or any other provider? (This could indicate a larger scale issue).

TROUBLESHOOTING GUIDE

Please refer to the appropriate flowchart and the information following to perform basic troubleshooting on the SIP Trunk service.

TROUBLESHOOTING IP PBX WITH SIP
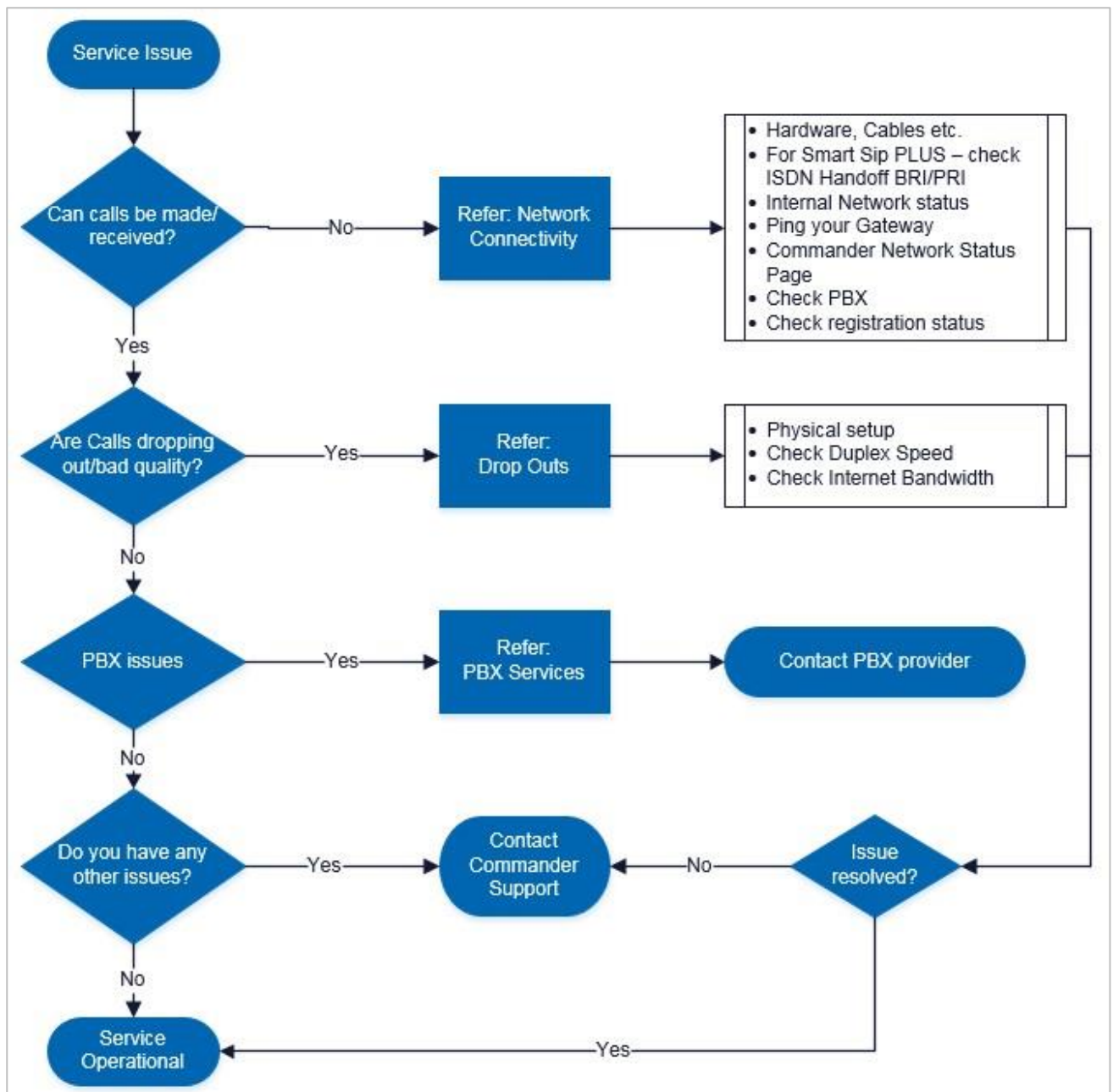
NETWORK CONNECTIVITY

Where the PBX is getting no information from the network and you are unable to make or receive calls.

TROUBLESHOOTING

- Check all physical connections are in the correct locations and have not been moved or dislodged. Test with alternate cables.
- Confirm that your internal network is operating correctly.
- Ping your default Gateway and beyond.
- Check your Internet service provider for any network issues.
- Check the PBX is powered and not displaying any errors. Refer to user manuals provided by the manufacturer for your PBX.
- Check Registration status on the PBX.

DROP OUTS

If you experience call drop outs.

TROUBLESHOOTING

- Check all physical connections are in the correct locations and have not been moved or dislodged. Test with alternate cables.
- Check Duplex/Speed on the interface.

PBX SERVICES

If the PBX is not actioning calls correctly consult your PBX provider.

If basic troubleshooting does not resolve the issue the please contact Commander Support on 132777.

When logging a fault please provide information on the symptoms and the troubleshooting already performed.

NO POWER

No lights at all on the IAD will generally indicate a power issue. Check power supply to the area and the power supply unit.

TROUBLESHOOTING

- Confirm mains power supply.

NO SERVICE

There is power to the unit, but the service is still not operational. There is no green light on the GE WAN light indicating that no network is available.

TROUBLESHOOTING

- Check all physical connections from the network to the IAD are in the correct locations and have not been moved or dislodged. Test with alternate cables.
- Check your service providers status page for any network issues.

DROP OUTS

If you experience call drop outs.

TROUBLESHOOTING

- Check all physical connections are in the correct locations and have not been moved or dislodged. Test with alternate cables.
- Confirm adequate internet bandwidth es available.

PBX SERVICES

If the PBX is not actioning calls correctly consult your PBX provider.

If basic troubleshooting does not resolve the issue the please contact Commander Support on 132777.

FAQ

SERVICE DELIVERY

Q: Can I take my phone numbers with me?

Yes. Number porting is offered as a part of the SIP and SIP Plus products. (Subject to current carrier's policies)

Q: How long does number porting take?

On average it can take 6 – 8 weeks for porting to be completed. (Subject to current carrier's policies) **Q: What are some of the things that can delay number porting?**

- If changes are made to an account during the number porting period (such as diversions/cancelling/activating new numbers)
- Incorrect account details provided on the porting application
- Incomplete porting applications
- Current Carrier actions

Q: Can I use the service before the number port is complete?

In most cases this can be arranged.

BILLING

Q: What is the contract length?

Commander SIP has 0, 24, 36, 48 or 60-month contract options.

Q: Are calls included?

Depending on the SIP product you choose local, national and mobile calls can be included in the monthly charge. (Subject to reasonable use policies)

OPERATIONAL

Q: Will I lose any of my current PBX features?

All the current features of your PBX should continue to operate normally. In fact, with the newer technology you may find more features available than you had before.

Q: What is Toll Fraud and am I liable?

Toll Fraud is PBX hacking where a fraudster compromises a PBX to generate calls without the owner's knowledge. These calls are generally to premium or overseas services and can result in costly charges to the customer, for which they are liable. Security of the PBX is the owner's responsibility and we recommend ensuring that passwords are strong and changed regularly. Please refer to your PBX vendor for the best security measures to apply.

Some measures can be implement including barring international calls to minimise your exposure to this fraud.

TECHNICAL

Q: Will heavy internet usage affect call quality?

You will need 100kbps of available bandwidth per concurrent call (without congestion) otherwise your call quality can be affected. We recommend using a Dedicated Access service so that your voice traffic is not impacted by other Internet usage on the same connection.

Q: My PBX only has an ISDN interface. Can I still use SIP Trunks?

Yes. We can supply an IAD which will connect to the ISDN interface on your PBX.

SUPPORT RESOURCES

GENERAL ENQUIRIES

If you require assistance with a Commander Voice service, please contact Commander on:

132777

commander.com.au/support/office-phones/sip

customerservice@commander.com

Account Manager

If you are an existing customer, then please contact your Account Manager.

TECHNICAL SUPPORT

To resolve technical support requests as quickly as possible we will need a few details:

- Your first name, surname and contact details (email and/or phone),
- The Company name you are calling on behalf of
- Your Commander Service ID or Contract ID
- A detailed description of the incident or request including the impact, urgency, any troubleshooting already performed and if there have been any recent changes to your service.

For all Commander technical issues please contact the Commander Support Centre on:

132777

techsupport@commander.com

Please note: Email will only be attended to within business hours.

For high priority incidents, please call us.

LEGAL

Standard Terms and Conditions are available on Commander website:

commander.com.au/customer-terms

DISCLAIMER:

Commander makes no representations about the suitability of the information contained in this guide for any purpose. This guide and the content and information contained within could include inaccuracies or errors.

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Document version 1.1   11/11/2020