JT SIP TRUNK

Technical Specification Exchange Line Services

Contents

1 SIP TRUNK SERVICES

JT SIP Trunk service is delivered over the following interface:

SIP Trunk Delivery Line (Jersey) – A fibre delivered, Ethernet presented interface providing secure dedicated bandwidth across the JT MPLS core. Each voice session will be allocated 128Kbit/s bidirectional bandwidth.

1.1 CUSTOMER PREMISE EQUIPMENT (CPE)

The JT SIP Trunk service is designed to operate with and support Customer's Premises Equipment (CPE) complying with the SIP Connect Technical Recommendation Version 1.1.

JT does not mandate the use of a SIP security proxy (such as a Session Boarder Controller) on the customers premise.

2 SIP TRUNK DELIVERY LINE

The services provided are as follows:

2.1 USER-NETWORK INTERFACE LAYER 1

This is presented as RJ-45 and supports Category 5 UTP cabling in accordance with TIA/EIA-568

The RJ-45 interface will be the JT demarcation point.

2.2 USER-NETWORK INTERFACE LAYER 2

This is provided as 10/100/1000Base-T in accordance with IEEE 802.3ab.

2.3 USER-NETWORK INTERFACE LAYER 3

This is provided as an IPV4 IP address in accordance with RFC 791.

JT does not support the use of DNS on the JT SIP Trunk Service.

2.4 USER-NETWORK INTERFACE SIP LAYER

This is provided in accordance with SIP Connect Technical Recommendation V1.1

The following options are provided:

- Use of SIP Options* messages for availability heartbeat.
- Use of Static or Registration Mode
- Use of UDP / TCP
- Use of E.164 or Subscriber numbering

The following options are not supported:

- Use of DNS

The Service will support a maximum of 210 simultaneous sessions per physical SIP Delivery Line.

*Note: - When using the SIP options message, the to/from header must be in format recognised in RFC 3261section 8.1.1. The to and from headers must in a recognised SIP URI format. The from header must be the actual user in phone number@IP address or phone number@ims.jt.com. Failure to comply will result in blacklisting of the sender of the request by the JT network as the user cannot be recognised.

2.5 SECURITY SERVICES

In addition to the inherent security provided over the JT core, some applications may demand additional authentication and encryption mechanisms. The JT SIP Trunk service supports the following options:

Security Option	Media / Signalling / Both	Standard
SIP Authentication	N/A	SIP Authentication can be used to provide additional authentication credentials when a PBX connects to the JT SIP Trunk Service.

2.6 REGISTRATION

Devices using JT SIP Trunk service will need to use the SIP registration process. This will require a username in the format of E.164@ims.jt.com e.g. (CC)44 (NDC)1534 (SN) XXXXXX@ims.jt.com known as a pilot number and a password. Both will be provided by JT.

Note: International destinations must be prefixed with either 00 or + to indicate that it is an international destination. All other national and local destinations in the E.164 format, the plus is optional, and JT will accept the call either way.

JT screens the Calling Line Identity (CLI) of calls originating from a JT SIP customer (PBX). If an invalid CLI is presented to the network, i.e. a number that doesn't belong to that customer, JT will replace the invalid CLI with the pilot number for that JT SIP Trunk service.

JT screens the CLI of all outbound calls from a SIP service (PBX) to the emergency services (999 & 112) and will present the pilot number for that trunk.

It is important that the pilot number is routable on the network and the user device (PBX). The PBX must be configured to route the pilot number to a valid extension or main number for the customer should a call be made back to the PBX as a result of CLI screening or emergency call activity.

If using DNS, user devices will need to resolve the pilot number to the IP Address(es) of JT's SBC's these will be provided by JT.

3 MEDIA SERVICES

The JT SIP Service allows the use of various media codecs between end points; however, it should be noted that to achieve a common interoperability between equipment, JT recommends minimum support for G.711 A-law and U-Law.

If a call is required to terminate on the legacy PSTN network, JT will pass the call through one of it's Media Gateways. The JT Media Gateways support the following codecs.

- G.711 - A
- G.711 - U
- G.729
- SILK
- iLBC
- G.728
- G.723
- GSM
- GSM-EFR
- GSM-HR-08
- AMR
- AMR-WB
- EVRC0
- EVRCB0
- EVRCNW0
- G.726-16
- G.726-24
- G.726-32
- G.726-40
- G.722
- G.7221
- EVS

Calls offered without support for one of these codes may be rejected by JT.

FAX calls from a customer premise using a suitable analogue gateway can be supported using the following methods:

- G.711 Pass Through
- T.38

JT recommends the use of T.38

3.1 LEGACY MEDIA SERVICES SUPPORTED

The following Legacy ISDN media services are supported on the JT SIP Service.

3.1.1 SPEECH / 3.1KHz AUDIO

Speech services are supported on the following CODECs:

- G.711 A-Law
- G.711 U-Law
- G.729

3.1.2 64KBIT/S UNRESTRICTED DIGITAL INFORMATION

Legacy 64Kbit/s Unrestricted Digital Information can be transited across the JT SIP Trunk service using the CLEARMODE codec.

Outbound calls to TDM networks with CLEARMODE codec.

3.2 LEGACY MEDIA SERVICES UNSUPPORTED

3.2.1 GROUP 4 FAX

JT does not support Group 4 Fax over JT SIP Trunk Service.

3.2.2 LEGACY TELEPHONY 7KHz

JT does not support the use of 7KHz Telephony using G.722 codec. 7KHz telephony can be supported if transmitted as CLEARMODE codec.

3.2.3 LEGACY VIDEOTELEPHONY 7KHz

JT does not support Legacy 7KHz Video Telephony over SIP.

JT Media Gateways do not support break-out of IP Video Telephony to Legacy PSTN network.

JT SIP will support IP to IP Video Telephony.

4 SIP TRUNK SUPPLEMENTARY SERVICES

The following supplementary services are provided on the JT SIP Trunk service:

Note – User server definitions. Although technically a user's device at certain stages of a call can act as both client and server. For the avoidance of doubt the term UAC (user access client) means the customers equipment attaching to the JT SIP trunk service. The term UAS (user access server) means the JT network element serving the SIP trunk.

4.1 TRUNK HUNTING

JT Provide a trunk hunting services whereby a customer's trunks can be grouped together in a trunk group to facilitate the following.

a) Sequential hunting – the SIP trunks can be configured on request to hunt sequentially. This would mean that the 1st trunk within a trunk group would receive all calls until all allocated sessions were in use. It would then start sending calls to the second trunk with the same behaviour. If, however a session became free on the first trunk it would use that.

b) Cyclic hunting – the SIP trunks can be configured to hunt across each of the trunks within a trunk group. For example, in a three trunk group the 1st call would go to trunk 1 the next call to trunk 2 and the next call to trunk 3. A subsequent call would go to back to trunk 1 and start the sequence again.

c) Weighted Hunting – The SIP trunks can have the calls to them routed as a percentage of the total calls to that destination. For example, calls over two trunks could be weighted evenly 50% to trunk 1 and 50% to trunk 2 or 80% to trunk 1 and 20% to trunk 2. The weighting is configurable within the network and will be configured as part of JT's on-boarding process.

4.2 DIRECT DIALLING IN (DDI) SINGLES

This service enables the assignment of up to 40 individual numbers to a single SIP service. The directory numbers allocated to the JT SIP Trunk service are dependent on availability within the exchange numbering plan and may not be contiguous.

The following user options are supported at the called side for the delivery of the called party number to the user.

a) The DDI Singles digits are delivered from the network to the user in the Request URI information element. The number of digits delivered from the network to the user is in full E.164 format.

b) The full national number is delivered. In this case the Request URI information element is populated in accordance with SIP Connect Technical Recommendation v1.1 with a full E.164 address.

4.3 DIRECT DIALLING IN (DDI)

This service enables a user to call directly, via the PSTN, a user on a private PBX by using the public numbering plan.

DDI are provided in ranges of 10, 100 or 1000 contiguous digits.

The following user options are supported at the called side for the delivery of the called party number to the user.

a) The DDI digits are delivered from the network to the user in the Request URI information element. The number of digits delivered from the network to the user is in full E.164 format.

b) The full national number is delivered. In this case the Request URI information element is populated in accordance with SIP Connect Technical Recommendation v1.1 with a full E.164 address.

4.4 CALLING LINE IDENTITY PRESENTATION (CLIP)

This service enables the Called Party to receive identification of the Calling Party.

This service is provided in accordance with SIP Connect Technical Recommendation v1.1:

> *The SIP-PBX MUST include a "P-Asserted-Identity" header field in the INVITE request in accordance with the rules of [RFC 3325] and [RFC 5876] unless the SIP-PBX needs to withhold the identity for privacy reasons or the SIP-PBX is performing call forwarding and is unable to assert the identity of the original caller.The screening algorithm is based on DDI and DDI singles requirements. The SIP-PBX MUST populate the "From" header field URI with a URI that the SIP PBX wishes to be used for caller identification.*

JT employs a screening algorithm to ensure that the calling line identity provided by the SIP-PBX is valid for the subscriber.

4.5 CALLING LINE IDENTITY RESTRICTION (CLIR)

This is a network-based service which enables the Caller to prevent presentation of the Calling Party number to the Called Party.

This service is provided in three variations:

Calling Line Identity Restriction: - Permanent

Calling Line Identity Restriction: - Per Call Release

Calling Line Identity Restriction: - Per Call Withheld

This service will override any CLI preferences indicated by the PBX per SIP Connect Technical Recommendation v1.1:

> *If the SIP-PBX requires privacy for a call by suppressing delivery of caller identity to downstream entities, it MUST include a "Privacy" header field with value 'id' in the INVITE request, in addition to providing an anonymous "From" header field URI.*
>
> *In cases where the Enterprise Network needs to generate an anonymous URI on behalf of a caller (as opposed to passing on a received anonymous URI), the SIP-PBX MUST send a URI of the form* ***sip:anonymous@anonymous.invalid***

4.5.1 CALLING LINE IDENTITY RESTRICTION – PER CALL RELEASE

The Calling Line Identity Release on a Per Call Basis service allows the SIP PBX to override the default Calling Line Identity Restriction on pre-defined calls.

> *If SIP-PBX requires privacy to be overridden for a call, the SIP-PBX MUST include a "Privacy" header field with value 'none' in the INVITE request.*

To release the Calling Line Identity the **called number must be prefixed with '1470'.**

4.5.2 CALLING LINE IDENTITY RESTRICTION – PER CALL WITHHELD

The Calling Line Identity Withheld on a Per Call Basis service allows the SIP PBX to restrict the Calling Line Identity on pre-defined calls.

To withhold the Calling Line Identity the **called number must be prefixed with '141'.**

4.6 PRESENTATION NUMBER

Presentation Number (PN) is a service that allows JT to specify a telephone number for the Calling Line Identity (CLI) on outgoing calls, which may be different to the main telephone number.

The specified PN will be displayed on the called person's Caller Display equipment if available or used in conjunction with the 1471 Call Return service where calls terminate on analogue lines.

Customers may require this service because their business has separate incoming and outgoing lines for effective traffic management and resilience. Alternatively, customers may wish to direct return calls to a more appropriate number, for example a central help desk.

Calling Line Identity (CLI) is used by many companies to validate incoming calls and to route calls to appropriate departments. In addition, many individuals use the Presentation Number to screen calls before answering.

4.7 CALL FORWARDING

Call forwarding can be configured as a network setting on the JT SIP Trunk Service, or alternatively as a PBX setting.

4.7.1 JT SIP CALL FORWARDING UNCONDITIONAL (CFU)

This service enables a served PBX to have the SIP service re-direct calls to another PSTN number when calls are addressed to the served PBX's MSN or DDI number.

- user control of the service is not supported
- per basic service provision is not supported
- notification to any party is not supported
- capability to include a forward-to party sub-address is not supported

4.7.2 JT SIP CALL FORWARDING BUSY (CFB)

This service enables a served PBX to have the JT SIP Trunk service re-direct calls to another PSTN number when calls addressed to the served PBX's meet the busy (engaged) tone (all channels in use).

- user control of the service is not supported
- per basic service provision is not supported
- notification to any party is not supported
- capability to include a forward-to party sub-address is not supported

4.7.3 JT SIP CALL FORWARDING NO REPLY (CFNR)

This service enables a served PBX to have the JT SIP Trunk service re-direct calls to another PSTN number when calls addressed to the served PBX's are not connected within a time period of 3 minutes.

- user control of the service is not supported
- per basic service provision is not supported
- notification to any party is not supported
- capability to include a forward-to party sub-address is not supported

4.7.4 SIP TRUNK CALL FORWARD ON UNAVAILABILITY

JT supports call forwarding on unavailability i.e. when a user (UAC) is not registered on the UAS. This can be requested as part of JT's disaster recovery service.

4.7.5 PBX CALL FORWARDING (CFU / CFB / CFNR)

This service enables calls which arrive at a PBX to be re-directed to another PSTN number, via the JT SIP Trunk service, under control of the PBX. This service is off ered in accordance with SIP Connect Technical Recommendation v1.1:

In order to forward a call, the SIP-PBX MUST send an INVITE request to JT, populated as specified in accordance with the specification, with the Request-URI identifying the forwarded-to target destination

- The "To" header field URI can identify the originally targeted destination, in which case it will not match the Request-URI;
- The "P-Asserted-Identity" header field can be absent or can assert an identity that is not an Enterprise Public Identity;
- The "From" header field URI can contain an identity that is not an Enterprise Public Identity.

4.8 CALL HOLD (CH)

This service allows a customer to interrupt communications on an existing call and then subsequently, if desired, re-establish communications.

This service is controlled by the SIP PBX and is provided in accordance with SIP Connect Technical Recommendation v1.1:

- When the hold initiator (which may be the SIP-PBX or SP-SSE acting transparently as a Media Endpoint) provides music-on-hold (MOH) treatment:
  The MOH source in the SP-SSE/SIP-PBX is based on local policy. The hold initiator MUST set the SDP directionality attribute to "a=sendonly".
- If the hold initiator does not provide MOH, it MUST set the SDP directionality attribute to "a=inactive" or "a=sendonly".
- A SP-SSE/SIP-PBX MUST support the ability to receive SDP session descriptions that have the 'c=' field set to all zeros (0.0.0.0), when the addrtype field is IPV4

4.9 INCOMING CALL BARRING – NETWORK CONTROLLED (ICB)

This service enables all incoming calls to a JT SIP Trunk service to be inhibited.

Outgoing calls from the User are unaffected by the service.

Calls incoming to the User receive an appropriate failure indication (message and optionally a tone / announcement).

The ICB service is provided by administration control only.

4.10 OUTGOING CALL BARRING – NETWORK CONTROLLED (OCB)

This service enables outgoing calls from a JT SIP Trunk service to be inhibited or restricted to certain types of call.

Incoming calls from the User are unaffected by the service.

The following types of Outgoing Call Barring are supported:

- Permanent OCB
- Administration – Controlled Pre –arranged OCB

4.10.1 PERMANENT OCB

All outgoing calls from the User are inhibited.

The calling User receives an appropriate failure indication (message and optionally tone) immediately in response to an initial setup attempt. All subsequent digits inputted are ignored.

4.10.2 ADMINISTRATION – CONTROLLED PRE – ARRANGED OCB

Outgoing calls belonging to specific classes, based on charging parameters, are barred. These classes are:

- Local calls
- National calls
- International calls
- Operator calls
- Supplementary service calls
- National premium calls

Barring combinations are permitted from the above categories.

The OCB service is provided by administration control only.

5.0 SIP USER (UAC) CONFIGURATION TABLES

Item	Configuration Information	Notes
Pilot Number	JT will provide a pilot number for registration.	It is important that the pilot number is routable on the network and the user device (PBX). The PBX must be configured to route the pilot number to a valid extension or main number for the customer should a call be made back to the PBX as a result of CLI screening or emergency call activity.
SIP Options	JT supports SIP options for heartbeat and supported Codecs.	Please refer to section 2.4 of this document for details and restrictions.

5.0 SIP USER (UAC) CONFIGURATION TABLES (Cont.)

Item	Configuration Information	Notes
P-Asserted Identity (PAI)	See adjacent note.	The SIP-PBX MUST include a “P-Asserted-Identity” header field in the INVITE request in accordance with the rules of RFC 3325 and RFC 5876 unless the SIP-PBX needs to withhold the identity for privacy reasons or the SIP-PBX is performing call forwarding and is unable to assert the identity of the original caller. The screening algorithm is based on DDI and MSN requirements. The SIP-PBX MUST populate the “From” header field URI with a URI that the SIP PBX wishes to be used for caller identification.
UAC IPv4 Address	Static /29.	JT will provide this as part of the onboarding process.
UAS - SIP Server IP address information	Site 1: Signalling: 212.9.11.200; Media: 212.9.11.201; Site 2: Signalling: 212.9.11.204; Media: 212.9.11.205;	There are two SBCs for redundancy, the SIP UAC shall only register to one at any time. The UAC may use some heartbeat or failure detection mechanism, (outside of JT scope), to detect a failure of a site and re-register to another site. If the UAC does attempt to register to both, the service will be unstable. As part of the onboarding process JT will provide a preference of the SBC to use as the primary SIP server. Note: - If the UAC does not accept timer headers amongst other parameters defined within RFC 3261 JT may provide differing IP addresses to the ones stated here. This will enable specific header manipulation and ensure interoperability.
SIP Control Plane Port	UDP 5060	SIP signalling.
SIP Media Plane Ports	20000-29999	RTP traffic in the SIP offer answer model JT’s network SBC will negotiate a supported available port with UAC as per RFC 3261 call procedures.
Use of Session Timer header	Where the UAC is compliant Session timer should be used.	JT can support UAC’s that don’t support Session timer.

5.0 SIP USER (UAC) CONFIGURATION TABLES (Cont.)

Item	Configuration Information	Notes
Response to 180 ringing	Where SDP early media is not available the UAC must play the caller ring tone on receipt of a 180- ringing message.	UK ring tone is preferred.
To/Request-URI Header format	JT mandates that Called numbers must be in the ITU-T E.164 format, JT does not mandate that the E.164 must be prefixed with a + though will accept call with or without its presence in the dial string.	e.g. (CC)44 (NDC)1534 (SN)882882 Short codes that trigger Supplementary Services e.g. 141, must be in the following format: 141(+)441534882882 Short codes for national routable services e.g. 118118 must be in the following format: (+)441534118118 Please see Appendix A for reference.
PBX Session configuration	JT recommends that the PBX session licencing matches the number of purchased sessions.	
Registration	JT will respond with a different NONCE for each registration attempt.	

6 SERVICE TEMPLATE

Supplementary Service	Default	Ex-Directory	On Request
Calling Line Identity Presentation	Yes		
Calling Line Identity Restriction Permanent			Yes
Calling Line Identity Release per Call		Yes	
Calling Line Identity Withheld per Call	Yes		
Presentation Number			Yes
Call Forwarding	Yes		
Incoming Call Barring			Yes
Outgoing Call Barring			Yes
E.164 Digit Presentation	Yes		
Static Mode			Yes
Registration Mode	Yes		

7 APPENDIX A

Destination	Number Format	Notes
Local	(+)441534xxxxxx	
Local/National Mobile	(+)447797xxxxxxx	JT example
National Landline	(+)4420xxxxxxxx	London area code example
International	+ or 00 33xxxxxxxxxxx	French example
Emergency	999/112 or (+)441534999/112	
Routable Short Codes	100/123/1xx or (+)441534100/123/1xx	Example JT prefer E.164 format
Supplementary Services	141(+)441534xxxxxx	Local restrict CLI given as example, National and International follow same format

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