

# SIP Trunk Configuration Guide



## What is SIP Trunking?

Session Initiation Protocol (SIP) Trunking is a way to connect your business telephone system to the Public Switched Telephone Network (PSTN) so you can make and receive calls. Unlike traditional telephone lines or ISDN, which are provided over dedicated circuits, SIP Trunks are now being provided over the internet, making them more **accessible**.

Leveraging GO's investment in its fibre network, SIP trunks utilise this reliable and cutting edge network to provide voice services, unleashing new potential and improving your **business continuity**. SIP Trunks offer a more flexible medium in terms of capacity and functionality, making it a **cost effective** solution.



## What are the benefits of SIP?

### SIP is accessible

SIP trunks and the number and type of telephone numbers they handle can be easily increased. This proves to be useful, especially in times when businesses require more capacity due to an influx of calls. This is due to the fact that the SIP trunk is no longer bound by physical limitations in the form of an interface.

### SIP can be easily deployed

Because SIP trunks are handled by SIP servers on the GO secure cloud, these are not bound to be provisioned via a physical circuit which takes time to provision. SIP trunk channels and number ranges are bound to an account, which can be provisioned in a very short time.

### SIP is cost effective

SIP can often bring your telephony costs down over time, via more attractive rates and per second charging.

## Is my system GO SIP compliant?

### Is your current internet service adequate?

GO SIP utilizes 100Kb symmetrical bandwidth for each call, for example 10 channel SIP trunk will require at least 1Mb of bandwidth.

Apart from bandwidth, one must also take into consideration the JITTER and LATENCY of this connection. It is imperative that for optimal performance, one should have minimal JITTER and LATENCY. In this regard, it is important to establish the SIP trunk via a GO internet service, which ensures that the above criteria are met.

## Is my system GO SIP compliant?

### Edge Device/Firewall

Nowadays all internet connections are mainly connected to an Edge device commonly being a router or a firewall. The SIP trunk would need to traverse such devices to establish REGISTRATION. One needs to make sure that the below criteria are met before service provisioning. Device Health; CPU/Memory consumption needs to be low. In the event of high CPU usage, packets might be dropped, and this may impact the telephony experience.

Internet usage: nowadays most of our applications run over the cloud and hence internet usage has increased exponentially. Real time services, such as voice, need to be transmitted in real time without any delays, therefore it is of utmost importance that a proper QOS policy is in place on these devices.

These devices are able to handle SIP, however, specific configurations would be required, like SIP ALG and properly configured security rules.

## Is my system GO SIP compliant?

### Is your PABX SIP compatible?

Most of the recent PABX systems are SIP compatible. During the inquiry phase, it is recommended to provide the details requested in the form on the next page.

N.B—Should the current system not be SIP compliant, GO has a range of solutions such as;

- **SIP to ISDN convertor:** This will enable you to connect your legacy ISDN PBX to the GO SIP telephony infrastructure.
- **SIP to PSTN convertor:** This will enable you to connect your legacy PSTN PBX to the GO SIP telephony infrastructure.
- **SIP to SIP SBC:** Should the PBX not support DNS SRV, the SBC will handle the public SIP trunk whilst offering a private SIP trunk internally.

Additionally, GO offers complete PBX solutions via its partnership with leading PBX vendors, Alcatel & Cisco, together with the support of fully certified GO Specialists.

# Customer Compliance Form

Company Name	Contact	Contact Number	Comments

Equipment	Vendor	Model	Release
PBX			
Router			
Firewall			

Equipment	Service Identifier	Speed	Comments
Internet-1			
Internet-2			

# Native SIP Trunk – Data Sheet

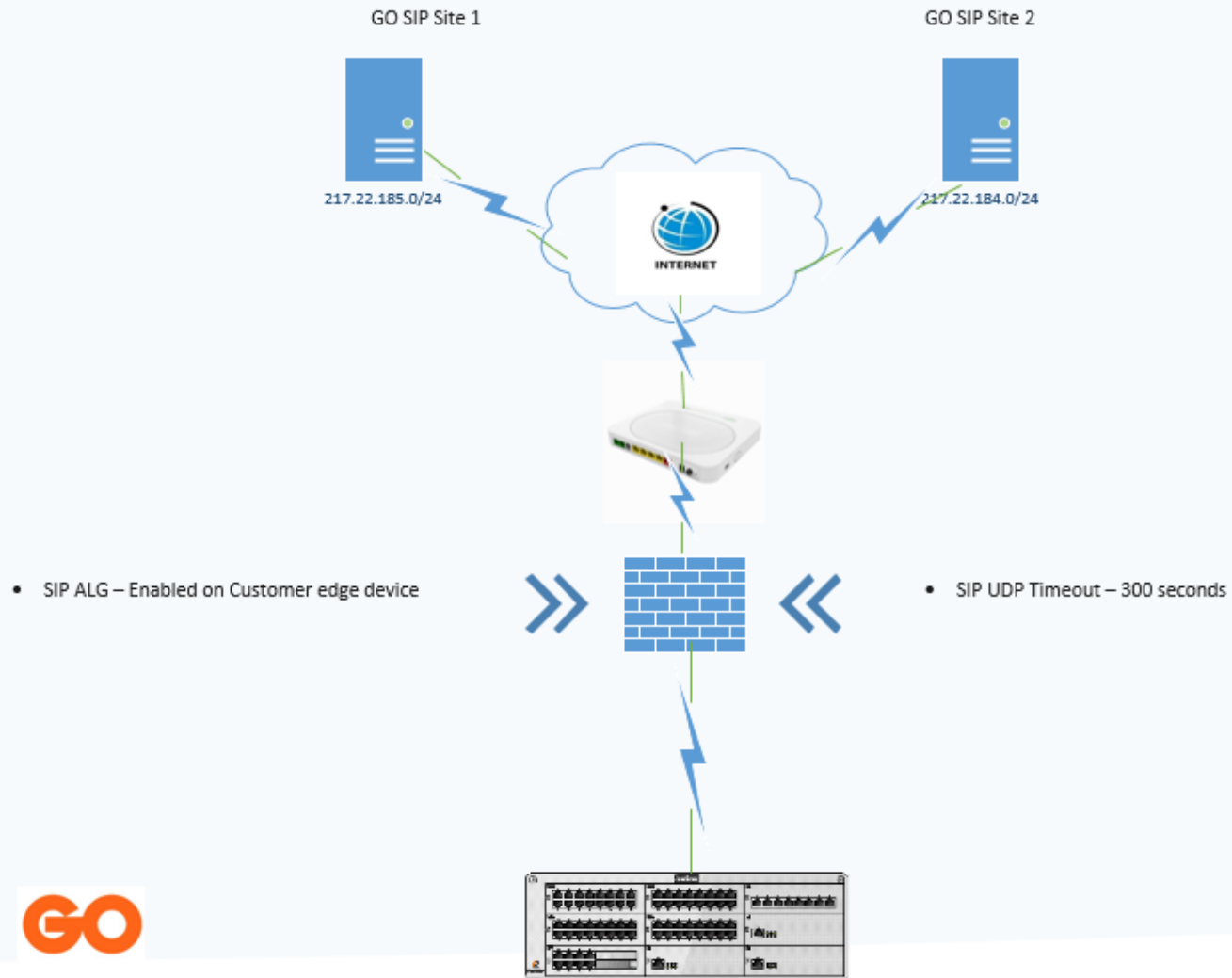
Protocols and encapsulation	SIP 2.0 ( <a href="#">RFC3261</a> and associated RFCs) for signalling  RTP for media encapsulation UDP only
GO SIP Trunk Gateways & Firewall Configuration	<b>trunk.gotel.com.mt</b> Firewall configuration: 217.22.185.0/24 [217.22.185.0-217.22.185.254] 217.22.184.0/24 [217.22.184.0-217.22.184.254]
DNS Query	Type = SRV
IP Ports	5060 UDP [SIP signalling] 1024-65535 UDP [RTP]
Outbound Call Authentication	Username/password authentication
DTMF (Tones)	RFC2833 [out-of-band]
Audio Codecs	G.711 a-Law
Codec Transcoding	Audio - Supported Video - Not Supported
Incoming number formats	Full E.164 format: + country code then number for international calls, e.g. +442035982801 Local Number Format for local calls e.g. 21212121
Outgoing number formats	Full E.164 format: + country code then number for international calls, e.g. +442035982801 Local Number Format for local calls e.g. 21212121
Re-Registration Timer	295 seconds
PPI Header Configuration	INVITE/REFER customer device shall include the registered number in the PPI header
Session Timer	900 Seconds

# SIP Trunk with Gateway – Data Sheet

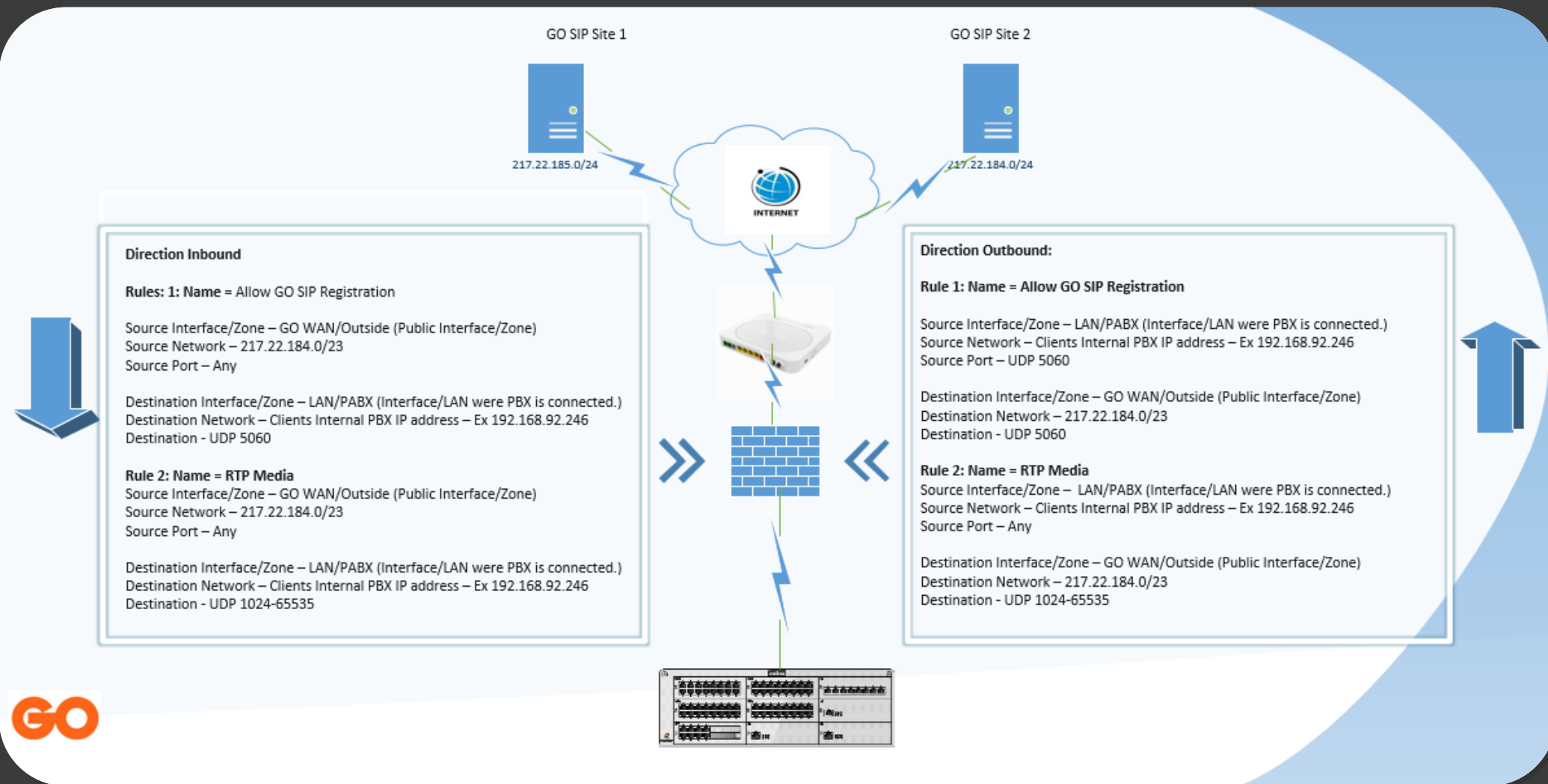
Protocols and encapsulation	SIP 2.0 ( <a href="#">RFC3261</a> and associated RFCs) for signalling  RTP for media encapsulation UDP only
GO SIP Trunk Gateways & Firewall Configuration	trunke.gotel.com.mt Firewall configuration: 217.22.185.0/24 [217.22.185.0-217.22.185.254] 217.22.184.0/24 [217.22.184.0-217.22.184.254]
DNS Query	Type = SRV
IP Ports	5061 TCP (SIP signalling) 1024-65535 UDP (RTP)
Additional IAD/SBC Management Port	<ul style="list-style-type: none"> <li>• ICMP [type 3 code 4 ICMP] <ul style="list-style-type: none"> <li>• 22 [SSH]</li> <li>• 443 [HTTPS]TCP</li> </ul> </li> <li>• To &amp; From 40.130.246.217 &amp; 40.130.246.218 (<a href="https://patton.io">https://patton.io</a>)</li> </ul>
NTP Servers	0.patton.pool.ntp.org 1.patton.pool.ntp.org 2.patton.pool.ntp.org 3.patton.pool.ntp.org

Outbound Call Authentication	Username/password authentication
DTMF (Tones)	RFC2833 (out-of-band)
Audio Codecs	G.711 a-Law
Codec Transcoding	Audio - Supported Video - Not Supported
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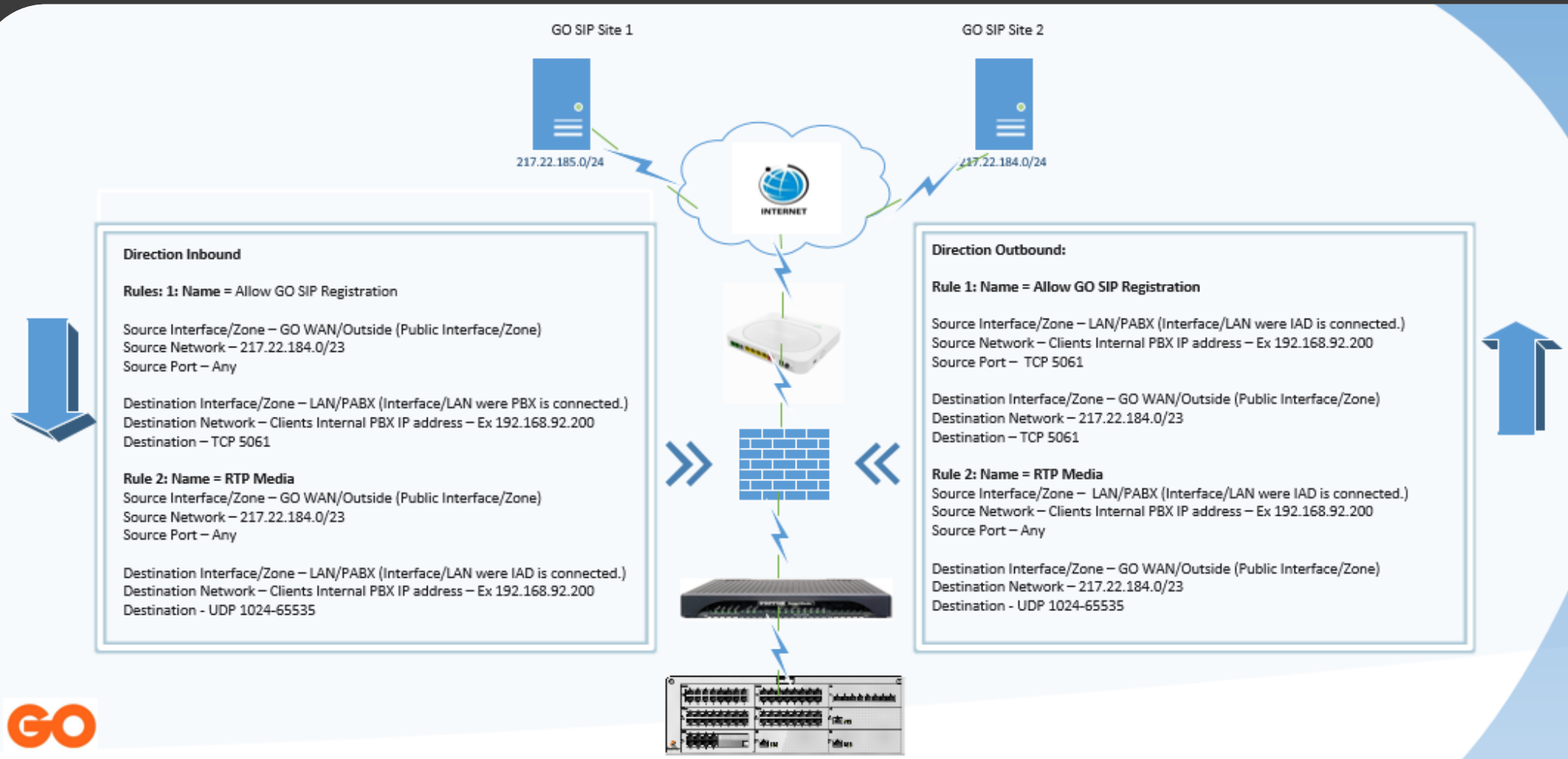
# Native SIP Trunk – Rules Diagram



# Native SIP Trunk – Rules Diagram



# SIP Trunk with Gateway – Diagram



# SIP Registration Flow

## Phase 1: Domain Resolutions

PBX Ability for SRV Query:

GO has geo-redundant sites, thus it is imperative that the PBX supports DNS SRV resolution to ensure customer 99.9% telephony service uptime.

Info

```
Standard query 0xcd4c SRV _sip._udp.trunk.gotel.com.mt  
Standard query response 0xcd4c SRV _sip._udp.trunk.gotel.com.mt SRV 45 50 5060 zt.trunk.gotel.com.mt  
Standard query 0xb3ce A zt.trunk.gotel.com.mt  
Standard query response 0xb3ce A zt.trunk.gotel.com.mt A 217.22.184.27
```

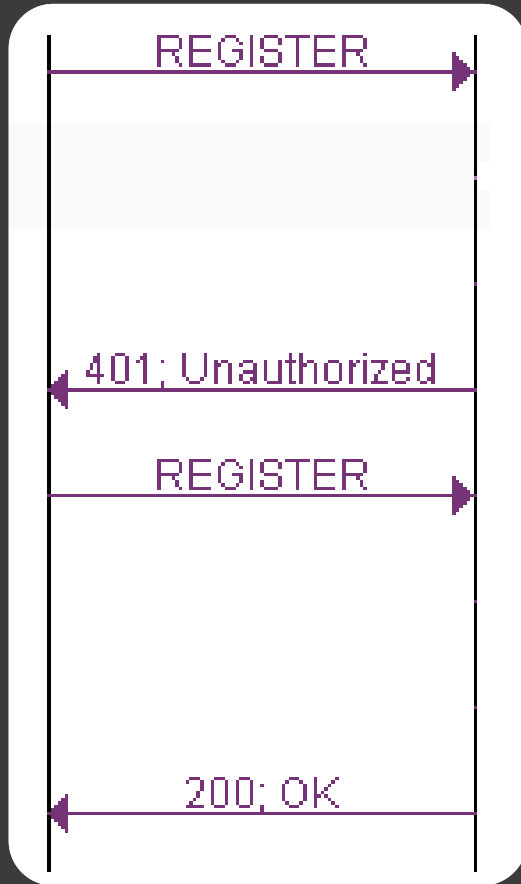
Highlighted is the priority which the PBX will parse all signalling to. Should a maintenance arise at ISP side, these priorities can and may be shifted around to ensure no disruptions to customer telephony service.

Info

```
Standard query 0xcd4c SRV _sip._udp.trunk.gotel.com.mt  
Standard query response 0xcd4c SRV _sip._udp.trunk.gotel.com.mt SRV 45 50 5060
```

# SIP Registration Flow

## Phase 2: Registration Register Flow:



# SIP Registration Flow

## Register Packet 1 – Initial Registration:

```
Headers
-----
CSeq: 2138474475 REGISTER
Expires: 300
User-Agent: OX0040/508.001 GW_040/508.001
To: <sip:35621669015@trunk.gotel.com.mt;user=phone>
From: <sip:35621669015@trunk.gotel.com.mt;user=phone>;tag=9da4cdebc6ece68120ca9e40dbd4c9b6
Contact: <sip:35621669015@Cust - Public IP :5060;transport=UDP;user=phone>;audio;class="business, personal";duplex="full
Call-ID: a910f9a988bebd20dc4c55e535d7e660@PBX - Internal IP
Via: SIP/2.0/UDP Cust - Public IP :5060;rport;branch=z9hG4bKc0690fd85cc2be9e2f0844b6b5fb821a
Max-Forwards: 70
Content-Length: 0
```

## Register Packet 2 – Challenge:

```
Headers
-----
Via: SIP/2.0/UDP Cust - Public IP 5060;received=Cust - Public IP;rport=5060;branch=z9hG4bKf845382266cb980506050124c2ea875e
To: <sip:35621669015@trunk.gotel.com.mt;user=phone>;tag=h7g4Esbg_1111985430-1626692840934
From: <sip:35621669015@trunk.gotel.com.mt;user=phone>;tag=d32a8fcb59967a9fdbb0536ef83be5e3
Call-ID: f38d79a4c51ca0cd1fda293010fd2c14@PBX - Internal IP
CSeq: 1128318778 REGISTER
Service-Route: <sip:217.22.184.27:5060;transport=udp;lr>
WWW-Authenticate: Digest realm="trunk.gotel.com.mt",nonce="BroadWorksXk",algorithm=MD5,qop="auth"
Content-Length: 0
```

# SIP Registration Flow

## Register Packet 3 – Reply to Challenge:

```

Headers
-Expires: 300
-User-Agent: OX0040/508.001 GW_040/508.001
-To: <sip:35621669015@trunk.gotel.com.mt:user=phone>
-From: <sip:35621669015@trunk.gotel.com.mt:user=phone>;tag=d32a8fcb59967a9fdbb0536ef83be5e3
-Contact: <sip:35621669015@Cust - Public IP :5060;transport=UDP:user=phone>;audio:class="business, personal";duplex="full, half";mobility="fixed";description="<OmniPCX Office>";methods="ACK, INVITE, CANCEL, BYE, I
-Call-ID: f38d79a4c51ca0cd1fda293010fd2c14@PBX - Internal IP
-CSeq: 1128318775 REGISTER
-Max-Forwards: 70
-Authorization: Digest username="35621669015", realm="trunk.gotel.com.mt", nonce="BroadWorksXkr", algorithm=MD5, qop=auth, cnonce="1626692840", nc=00000001, uri="sip:trunk.gotel.com.mt:transport=UDP",
-Via: SIP/2.0/UDP Cust - Public IP :5060;sport;branch=z9hG4bKf552b1017c03826472f4410e7e68dce5
-Content-Length: 0
  
```

Header	Accepted Format	Comments
To	35621669015@trunk.gotel.com.mt	GO Supplied – Customer SIP Username
From	35621669015@trunk.gotel.com.mt	GO Supplied – Customer SIP Username
Contact	35621669015@Customer-Public-IP	GO Supplied – Customer SIP Username
Expire Timer	295 Seconds	

# SIP Outgoing Call Flow

## Phase 3: Outgoing Calls

### Local Outgoing Calls:

```

Headers
- Allow: INVITE, ACK, CANCEL, BYE, OPTIONS, NOTIFY, PRACK, UPDATE
- Supported: 100rel, from-change, timer
- User-Agent: OX0040/508.001 GW_040/508.001
- Session-Expires: 900
- P-Preferred-Identity: sip:35621669015@trunk.gotel.com.mt;user=phone
- P-Early-Media: supported
- To: <sip:79019648@trunk.gotel.com.mt;user=phone>
- From: <sip:21669015@trunk.gotel.com.mt;user=phone>;tag=31a3ble43f58c19bfb3c7f6efe49a7be
- Contact: <sip:21669015@ .Cust - Public IP :5060;transport=UDP;user=phone>
- Content-Type: application/sdp
- Call-ID: 8f36a8af2b894653dbcb859651491ac6@ PBX - Internal IP
- CSeq: 380634015 INVITE
- Via: SIP/2.0/UDP .Cust - Public IP :5060;rport;branch=z9hG4bK300675143a10359c56fb4310c735da0f
- Max-Forwards: 70
- Content-Length: 323
    
```

### Headers - Local Outgoing Call:

Header	Format (E164 Standard)	Accepted Format	Comments
To	National	79019648@trunk.gotel.com.mt	
From	National	21669015@trunk.gotel.com.mt	
PPI		35621669015@trunk.gotel.com.mt	GO Supplied – Customer SIP Username
Contact		21669015@Customer-Public-IP	

# SIP Outgoing Call Flow

International Outgoing Calls:

```

Headers
--Allow: INVITE, ACK, CANCEL, BYE, OPTIONS, NOTIFY, PRACK, UPDATE
--Supported: 100rel,from-change,timer
--User-Agent: OX0040/508.001 GW_040/508.001
--Session-Expires: 900
--P-Preferred-Identity: sip:35621669015@trunk.gotel.com.mt;user=phone
--P-Early-Media: supported
--To: <sip:0044          ?trunk.gotel.com.mt;user=phone>
--From: <sip:21669015@trunk.gotel.com.mt;user=phone>;tag=3abc8b5f57aed8b3f4556446d2ddef56
--Contact: <sip:21669015@7 Cust-Public IP |:5060;transport=UDP;user=phone>
--Content-Type: application/sdp
--Call-ID: d25ce3d14c6e285c7706aee0c463b60e@ PBX-Internal IP
--CSeq: 791776742 INVITE
--Via: SIP/2.0/UDP Cust-Public IP |:5060;rport;branch=z9hG4bK13aa1392c8ac639893965da5f0bcbf4b
--Max-Forwards: 70
--Content-Length: 323
  
```

Header	Format [E164 Standard]	Accepted Format	Comments
To	International	0044#####@trunk.gotel.com.mt +44#####@trunk.gotel.com.mt	
From	National	21669015@trunk.gotel.com.mt	
PPI		35621669015@trunk.gotel.com.mt	Customer's Username of SIP account
Contact		21669015@Customer-Public-IP	

# SIP Incoming Call Flow

## Phase 4: Incoming Calls

### Local Incoming Calls:

```

Headers
Via:SIP/2.0/UDP 10.160.0.10;branch=z9hG4bKBroadWorks.-vrbhas-10.160.0.74V5060-0-271445029-1623834766-1630482978889-
From:<sip:79019648@trunk.gotel.com.mt;user=phone>;tag=1623834766-1630482978889-
To:"ess_admins_2 ess_admins_2"<sip:21669015@trunk.gotel.com.mt;user=phone;EriBindingId=30578025146488;eribind-generated-at=10.160.0.74>
Call-ID:BW095618889010921754774884@10.160.0.10
CSeq:271445029 INVITE
Contact:<sip:ascll.gotel.com.mt>
P-Asserted-Identity:<sip:79019648@trunk.gotel.com.mt;user=phone>
Privacy:none
P-Early-Media:supported
Supported:100rel,timer
Allow:ACK,BYE,CANCEL,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Recv-Info:x-broadworks-client-session-info
X-BroadWorks-Correlation-Info:838903e2-6f27-47c5-b15c-3a7a95f8e751
Accept:application/btbc-session-info,application/dtmf-relay,application/media_control+xml,application/sdp,application/vnd.etsi.aoc+xml,multipart/mixed
Min-SE:60
Session-Expires:900;refresher=uac
Max-Forwards:10
Content-Type:application/sdp
Content-Length:254
    
```

### Headers - Local Incoming Call:

Header	Format [E164 Standard]	Accepted Format	Comments
To	National	21669015@trunk.gotel.com.mt	
From	National	79019648@trunk.gotel.com.mt	

# SIP Incoming Call Flow

## International Incoming Call:

```

SIP
  Request Line
  - INVITE sip:35621669015@ Cust Public IP :5060;transport=udp;user=phone SIP/2.0
  Headers
  - Max-Forwards: 9
  - Via: SIP/2.0/UDP 217.22.184.27:5060;branch=z9hG4bKg3Zqkv7iw3m7u3junvz28m60lh9gviykj
  - To: "ess_admins_2 ess_admins_2" <sip:21669015@trunk.gotel.com.mt;user=phone;eribindingid=30554277693075;eribind-generated-at=10.160.64.74>
  - From: <sip:+44 @trunk.gotel.com.mt;user=phone>;tag=h7g4Esbg_225460398-1629790693471-
  - Call-ID: BW093813471240821884948513@
  - CSeq: 999044144 INVITE
  - Contact: <sip:sgc_c@217.22.184.27;transport=udp>
  - Record-Route: <sip:217.22.184.27;transport=udp;lr>
  - Min-Se: 900
  - P-Asserted-Identity: <sip:+44 @trunk.gotel.com.mt;user=phone>
  - P-Early-Media: supported
  - Privacy: none
  - Session-Expires: 900;refresher=uac
  - Supported: 100rel
  - Supported: timer
  - Content-Type: application/sdp
  - Content-Length: 259
  - X-Broadworks-Correlation-Info: flaad0c-7ccb-44b2-bd43-374dfd90053d
  - Recv-Info: x-broadworks-client-session-info
  - Allow: ACK, BYE, CANCEL, INVITE, OPTIONS, PRACK, REFER, NOTIFY, UPDATE
  - Accept: application/btbc-session-info, application/dtmf-relay, application/media_control+xml, application/sdp, application/vnd.etsi.aoc+xml, multipart/mixed
  
```

Header	Format [E164 Standard]	Accepted Format	Comments
To	National	21669015@trunk.gotel.com.mt	
		0044#####@trunk.gotel.com.mt	
From	International	+44#####@trunk.gotel.com.mt	

# Solutions for Legacy Systems



Model 4131—2 BRI Ports : Up to 4 Channels



Model 4131—4 BRI Ports : Up to 8 Channels



Model 4171—1 PRI Ports : Up to 30 Channels



Model 5301: — Up to 60 Channels



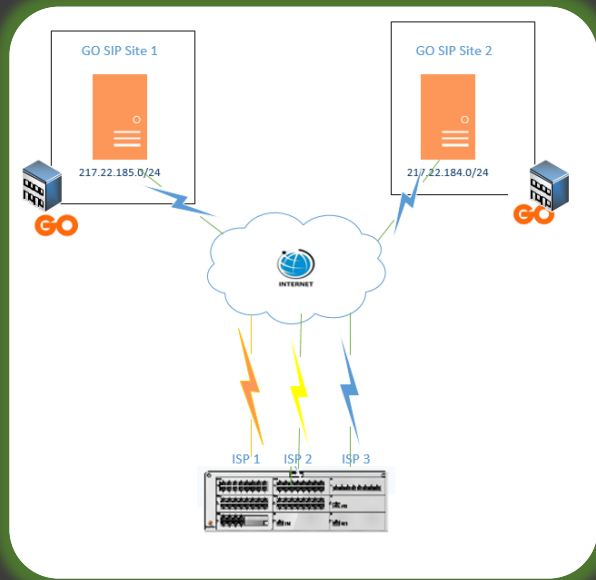
Model 5501: — Up to 200 Channels



Model 4141— 8 PSTN Ports: Up to 8 Channels

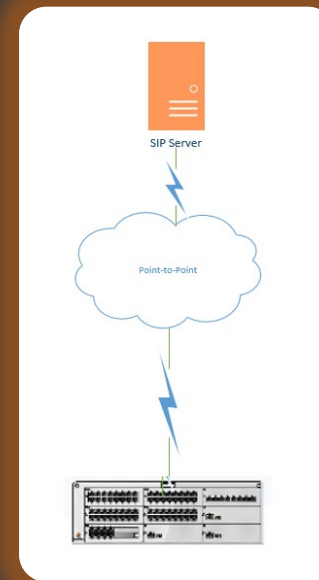
## Deployment Flexibility – O.T.T Service vs Legacy Point-to-Point

The GO SIP Service can be deployed from any location around the globe, eliminating the constraints of local connectivity and the need for additional hardware. This stands in contrast to legacy Point-to-Point Services, which necessitate the installation and maintenance of dedicated circuits. This not only enhances service flexibility but also reduces the overall cost of ownership, as no additional expenses are incurred.



### O.T.T Service

Service is not bound to any physical circuit. Furthermore, should an internet failure occur, the trunk can easily re-establish via any other available internet connection\*.



### Static IP Configurations

As per name, these are bound to physical connections, and consequently, to specific circuits, hence these cannot leverage the client's internet resiliency.

Malicious Users can also easily spoof the IP address and simulate the client's system.

N.B \* Non-GO Public IP Addresses would require to be whitelisted in order to be able to connect to the GO SIP Servers.

# Certified SIP Trunk

GO SIP Trunk is constantly undergoing certification with various vendors such as 3CX, Alcatel, Alcatel Rainbow, Cisco. This will ensure the compatibility of these platforms with the GO SIP Service.

The certification implies that the service goes under exhaustive testing for all its features such as;

- Call Forwarding
- Music on Hold
- Call Line Identification etc..

GO SIP Trunk is the only locally certified SIP service with leading industry vendors, providing users with pre-defined templates to facilitate service installation.



https://www.3cx.com/partners/sip-trunks/malta/

### 3CX SIP Trunk Providers – Malta

These SIP Trunk providers in Malta have been tested and certified by 3CX. We strongly recommend using a preferred or supported SIP Trunk service provider. The 3CX team makes sure that all templates from listed SIP Trunk Providers are updated and tested with every release ensuring no issues arise for our users.

Registration-based trunks with no IP fixation remain the preferred option in a hosted environment. This is because in certain cases the Public IP of your Hosted by 3CX can change unexpectedly. Although this is not a regular occurrence, if this happens, you will temporarily lose the ability to make and receive calls via your IP-based trunks.

Read our guide on [how to configure a SIP Trunk / VoIP Provider](#). This covers the generic steps required to configure a trunk with 3CX. Some providers may need special configuration requirements. Before choosing a SIP Trunk provider in your region, carefully check our listing for these special requirements and be sure to follow any extra configuration guides added.

Supported (Dedicated Servers Only)

Supported Providers (Dedicated Servers only) are tested against 3CX Hosted and on-premise installations. Not compatible with 3CX FREE/SMB. Configuration may require a few extra steps.

Capabilities	Important Configuration Notes
SMS Support	No
Anonymous Calling	Yes
Clip No Screening Support	No
Fax Support	Yes - Audio
SIP TLS Support	No
IPV6 Ready	No
Online Sign-Up & Portal	No

Enter the main trunk number in the national format (e.g. 7955439).  
You can only present numbers associated with your account as Outbound Caller ID.

TC1284en-Ed127\_OXO\_Connect\_Public\_SIP\_Trunking\_Interoperability\_and\_Technical\_SupportProcedure PDF - 3.3 MB

			● R4.0	● R5.2	● R6.0
Malta	GO (Router with SIP-ALG TOPO B)	Limited Availability			
	GO (Router with SIP-ALG TOPO B)	General Availability			
	GO (Remote Hosted NAT-TOPO C)	Limited Availability			
	GO (Remote Hosted NAT-TOPO C)	General Availability	TC2815 (R4.0)	● R4.0	● R5.2
Mauritius	EMTEL Feel Free	Limited Availability	● R4.0	● R5.2	● R6.0
		General Availability			

https://www.openrainbow.com/cloud-telephony/

Solutions Plans Security Help center

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# Service flexibility – Redundancy Options

In resilient setup scenarios, one would normally have multiple SIP trunks to achieve the desired resiliency. As previously remarked, GO is able to provide such resiliency with the below enhanced call handling mechanisms.

- **Overflow**
  - Once a primary SIP Trunk Channel Capacity is exhausted, the calls are overflowed to the other SIP Trunk which forms part of the Enterprise Group. [Diagram 1]
- **Load Sharing**
  - Priority and Weight metrics play a vital role in this mechanism since these will dictate how the calls are served to each SIP Trunk in the Enterprise Group. [Diagram 2]

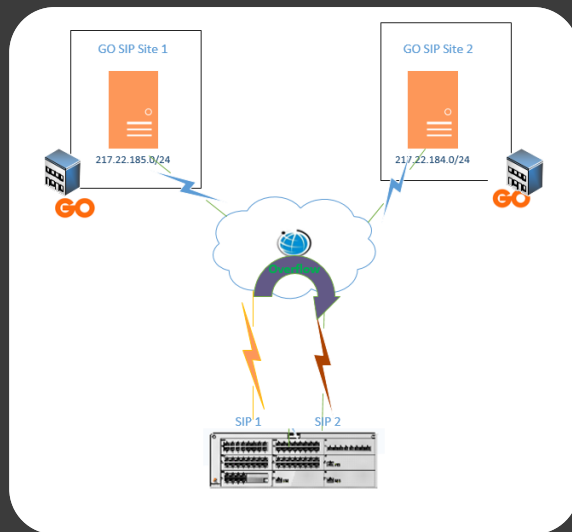


Diagram 1

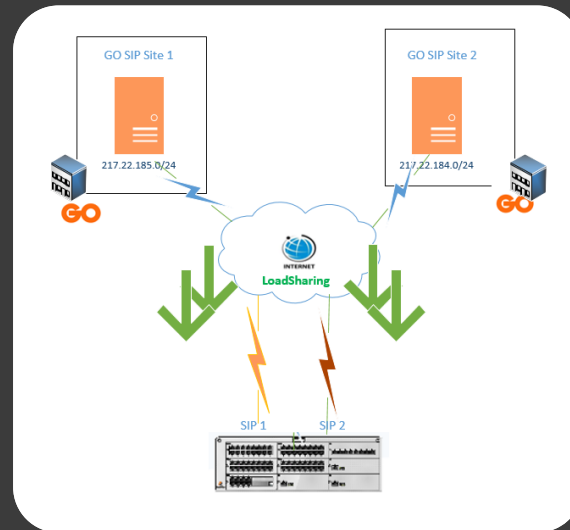


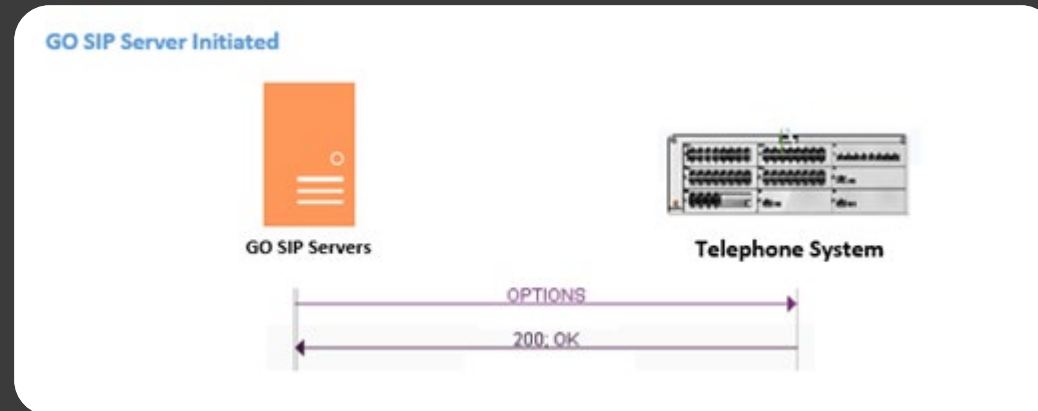
Diagram 2

N.B. GO offers full service high availability within its own network components, whilst customers must ensure that they have full resiliency within their infrastructure, and the SIP Trunk Firewall configurations are in accordance with GO's specifications in this document.

## Redundancy – Keep-Alive Messages

A core feature of our service is to ensure uptime /security and high availability. GO sends keepalive options message to ensure that the client's system is still up and running. In case of high availability systems, should the primary system not reply to the options message, calls will be routed to the secondary system.

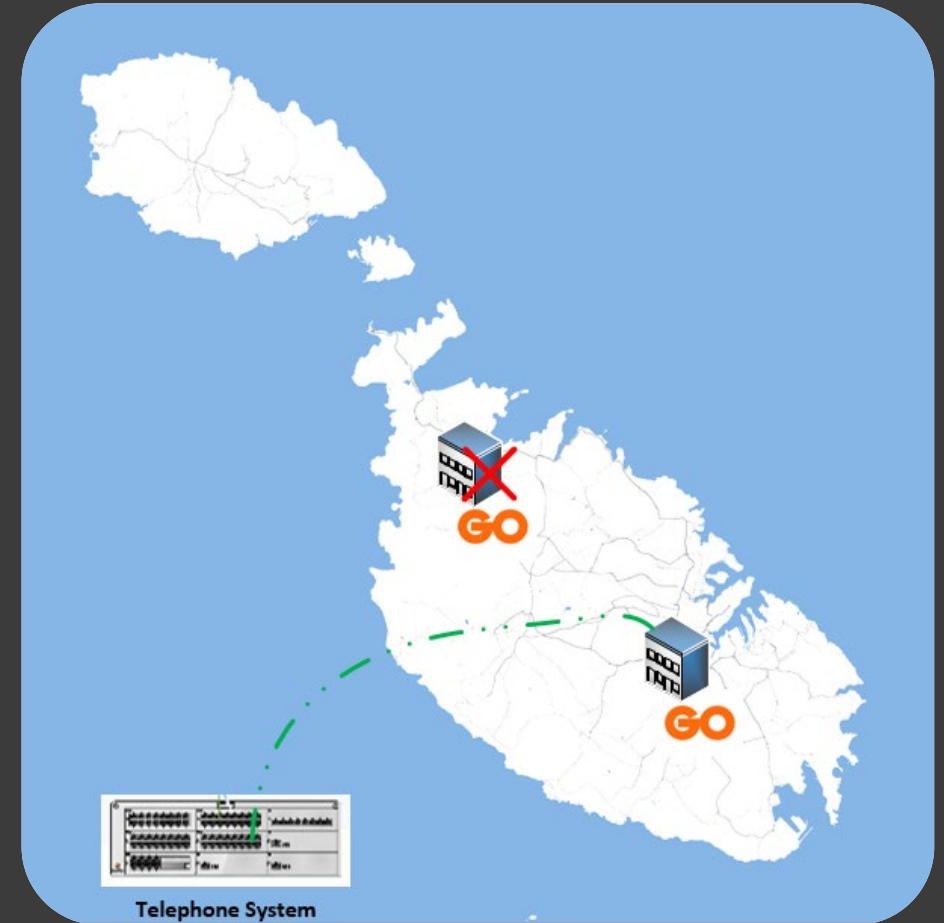
A keepalive [KA] is a communication between devices to query the state of the authenticated session.



## Redundancy – Geo-Redundant Setup

As part of GO's high availability strategy, GO has strategically deployed its redundancy system in two very distant geo-locations. This is to achieve full resiliency in case of a hazard or damage to any of its infrastructure in a particular area.

Should an issue occur on an element in one of the sites, the GO SIP Server will seamlessly divert traffic to the geo-redundant site.



# Authentication

GO considers security as being a core part of our service, therefore all SIP trunks are authenticated via the use of User Credentials and TLS options. These credentials authenticate the telephone system with the provider's SIP servers, allowing it to place and receive calls. This security measure aligns well with number validation (covered in the next slide).

Static legacy SIP trunks lack authentication as part of their core feature list. Since this service utilises a private circuit, this might cause security risks to the telephony setup. For instance, another customer with the same static legacy SIP service could incorrectly or maliciously enter someone else's telephone number, leading to spoofed calls. Such usage will be reflected in the bill and may also result in the impersonation of your business, impacting its reputation. Further information about call spoofing will be provided in the upcoming slide.



# Call identification / Validation

The calls undergo thorough inspection to ensure that no misuse of the SIP account occurs in any possible manner, such as false number presentation (spoofing).

Number presentation spoofing is a method used by scammers to impersonate your business telephone connection, deceiving end users, such as your esteemed clientele.

To prevent such occurrences, GO SIP Servers only permit registered SIP accounts to present telephone numbers that are bound to their service. Therefore, calls presenting a caller ID not part of the same account will be rejected.

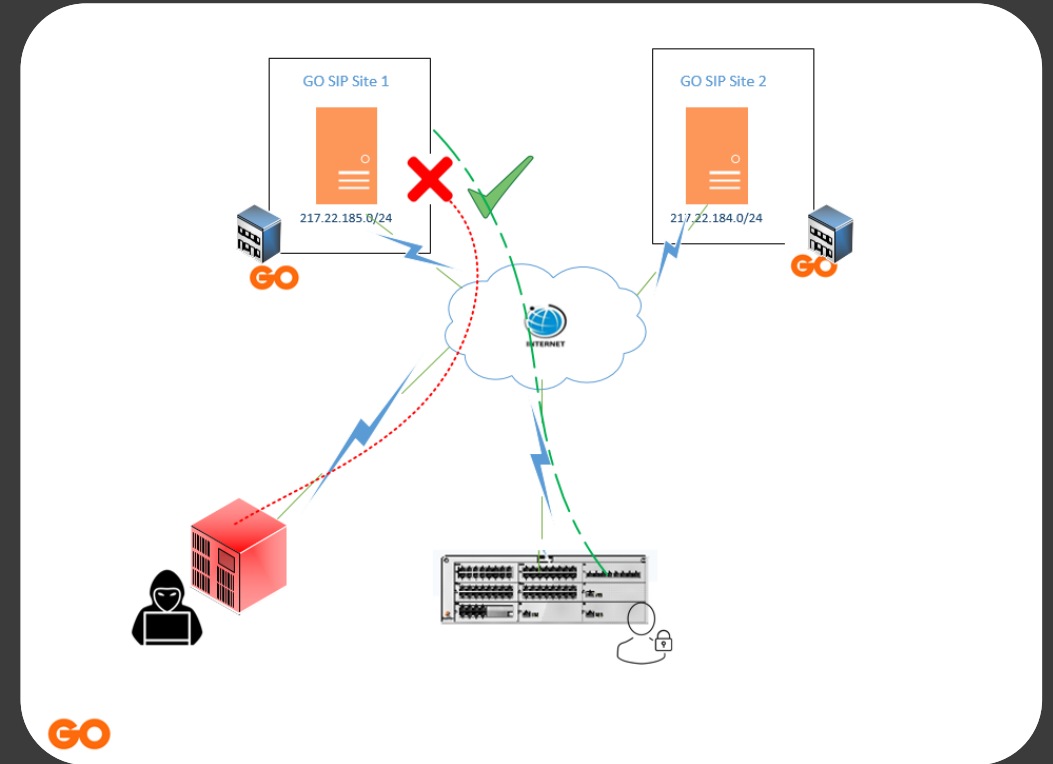


# Account Lockouts

Another imperative security feature to enhance the authentication mechanism is to activate an account lock should 3 consecutive registration failures occur.

This prevents malicious users from attempting to gain unauthorized access to the SIP account.

The account will be forbidden from further attempts until a GO Engineer unlocks the account.



# Penetration Testing

All GO's systems, including the SIP Servers, periodically undergo penetration testing via certified cybersecurity professionals as part of the ISO certification.

This goes hand in hand with continuous updates and patching of the system to mitigate any vulnerabilities which would have been identified in the mentioned test or specified by the vendor.



# Fraud Prevention Measures

As an additional layer of security, apart from the features outlined earlier, GO employs a dedicated team of anti-fraud experts who use state-of-the-art tools and platforms for real-time monitoring of all voice traffic. Our Anti-Fraud team can detect and block fraudulent activity until the client remedies the situation on their telephony system.



## Significant Savings

Prevented €2.3 million in fraud



## Targeted Blocking

Successfully blocked nearly 130,000 ranges



## Proactive Measures

Blocked 339,000 fraudulent calls and 55,000 SMS



## Effective Fraud Response

Of 347 fraud cases, only 24 led to actual usage and billing



## Rapid Fraud Detection

Prevented a bill from reaching €94k, capping it at €136 within an hour

# Thank you

Should you encounter any issues, or require further clarification, the GO Enterprise Solutions team is here to assist.

Reach out to us at: [customer@go.com.mt](mailto:customer@go.com.mt)