Stage 1 – SIP Trunking Acceptance Testing

The following tests have been defined to be undertaken in commissioning the Customer Premises Equipment (CPE) that is connected to the Gamma SIP Trunking service.

It will be the responsibility of the Channel Partner to carry out these tests and to forward the results to Gamma for future reference as appropriate. Gamma accepts no responsibility for any subsequent faults if these tests haven't been carried out and signed off.

A summary of each of these tests is below, each test should be completed a number of times to ensure complete confidence in the service.

Test 1 – 9: Inbound and Outbound Calls

Test 1 – Gamma SIP Trunks Call – SIP Clear

Step	Description
Name:	Successful call – User A calling PSTN user. User A clears.
Preconditions:	Valid PSTN/Mobile in place, that can be called and verified

Test Steps

Step	Description
1	Place a call to the PSTN number using SIP service
2	Answer the call
3	Wait approx 10 secs
4	Clear the line on the caller side

Test 2 – SIP Call – PSTN Clear

Step	Description
Name:	Successful call – User A calling PSTN user. PSTN user clears
Preconditions:	Valid PSTN/Mobile in place, that can be called and verified

Test Steps

Step	Description
1	Place a call to the PSTN number using SIP service
2	Answer the call
3	Wait approx 10 secs
4	PSTN clears call down
5	Wait 120 Seconds
6	Caller receives clear down

User Initiated SUSPEND and RESUME on the PSTN means a called PSTN user doesn't clear the call down until an exchange timer expires. This applies to calls originated from any network therefore the behaviour would be the same on a call between a mobile and PSTN.

Test 3 – PSTN Call – PSTN Clear

Step	Description
Name:	Successful call – PSTN user calling User A. PSTN user clears
Preconditions:	Valid PSTN/Mobile in place, that can place a call

Test Steps

Step	Description
1	Place a call from the PSTN line calling the SIP assigned Gamma geo number
2	User A answers the call
3	Wait approx 10 secs
4	Clear the line on the caller (PSTN) side

Test 4 – SIP Call – SIP User Release without answer

Step	Description
Name:	Calling party release before answer – calling party is User A1.
Preconditions:	Valid PSTN/Mobile in place, that can be called

Test Steps

Step	Description
1	Place a call on the SIP line calling the PSTN number
2	Release the call without answer

Test 5 – SIP Call – PSTN User Release without answer

	Description
Name:	Calling party release before answer – calling party is PSTN user
Preconditions:	Valid PSTN/Mobile in place, that can place a call

Test Steps

Step	Description	Expected Result
1	Place a call on the PSTN line calling the SIP assigned Gamma geo number	Ring-back (Ring tone)
2	Release the call without answer	Ringing stops

Test 6 – Invalid Number Call Test

	Description
Name:	Invalid number – User A calling from Gamma SIP Trunks
Preconditions:	

Test Steps

Step	Description	Expected Result
1	Call the invalid number	Numberunobtainable tone or not recognizedprompt

Test 7 – Incoming Call – PSTN Busy

	Description
Name:	Incoming call – user busy. PSTN busy
Preconditions:	Valid PSTN/Mobile in place, that has no answering machine or call waiting feature installed and a third phone number (User C) which can answer a call

Test Steps

Step	Description	Expected Result
1	Call the third phone number (User C) from thePSTN line, wait until it gets answered	

2	Place a call on the SIP line calling the PSTN number	Busy tone

Test 8 – Incoming Call – SIP User Busy

	Description
Name:	Incoming call – user busy. User A busy
Preconditions:	Valid PSTN/Mobile in place, and a third phone number which can answer a call

Test Steps

Step	Description	Expected Result
1	Call the third phone number from the SIP line, wait until it gets answered	
2	Place a call on the PSTN line calling the SIP number	Busy tone

Test 9 – Address Incomplete

Step	Description
Name:	Address incomplete – User A calling
Preconditions:	

Test Steps

Step	Description	Expected Result
1	Call the following incomplete number from the SIP line: 654321	Fast busy orService announcement

Test 10 – 13: CLI Presentation

In order to successfully present an ‘A Number’ we expect this to be sent to Gamma in either of the following formats in the FROM header of the SIP INVITE:

10 Digits without leading zero

From: <sip:1625827748@83.245.6.117>;tag=3541226335-339769

E164 Format (+44)

From: <sip:+441625827748@83.245.6.117>;tag=3541226335-339769

The A-number is checked against a database on the Gamma network of geographic numbers that are allocated to the SIP Trunking Endpoint. If the number presented does not meet the above criteria, the A-Number CLI presented will be a default CLI, which is the first number in the Gamma allocated Geographic DDI range.

Test 10 – CLI Presentation Test – PSTN > SIP

	Description
Name:	CLIP – PSTN calling end. Confirm A-end phone rings and number is displayed
Preconditions:	Valid PSTN/Mobile in place which has present CLIsetting enabled, and CLI capable phone set connected to SIP line

Test Steps

Step	Description	Expected Result
1	Place a call on the PSTN line calling the SIP assigned Gamma geo number	Ring-tone and correct CLI shown with leading zero

Test 11 – CLIR Test PSTN > SIP

	Description
Name:	CLIR – PSTN calling end. Confirm A-end phone rings and number is withheld
Preconditions:	Valid PSTN/Mobile in place which has CLI setting set to withheld, and CLI capable phone set connected to SIP line

Test Steps

Step	Description	Expected Result
1	Place a call on the PSTN line calling the SIP assigned Gamma geo number	Ring-tone and no CLI or withheld shown as CLI

Test 12 – CLI Presentation Test – SIP > PSTN

	Description
Name:	CLIP – A-user calling end. Confirm that PSTN phone rings and caller number is displayed
Preconditions:	Valid PSTN/Mobile in place which is capable showing CLI , and SIP CPE set to enable CLI sending

Test Steps

Step	Description	Expected Result
1	Place a call on the SIP line calling the PSTN number	Ring-tone and correct CLI shown with leading zero

Test 13 – CLIR Test SIP > PSTN

	Description
Name:	CLIR – A-user calling end. Confirm PSTN phone rings and caller number is withheld.
Preconditions:	Valid PSTN/Mobile in place which is capable showing CLI , and SIP CPE set to privacy full (hide CLI)

Test Steps

Step	Description	Expected Result
1	Place a call on the SIP line calling the PSTN number	Ring-tone and no CLI or withheld shown as CLI

Test 14 – 15: Fax

The Gamma SIP Trunking service will support Fax and Modem transmission subject to the following constraints

FAX and Modem transport in band using G.711 a-law codec is supported.

Renegotiation to T.38 is supported (subject to interoperability testing).

The use of G729 for in-band faxes is not supported, as its compressed nature may cause tones and messages to be lost.

Test 14 – Fax Call from PSTN (If configured for FAX Support)

	Description

Name:	Fax call – from PSTN (G.711)
Description:	
Preconditions:	Fax equipment connected to PSTN line, second fax machine connected to SIP line, SIP Service and CPE set to g.711 fax pass through

Test Steps

Step	Description	Expected Result
1	Send a 5 page fax call on the PSTN line calling the SIP assigned Gamma geo number	Ring-tone, training and answer
2	Check fax transmission	Fax machines should train and send all pages with no errors.

Test 15 – Fax Call to PSTN (If configured for FAX Support)

	Description
Name:	Fax call – from fax user in ‘A’ domain (G.711)
Description:	
Preconditions:	Fax equipment connected to PSTN line, second fax machine connected to SIP line, SIP Service and CPE can negotiate the codec to g.711

Test Steps

Step	Description	Expected Result
1	Send a 5 page fax call on the PSTN line calling the SIP assigned Gamma geo number	Ring-tone, training and answer
2	Check fax transmission	Fax machines should train and send all pages with no errors.

Test 16 – 17: Call Barring

Call Barring is applied and managed via the Gamma Portal.

Test 16 – Call Barring – If requested to be set up on the SIP Account

	Description
Name:	Call barring – confirm that prohibited numbers are blocked.
Description:	
Preconditions:	Provisioning of call barring SIP service should be requested and completed

Test Steps

Step	Description	Expected Result
1	Call a number which is barred from the SIP line (Premium, Mobile, International)	Fast busy signal or barred prompt

Test 17 – Call international number (if no international bar is in place)

	Description
Name:	Call Barring – confirm international numbers can be dialled
Preconditions:	International call barring = No

Test Steps

Step	Description	Expected Result
1	Call the following number from Gamma SIP Trunks (Paris, France): 0033170758109	Ring-tone, will be unanswered

Test 18 – 24: Shortcode Dialling

As part of the provisioning process the endpoint is automatically configured to a range of short codes for Emergency Services and Directory Enquiries.

Test 18 – Dial 999 Shortcode

	Description
Name:	Dial 999 Shortcode
Description:	Confirm that a valid End User 999 address is provisioned against the calling party number (All 999 calls without EU address details are reported to OFCOM)

Test Steps

Step	Description	Expected Result
1	Place a call to 999 via SIP line	Ring-tone
2	B party answers call	Speech
3	Confirm to B party that test call is being made and terminate call	Clear down (BYE message on SIP)

Test 19 – Dial 100 Shortcode

	Description
Name:	Dial 100 Shortcode

Description:	
Preconditions:	Call 100 from the SIP line

Test Steps

Step	Description	Expected Result
1	Place a call to 100 via SIP line	Ring-tone
2	B party answers call	Speech
3	Confirm to B party that test call is being made and terminate call	Clear down (BYE message on SIP)

Test 20 – Dial 101 Shortcode

	Description
Name:	Dial 101 Shortcode
Description:	
Preconditions:	Call 101 from the SIP line

Test Steps

Step	Description	Expected Result
1	Place a call to 101 via SIP line	Ring-tone
2	B party answers call	Speech
3	Confirm to B party that test call is being made and terminate call	Clear down (BYE message on SIP)

Test 21 – Dial 111 Shortcode

	Description
Name:	Dial 111 Shortcode
Description:	
Preconditions:	Call 111 from the SIP line

Test Steps

Step	Description	Expected Result
1	Place a call to 111 via SIP line	Ring-tone
2	B party answers call	Speech
3	Confirm to B party that test call is being made and terminate call	Clear down (BYE message on SIP)

Test 22 – Dial 112 Shortcode

	Description
Name:	Dial 112 Shortcode
Description:	
Preconditions:	Call 112 from the SIP line

Test Steps

Step	Description	Expected Result
1	Place a call to 112 via SIP line	Ring-tone
2	B party answers call	Speech

3	Confirm to B party that test call is being made and terminate call	Clear down (BYE message on SIP)

Test 23 – Dial 195 Shortcode

	Description
Name:	Dial 195 Shortcode
Description:	
Preconditions:	Call 195 from the SIP line

Test Steps

Step	Description	Expected Result
1	Place a call to 195 via SIP line	Ring-tone
2	B party answers call	Speech
3	Confirm to B party that test call is being made and terminate call	Clear down (BYE message on SIP)

Test 24 – Dial 123 Shortcode

	Description
Name:	Dial 123 Shortcode
Description:	
Preconditions:	Call 123 from the SIP line

Test Steps

Step	Description	Expected Result
1	Place a call to 123 via SIP line	Ring-tone
2	B party answers call	Speech
3	Confirm to B party that test call is being made and terminate call	Clear down (BYE message on SIP)

Test 25: DTMF

The following methods will be supported to transport DTMF tones:

The Gamma core network will support the generation of ‘In-band’ or ‘RFC2833’ DTMF transport based on end to end negotiation.

RFC2833 is the preferred method for the transport of DTMF tones. Support of RFC 2833 is dependent on successful codec negotiation and requires the payload type 101 to be assigned. RFC2833 will be used with both G.711.and G.729 codecs.

In band over G.711 codec only. If a G729 codec is being used then DTMF tones should not be sent in-band, Gamma will not guarantee the delivery of in-band DTMF over a G729 codec.

.

Test 25 – DTMF

	Description
Name:	Test DTMF
Description:	
Preconditions:	Call 08081788000

Test Steps

Step	Description	Expected Result
1	Place a call to 08081788000 via SIP line	Ring-tone
2	Connect to Gamma IVR	Speech
3	Press 1 to navigate menu	Tone recognised, forwarded to next stage of IVR