ATTACHMENT C

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AT&T ESInet to i3 PSAP CPE Interface Specification

Version 2.0 – 01/27/2012

xSR™ AT&T ESInet to i3 PSAP CPE Interface

Contents

Notices

This document describes AT&T's ESInet Services interfaces to PSAP CPE that support the National Emergency Number Association's i3 architecture as it is proposed in NENA's 08-002 document entitled 'NENA Functional and Interface Standards for Next Generation 9-1-1 Version 1.0 (i3)'.

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Abstract

This document specifies the interfaces and functions provided by AT&T's ESInet to i3 capable PSAP CPE. This document also specifies the interactions and associated interfaces used for emergency services call handling within the i3 environment. It is a technical document intended for a technical audience including software developers and architects.

AT&T's ESInet supports the i3 architecture as it is proposed in NENA's 08-002 document entitled 'NENA Functional and Interface Standards for Next Generation 9-1-1 Version 1.0 (i3).' The ESInet currently implements the core functionality of the i3 specification and is continually being updated to further comply with future enhancements. AT&T's ESInet utilizes microDATA's xSR™ which implements the following core i3 functions:

ESRP – Emergency Services Routing Proxy; xSR™ routes calls based on location information to downstream ESRPs

LIF – Location Interwork Function; xSR™ creates a PIDF-LO as needed based on legacy ALI information for all call types.

NIF – NG911 Interwork Function; xSR™ uses an ECRF to spatially (i.e. derived from geo/civic location information) determines the correct PSAP to deliver emergency calls.

ECRF – Emergency Call Routing Function; PSAPs may use this ECRF for routing and location based selective transfer information.

PRF – Policy Routing Function; xSR™ stores and utilizes policy and state information which determines alternate PSAP routing.

SDB – Subscriber Database; xSR™ provides a web service for retrieval of additional information relating to call location

SIP Conferencing Focus; xSR™ will provide RTP media mixing capabilities for up to 6 participants without the need for PSAPs to have their own conferencing servers.

About this Document

This document is intended to provide interface specifications for i3 compatible PSAP CPE. Therefore, no attempt has been made to describe the physical makeup of the AT&T ESInet system. The AT&T ESInet system xSR™ system is intended to be highly reliable, with no single point of failure, load balanced, and highly redundant.

Terms such as 'ESInet', 'xSR™', 'Subscriber Database', and others should be interpreted functionally, not physically. AT&T's ESInet is designed with multiple layers of diversity and redundancy to help avoid single points of failure.

TCP Protocol

TCP and UDP

End to end testing between the AT&T ESInet solution and certain call handling vendors has revealed that, in some cases, there is packet fragmentation possible during call set up. In particular, if the size of the SIP Invite is within 200 bytes of the the Maximum Transmission Unit (MTU) of the underlying layer 2, fragmentation is likely to occur. Fragmentation may have impacts ranging from call set-up delays of unknown duration and quantity, to blocked or abandoned calls. In some instances, fragmentation has no discernible impact to the call.

Packet fragmentation is not unexpected, and it can be handled appropriately with the use of TCP packet transmission protocol. Another protocol, UDP, is commonly used in VoIP implementations. This protocol differs from TCP, and its mechanisms for handling packet fragmentation are weaker.

While AT&T's ESInet solution can support both UDP and TCP packet transmission protocols, AT&T recommends that TCP be used. This recommendation is based upon the packet size experienced within AT&T's ESInet solution, the anticipated growth of such packet sizes with forward looking NG 9-1-1 message sets, and applicable standards, including the NENA i3 specification.

AT&T supports NENA i3 standards and strongly recommends that TCP be utilized by Call Handling solutions interfacing with AT&T's ESInet solution. See Appendix I: TCP References for standards information.

Receiving Calls

One of the most important functions of the xSR™ AT&T ESInet is to deliver emergency calls to PSAPs. Therefore, PSAPs must be ready to receive SIP calls and, as needed, deliver these calls to the most appropriate call taker. CPE providers should assume that xSR™ does not recognize the state of any call taker centric ACD related information (e.g. longest idle call taker, station priorities, etc). This functionality, if required, should be provided by the CPE Vendor.

xSR™ will attempt to deliver location information with the call, if available. In the event xSR™ receives an ECRIT (IP) call with an embedded PIDF-LO (RFC 5139 etc), this PIDF-LO will be relayed to the PSAP. If xSR™ receives a call without location information (e.g. from a Legacy Network Gateway), xSR™ will query ALI to determine location information sufficient to define a PIDF-LO. This crafted PIDF-LO will then be passed to PSAPs in the SIP INVITE. xSR™ will always use valid PIDF-LO fields as documented in the NENA Civic Location Exchange Format (NENA work in progress).

Here's an example PIDF-LO format created by xSR™ (complies with RFC 5139):


```
<civicAddress xml:lang="en-US" xmlns="urn:ietf:params:xml:ns:pidf:geopriv10:civicAddr">
    <country>US</country>
    <A1>Vermont</A1>
    <A3>Saint Johnsbury</A3>
    <A6>US Route 5</A6>
    <HNO>1016</HNO>
    <NAM>microDATA GIS</NAM>
    <PC>05819</PC>
</civicAddress>
```


The PIDF-LO passed within the SIP INVITE to PSAPs can be provided by value (LbyV) or by reference (LbyR). For all ECRIT (IP) originated 911 calls, which format is provided depends entirely on the PIDF-LO originally sent by the 911 caller. xSR™ will always dereference a PIDF-LO received by reference and send the resulting value and the reference to the PSAP. For all wire line PSTN calls, location will be passed by value. For all wireless calls, xSR™ will provide both location by value and location by reference so that the i3 capable PSAP may obtain updates to location as needed.

Since ultimately a call taker must answer all 911 calls originating from the xSR™, it is the responsibility of the PSAP CPE to deliver inbound calls from the xSR™ to the correct call taker. The xSR™ will simply deliver SIP calls to the predefined SIP URI of the PSAP. The PSAP SIP endpoint may answer this INVITE normally and then establish a bridged connection to the call taker in a B2BUA fashion if desired. Alternatively, the PSAP SIP endpoint may respond back to the xSR™ with a 'SIP 302 Temporarily Moved' response after applying ACD intelligence to determine the most appropriate call taker to deliver the 911 call to. Another way is to have the PSAP Proxy Server communicate with a PSAP Redirect Server to determine the correct call taker. Yet another way which is acceptable is for the PSAP to answer the call, make a call announcement (e.g. 'please hold for the next available operator'), and then send a REFER back to xSR™. Any method which correctly establishes a 2-way RTP stream from the xSR™ to the call taker is acceptable.

B2BUA style PSAP CPE - Call Flow

```
xSR™-Focus                                   PSAP                                   Call taker
|                                             |                                      |
| xSR™-Focus delivers 911 call to PSAP.       |                                      |
|                                             |                                      |
| INVITE sip:esrp@psap1.e911.abc.gov          |                                      |
|------------------------------------------->  |                                      |
|              180 Ringing                    |                                      |
| <-------------------------------------------  |                                      |
|              200 OK                         |                                      |
| <-------------------------------------------  |                                      |
|              ACK                            |                                      |
| ------------------------------------------>  |                                      |
|              RTP                            |                                      |
| <==========================>               |                                      |
|                                             |                                      |
|                                             |PSAP ESRP/CPE determines most correct call taker
|                                             |                                      |
|                                             | INVITE sip:calltaker1@psap1.e911.abc.gov
|                                             | ------------------------------------> |
|                                             |          180 Ringing                  |
|                                             | <------------------------------------ |
|                                             |          200 OK                       |
|                                             | <------------------------------------ |
|                                             |          ACK                          |
|                                             | ------------------------------------> |
|                                             |          RTP                          |
|                                             | <=====================>               |
|                                             | PSAP CPE/ESRP bridges inbound and outbound calls
|              RTP                            |          RTP                          |
| <=============================>             | <======================>              |
```

1ST Party Call Control Style PSAP CPE - Call Flow

```
xSR™-Focus                                   PSAP                                   Call taker
|                                             |                                      |
| xSR™-Focus delivers 911 call to PSAP.       |                                      |
|                                             |                                      |
| INVITE sip:esrp@psap1.e911.abc.gov          |                                      |
|------------------------------------------->  |                                      |
|                                             |                                      |
|    PSAP CPE/ESRP Redirect Server determines most correct call taker                |
|                                             |                                      |
```

```
|        302 Temporarily Moved                 |                          |
| <---------------------------------------------  |                          |
|                  ACK                          |                          |
| --------------------------------------------->  |                          |
| INVITE sip:call taker1@psap1.e911.abc.gov                                  |
| --------------------------------------------------------------------------> |
|                                               |        180 Ringing       |
| <-------------------------------------------------------------------------- |
|                                               |        200 OK            |
| <-------------------------------------------------------------------------- |
|                  ACK                          |                          |
| --------------------------------------------------------------------------> |
|                                              RTP                         |
| <=========================================================================> |
```

1st Party Call Control Style with Proxy - Call Flow

```
xSR™-Focus                          PSAP-Proxy                      Redirect Svr      Call taker
|                                           |                               |                    |
| xSR™-Focus delivers 911 call to PSAP.     |                               |                    |
|                                           |                               |                    |
| INVITE sip:esrp@psap1.e911.abc.gov        |                               |                    |
|-------------------------------------------->                              |                    |
|                                           |INVITE sip:esrp@psap1.e911.abc.gov                  |
|                                           | ------------------------------------> |            |
|                                           |                               |                    |
|                                           |     Redirect Server determines correct call taker  |
|                                           |                               |                    |
|                                           |    302 Temporarily Moved      |                    |
|                                           | <-----------------------------------  |            |
|                                           |              ACK              |                    |
|                                           | ------------------------------------> |            |
|                                           | INVITE sip:calltaker1@psap1.e911.abc.gov           |
|                                           | ------------------------------------------------------------------> |
|                                           |                       180 Ringing                  |
|                                           | <------------------------------------------------------------------ |
|          180 Ringing                      |                               |                    |
| <---------------------------------------------- |                         |                    |
|                                           |                       200 OK  |                    |
|                                           | <------------------------------------------------------------------ |
|          200 OK                           |                               |                    |
| <---------------------------------------------- |                         |                    |
|           ACK                             |                               |                    |
```

```
| ----------------------------------------------> |                       |                    |
|                                                 |                      ACK                   |
|                                                 | ------------------------------------------------------------------> |
|                                   RTP                                                        |
| <===========================================================================>|
```

1ST Party Call Control Style with REFER - Call Flow

```
xSR™-Focus                                     PSAP                                   Call taker
|                                                |                                        |
| xSR™-Focus delivers 911 call to PSAP.         |                                        |
|                                                |                                        |
| INVITE sip:esrp@psap1.e911.abc.gov            |                                        |
|--------------------------------------------->  |                                        |
|              180 Ringing                       |                                        |
| <--------------------------------------------  |                                        |
|               200 OK                           |                                        |
| <--------------------------------------------  |                                        |
|                ACK                             |                                        |
| --------------------------------------------> |                                        |
|                RTP                             |                                        |
| <==========================>                  |                                        |
|                                                |                                        |
| PSAP makes  announcements to caller and finds an available call taker                 |
|                                                |                                        |
| REFER (Refer-To: sip:calltaker1@psap1.e911.abc.gov)                                   |
| <--------------------------------------------  |                                        |
|             202 Accepted                       |                                        |
| --------------------------------------------> |                                        |
|              NOTIFY                            |                                        |
| --------------------------------------------> |                                        |
|               200 OK                           |                                        |
| <--------------------------------------------  |                                        |
|                                                |                                        |
|     PSAP removes itself from the call since it is no longer needed                     |
|                                                |                                        |
|                BYE                             |                                        |
| <--------------------------------------------  |                                        |
|               200 OK                           |                                        |
| --------------------------------------------> |                                        |
|                                                |                                        |
| xSR™-Focus attempts to reach Refer-To address for call taker                          |
```

```
|          INVITE                |                   |
|------------------------------------------------------->|
|                                |    180 Ringing    |
|<------------------------------------------------------ |
|                                |      200 OK       |
|-------------------------------------------------------> |
|                                |       ACK         |
|<------------------------------------------------------ |
|                              RTP                    |
|<======================================================> |
```

Refusing Calls

Calls sent from the xSR™ to a PSAP can be rejected/refused by the PSAP as needed. If xSR™ receives a 4xx, 5xx, or 6xx SIP response to an INVITE, it will query the xSR™ Policy Routing Function (PRF) data to determine alternate PSAP(s). It is entirely up to the PSAP to determine when to refuse calls. xSR™ will not attempt to limit the number of calls to a PSAP. Rather, xSR™ will assume that the PSAP will refuse calls which it cannot handle.

It is possible (although possibly undesirable due to delayed call delivery) for a PSAP to accept and queue a call from xSR™ if no call takers are available. In this case, xSR™ will treat the call as delivered and will not attempt any alternate routing.

If a CPE supports multiple PSAPs, it is free to alternate route to any available PSAP. However, xSR™ will treat the call as delivered to its originally intended PSAP. xSR™ will only engage its alternate routing mechanisms when it receives a 4xx, 5xx, or 6xx SIP response to its INVITE.

If a CPE wants to refuse calls but have xSR™ redirect the call to a specific alternate SIP URI, it may do so by replying with a 302 Redirect response to the xSR™ INVITE.

Virtual Trunk Groups

xSR™ supports a concept of Virtual Trunk Groups to approximate the concept of physical trunk groups found in a traditional TDM deployment. Many CPE vendors use physical trunk groups to know where to route calls and this is important for proper operation. XSR™ satisfies this requirement by providing the name of a Virtual Trunk Group on the user-part of the SIP TO header. This user-part may be alphanumeric and is pre-configured within the PRF data of xSR™.

xSR™ determines which Virtual Trunk Group to use based on a variety of data including location of caller (from PIDF-Lo or ALI), predefined routes, etc. If a CPE does not distinguish between calls for different trunk groups or only supports a single trunk group, this user-part of the SIP TO header may be ignored.

Adding Call Participants

The job of adding new call participants to existing calls necessarily involves a mixing of RTP streams. The SIP endpoint which performs this mixing can be the xSR™ or a downstream media server located within the PSAP. There are 3 basic types of call participants which can be added to an existing call: Admin Trunks, Stations, and Focus Trunks.

Admin Trunks are trunks typically provided by a PSAP PBX. Since these trunks are PSAP centric, it is the responsibility of the PSAP CPE to provide the necessary conferencing focus or a SIP client to accomplish this mixing. The xSR™ in this scenario will know nothing about the addition or removal of these Admin Trunks since these are downstream from it and no SIP requests or responses to the xSR™ are generated as a result of adding/removing Admin Trunks. Warning: The list of 'users' in the Conference Package presented on the NOTIFY from xSR™ will not contain Admin Trunks. Admin Trunks should be used for all outbound dialing to destinations which are not selective transfer agencies.

Stations are call taker SIP UAs. This could be a SIP Phone or CTI workstation. There are times during the course of a 911 call in which additional call taker stations need to be added as participants (e.g. supervisor joins or monitors a call). In this case, it is the responsibility of the PSAP CPE to provide the necessary conferencing focus to accomplish this mixing. No SIP requests or responses are expected by xSR™ when Stations are added or removed. Warning: The list of 'users' in the Conference Package presented on the NOTIFY from xSR™ will not contain Stations added in this fashion. This 'user' list will only include the original station dialed by xSR™.

Focus Trunks are either Legacy PSTN trunks or SIP trunks terminated at the xSR™. These trunks are not directly accessible by PSAPs. They are, however, indirectly accessible for selective transfers. xSR™ is the endpoint for all selective transfer mixing. This guarantees an available outbound trunk and also greatly minimizes bandwidth requirements at each PSAP. When the xSR™ is the conferencing focus, all RTP streams are terminated at it and therefore no additional RTP streams are required from each PSAP.

Where LoST is deployed, the actual URI used within the REFER-TO header can be determined via the ECRF function of the xSR™ solution. See 'Using ECRF to Determine Responders where LoST is Deployed' for more details on how to do this.

Focus Trunks can be added to existing 911 calls sent through the xSR™ through the SIP REFER request. RFC 4579 section 5.5 describes the procedure required to add a new resource to a conference. The

PSAP specifies the URI to add within the 'Refer-To' header. This URI can be a standard SIP URI (sip:xxxx@xxxx) or a global TEL URI (tel:xxxxxxxxxxx) according to RFC 3966. If a TEL URI is specified, xSR™ will add the party through one of its legacy gateways. The following call flow can be expected:

Addition of PSTN based party – Call Flow

```
PSAP                                    xSR™-Focus                          Legacy Gateway
|                                        |                                   |
| PSAP requests the addition of a PSTN party to an existing call.            |
|                                        |                                   |
|              RTP                       |                                   |
| <============================>         |                                   |
|REFER sip:focus@ecdc1.e911.abc.gov Refer-To: tel:+1-802-748-5503
|---------------------------------------------->  |                          |
|          202 Accepted                  |                                   |
| <----------------------------------------------  |                         |
|          NOTIFY (Trying)               |                                   |
|  <----------------------------------------------  |                        |
|            200 OK                      |                                   |
|  ---------------------------------------------->  |                        |
|                                        |                                   |
|      xSR™ dials out to join +1-802-748-5503 to the call          |         |
|                                        |                                   |
|                                        | INVITE tel:+1-802-748-5503
Contact:focus@ecdc1.e911.abc.gov; isfocus
|                                        | -------------------------------------> |
|                                        |        180 Ringing                |
|                                        | <------------------------------------- |
|             180 Ringing                |                                   |
| <--------------------------------------------  |                           |
|                                        |          200 OK                   |
|                                        | <------------------------------------  |
|                                        |           ACK                     |
|                                        | -------------------------------------> |
|                                        |           RTP                     |
|                                        | <=====================>           |
|          NOTIFY (OK)                   |                                   |
|  <----------------------------------------------  |                        |
|          200 OK                        |                                   |
|  ---------------------------------------------->  |                        |
```

Addition of SIP UA party – Call Flow

```
PSAP                                   xSR™-Focus                    SIP UA
|                                      |                             |
| PSAP requests the addition of a SIP UA party to an existing call.  |
|                                      |                             |
|              RTP                     |                             |
| <==============================>     |                             |
|REFER sip:focus@ecdc1.e911.abc.gov Refer-To: sip:police@county1.abc.gov
|---------------------------------------------->  |                  |
|            202 Accepted              |                             |
| <----------------------------------------------  |                 |
|           NOTIFY (Trying)            |                             |
| <----------------------------------------------  |                 |
|              200 OK                  |                             |
|  ---------------------------------------------->  |                |
|                                      |                             |
|     xSR™ "dials out" to join sip:police@county1.abc.gov to the call
|                                      |                             |
|                                      | INVITE sip:police@county1.abc.gov; isfocus
|                                      | -------------------------------------> |
|                                      |        180 Ringing          |
|                                      | <------------------------------------- |
|                                      |          200 OK             |
|                                      | <------------------------------------  |
|                                      |           ACK               |
|                                      | -------------------------------------> |
|                                      |           RTP               |
|                                      | <=====================>     |
|          NOTIFY (OK)                 |                             |
| <----------------------------------------------  |                 |
|            200 OK                    |                             |
|  ---------------------------------------------->  |                |
```

Removing Call Participants

Call participants are removed similarly to the way that they are added. The only significant difference between adding and removing from a SIP point of view is the addition of Refer-To header method of 'BYE'. The Refer-To header field value indicates exactly which party should be removed. See RFC 4579 section 5.11 for details. You can optionally specify a TEL URI of 'tel:0000000000' which will tell xSR™-Focus to remove the last leg of the call. The call flow is as follows:

```
PSAP                                    xSR™-Focus                         Legacy Gateway
|                                          |                                   |
| PSAP requests the removal of a PSTN party from an existing call.             |
|                                          |                                   |
|                RTP                       |                                   |
| <==========================>             |                                   |
|REFER sip:focus@ecdc1.e911.abc.gov Refer-To: tel:+1-802-748-5503; method=BYE
|------------------------------------------->  |                               |
|           202 Accepted                   |                                   |
| <--------------------------------------------|                               |
|          NOTIFY (Trying)                 |                                   |
|  <-------------------------------------------|                               |
|              200 OK                      |                                   |
|  ------------------------------------------->|                               |
|                                          |                                   |
|     xSR™ removes  Legacy Gateway participant +1-802-748-5503 from the call
|                                          |                                   |
|                                          | BYE tel:+1-802-748-5503           |
|                                          | ------------------------------------> |
|                                          |           200 OK                  |
|                                          | <------------------------------------ |
|          NOTIFY (OK)                     |                                   |
|  <-------------------------------------------|                               |
|           200 OK                         |                                   |
|  ------------------------------------------->|                               |
```

Knowing Call Information

Call participants will change over the course of any call. Since xSR™ is a conference focus, it will be the first to recognize when participants have been successfully added or removed. As such, it is the most appropriate SIP UA to publish this call information. xSR™ has the ability to NOTIFY subscribers with call information according to RFC 4575, the 'conference package.' It is highly recommended that PSAPs subscribe to this 'conference package' as this is truly the only way that PSAPs can maintain an accurate representation of call participants.

As with any other subscription, an initial NOTIFY with full state information is sent as soon as a PSAP subscribes to this package. In addition, xSR™ will send NOTIFYs any time the state of the conference changes (e.g. every time a call participant is added or removed, by request or spontaneously).

The following is an example SUBSCRIBE/NOTIFY exchange between a PSAP and xSR™. Note that the SUBSCRIBE request is sent to the FROM header value identifying the specific call previously established on the xSR™. Also note that the 'Call-ID' header value is created by the PSAP and is new, unique and different from the call-ID value of any previously established SIP dialog. This call-ID will be used within all subsequent NOTIFY messages to uniquely identify this subscription.

```
PSAP                                     xSR™-Focus
|                                         |
| PSAP subscribes to the 'conference' event package, after a call has been established.
|                                         |
|                RTP                      |
| <============================>          |
|SUBSCRIBE 8027485503@xs1.ecdc1.e911.abc.gov to 'conference' event
|---------------------------------------------->  |
|              200 OK                     |
| <----------------------------------------------  |
|     NOTIFY (w/full state information)   |
|  <----------------------------------------------  |
|              200 OK                     |
|  ---------------------------------------------->  |
```

The following information will be included in the <conference-info> document included in the body of the NOTIFY:

Conference-description
Host-info
Conference-state
Users
Sidebars-by-ref
Sidebars-by-val

In addition, there are some attributes defined for the <conference-info> root element tag:

Entity: this is the xSR™ SIP URI for a single call (e.g. sip:8027485503@172.12.101.35, where 8027485503 is the ANI of the 911 caller and 172.12.101.35 is the IP address of the xSR™)
State: always set to "full"
Version: a sequence number assigned by xSR™ which provides ordering to received notifications

Conference-description

This element describes the conference as a whole and includes the following child elements:

Display-text: set to "conference with <CBN>", where <CBN> is the callback number or SIP URI of the 911 caller (e.g. "conference with 802-748-5503" or "conference with sip:ipuser@userdomain.com")

Subject: set to "<CBN>" where <CBN> is the callback number of the 911 caller

Maximum-user-count: set to the maximum number of participants allowed in the conference call

Available-media

- o Entry with label attribute set to "media"
  - Display-text: set to same value as type below
  - Type: set to "audio", "video", or "text"
  - Status: set to "sendrecv" if bi-directional media streaming is possible, "sendonly" if the media stream is streaming towards the PSAP but not back up to the xSR™, and "recvonly" if the media is streaming from the PSAP to the xSR™, but not from the xSR™ to the PSAP

Host-Info

This element describes the xSR™ itself and includes the following elements:

Display-text: set to "xSR <focus-id>", where <focus-id> is a unique name given to the particular xSR™ delivering the call

Conference-State

User-count: set to the actual number of participants in the conference; will match the number of entries in the user container

Active: always set to "true"

Locked: always set to "false" unless the conference has reached its capacity

Users

The Users element is a container for user child elements, each describing a conference participant. xSR™ will always send a complete list of users. No partial lists will ever be sent. Each user element has the following attributes:

Entity: the SIP URI of the participant (e.g. "tel:+1-802-748-5503"or "sip:station5@psap1.abc.gov")

State: always set to "full"

Display-text: set to the user-friendly name of the participant (e.g. "Mary Jane" or "Bob Jones"); note that this is different than the name of the device being used by the user.

Associated-aors: not included

Roles: not included

Languages: not included, but will be used in the future when knowledge the native language of each participant is available

Cascaded-focus: not included

Endpoint: this is a container element included for each SIP endpoint associated with a user. It includes the following attributes:

- Entity: the Contact URI (e.g. tel:xxxx or sip:xxxx) of this endpoint
- State: always set to "full"
- The following child elements are included for the endpoint element:
  - Display-text: the user-friendly name of the device being used by the user to terminate this endpoint (e.g. 'Station 5, PSAP 100' or 'WSPC123')
  - Referred: the SIP URI of the requesting user who originally requested this user be added to the conference

    When: not included

    Reason: not included
  - Status: set to

    "connected" when this endpoint is participating as a full-duplex participant

    "alerting" when this endpoint is being called by xSR™

    "on-hold" when this endpoint is holding

    "pending" when this endpoint is parked or queued

    "alerting" when this endpoint is ringing

    "dialing-out" when this endpoint is being dialed

    "disconnected" when this endpoint is hung up or otherwise no longer participating in conference
  - Joining-method: set to "dialed-out" for call participants that xSR™ has dialed; set to "dialed-in" for call participants who have called xSR™
  - Joining-info

    When: the UTC (Zulu time) indicating when this user was added to the conference; time is always in the form of yyyy-mm-ddThh:mm:ssZ (e.g. '2008-03-20T14:12:02Z')

    Reason: always set to 'SIP;text="Ad-hoc Invitation"'
  - Disconnection-method: not included since only connected or connecting endpoints are included
  - Disconnection-info: not included since only connected or connecting endpoints are included
  - Media: not included (nor any of its children); future versions will support this element

Sidebars-by-ref: not included

Sidebars-by-val: not included

Call Information Example

The following is an example of what this <conference-info> element might look like:


```
<conference-info version=”0” state=”full” entity=sip:8027485503@172.12.101.35>
        <conference-description>
                <display-text>conference with 8027485503</display-text>
                <subject>8027485503</subject>
                <maximum-user-count>8</maximum-user-count>
                <available-media>
                        <entry label=”media”>
                                <display-text>audio</display-text>
                                <type>audio</type>
                                <status>sendrecv</status>
                        </entry>
                </available-media>
        </conference-description>
        <host-info>
                <display-text>xSR AppServer1</display-text>
        </host-info>
        <conference-state>
                <user-count>3</user-count>
                <active>true</active>
                <locked>false</locked>
        </conference-state>
        <users>
                <user entity=”tel:8027485503>
                        <state>full</state>
                        <display-text>8027485503</display-text>
                        <endpoint entity=”tel:8027485503”>
                                <state>full</state>
                                <display-text>8027485503 Endpoint</display-text>
                                <referred>
                                        <by>sip:AppServer1@ecdc1.abc.gov</by>
                                </referred>
                                <status>connected</status>
                                <joining-method>dialed-in</joining-method>
                                <joining-info>
                                        <when>2008-03-20T02:35:02Z</when>
                                        <reason>SIP;text=”Ad-hoc Invitation”</reason>
                                </joining-info>
                        </endpoint>
                </user>
                <user entity=”sip:call taker-station@psap1.abc.gov>
                        <state>full</state>
                        <display-text>call taker-station</display-text>
                        <endpoint entity=”sip:call taker-station@psap1.abc.gov”>
                                <state>full</state>
                                <display-text>call taker-station Endpoint</display-text>
                                <referred>
                                        <by>sip:AppServer1@ecdc1.abc.gov</by>
                                </referred>
                                <status>connected</status>
                                <joining-method>dialed-out</joining-method>
                                <joining-info>
                                        <when>2008-03-20T02:35:12Z</when>
                                        <reason>SIP;text=”Ad-hoc Invitation”</reason>
                                </joining-info>
```

```
                    </endpoint>
          </user>
          <user entity="sip:police@mycity.abc.gov>
                    <state>full</state>
                    <display-text>myCity Police</display-text>
                    <endpoint entity="sip:police@mycity.abc.gov">
                              <state>full</state>
                              <display-text>myCity Police Endpoint</display-text>
                              <referred>
                                        <by>sip:AppServer1@ecdc1.abc.gov</by>
                              </referred>
                              <status>connected</status>
                              <joining-method>dialed-out</joining-method>
                              <joining-info>
                                        <when>2008-03-20T02:36:25Z</when>
                                        <reason>SIP;text="Ad-hoc Invitation"</reason>
                              </joining-info>
                    </endpoint>
          </user>
     </users>
</conference-info>
```

Obtaining ALI and Location Updates

During transition from the traditional 911 system to i3, it is critical that the traditional call information, ALI, continues to be provided with calls. In many AT&T ESInet deployments, traditional ALI query methods will continue to be used to supply the ALI information. In these situations, the CPE will use the ANI contained in the SIP header to query the ALI database for initial ALI information and will re-query the ALI database for Location updates. The format for the ALI response is dependent on the serving ALI Database and outside the scope of this specification.

Obtaining ALI and Location Updates when AQS is Provided

In the i3 environment, there can be many entities which supply location information for a call such as telematics providers, carriers, government databases, etc. NENA has defined an XML based protocol, XML ALI Query Service (AQS) that defines the set of XML messages and message exchange patterns to be used in the delivery of XML ALI. Using AQS, the query key provided in the SIP header (in the form of an NPA NXX TN or a SIP URI) is used to query XML ALI. In ESInet deployments where AQS is available, please refer to NENA 04-005, NENA ALI Query Service Standard, for specific protocol information.

Using ECRF to Determine Responders where LoST is Deployed

LoST may not be available in some AT&T ESInet deployments. Where LoST is deployed, the xSR™ ECRF service implements the LoST protocol as defined by RFC 5222. It supports listServices and findService methods. The listServicesByLocation query can be used to determine which services are available for a particular location. The findService query can be used to determine SIP URIs for the services returned in the listServicesByLocation query..

The ECRF service supports both a HTTP POST query method and a SOAP query method. If using SOAP, the following is the WSDL specification supported:

```
<?xml version="1.0" encoding="UTF-8"?>
<wsdl:definitions xmlns:wsdl="http://schemas.xmlsoap.org/wsdl/" xmlns:soap="http://schemas.xmlsoap.org/wsdl/soap/"
xmlns:http="http://schemas.xmlsoap.org/wsdl/http/" xmlns:xs="http://www.w3.org/2001/XMLSchema"
xmlns:soapenc="http://schemas.xmlsoap.org/soap/encoding/" xmlns:mime="http://schemas.xmlsoap.org/wsdl/mime/"
xmlns:tns="http://new.webservice.namespace" xmlns:cl="urn:ietf:params:xml:ns:pidf:geopriv10:civicLoc"
xmlns:lost="urn:ietf:params:xml:ns:lost1" xmlns:gml="http://www.opengis.net/gml" xmlns:ns="http://www.opengis.net/gml"
targetNamespace="http://new.webservice.namespace">
	<wsdl:types>
		<xs:schema targetNamespace="urn:ietf:params:xml:ns:pidf:geopriv10:civicLoc">
			<xs:include schemaLocation="civicLoc.xsd"/>
		</xs:schema>
		<xs:schema targetNamespace="urn:ietf:params:xml:ns:lost1">
			<xs:include schemaLocation="lost.xsd"/>
		</xs:schema>
		<xs:schema targetNamespace="http://www.opengis.net/gml">
			<xs:include schemaLocation="gml.xsd"/>
		</xs:schema>
	</wsdl:types>
	<wsdl:message name="FindServiceRequestMessage">
		<wsdl:part name="parameter" element="lost:findService"/>
	</wsdl:message>
	<wsdl:message name="NewMessageResponse">
		<wsdl:part name="parameter" element="lost:listServicesResponse"/>
	</wsdl:message>
	<wsdl:message name="ListServicesRequestMessage"/>
	<wsdl:message name="NewMessage2">
		<wsdl:part name="parameter" element="lost:listServices"/>
	</wsdl:message>
	<wsdl:message name="NewMessage3"/>
	<wsdl:message name="NewMessage4"/>
	<wsdl:portType name="SOAPPort">
		<wsdl:operation name="FindService">
			<wsdl:input name="FindServiceRequest" message="tns:FindServiceRequestMessage"/>
			<wsdl:output message="tns:NewMessageResponse"/>
		</wsdl:operation>
		<wsdl:operation name="ListServices">
			<wsdl:input name="ListServicesRequest" message="tns:ListServicesRequestMessage"/>
			<wsdl:output message="tns:NewMessageResponse"/>
		</wsdl:operation>
	</wsdl:portType>
	<wsdl:binding name="LoSTSOAP" type="tns:SOAPPort">
		<http:binding verb="POST"/>
		<wsdl:operation name="FindService">
			<http:operation location="urn:#findService"/>
```

```
				<wsdl:input>
					<mime:content type="mimeXml"/>
				</wsdl:input>
				<wsdl:output>
					<mime:mimeXml/>
				</wsdl:output>
			</wsdl:operation>
			<wsdl:operation name="ListServices">
				<http:operation location="urn:#NewOperation"/>
				<wsdl:input name="ListServicesRequest">
					<mime:content type="mimeXml"/>
				</wsdl:input>
				<wsdl:output>
					<mime:mimeXml/>
				</wsdl:output>
			</wsdl:operation>
		</wsdl:binding>
		<wsdl:service name="LoST">
			<wsdl:port name="LoSTPort" binding="tns:LoSTSOAP">
				<http:address location="http://qalab20/xLoST.asmx"/>
			</wsdl:port>
		</wsdl:service>
</wsdl:definitions>
```

Indicating PSAP Presence

It is important that the xSR™ maintain as accurate as possible presence information for PSAPs. The xSR™ PRF (Policy Routing Function) will be used not only for maintaining various PSAP policies but also for maintaining a state element used to track PSAP presence. There are many ways of reporting state, each having its advantages and disadvantages.

xSR™ currently maintains state via a simple SIP OPTIONS request/response mechanism. xSR™ will periodically send OPTIONS requests and will assume a PSAP is available so long as it continues to respond to it.

The long term method for knowing PSAP state is via a SUBSCRIBE/NOTIFY mechanism, as detailed in the i3 stage 3 specification. xSR™ will support both subscription (for downstream entities) and notification (for upstream entities) services. Currently, support for this mechanism is mandated in both the emerging i3 specification as well as other standards. The event packages associated with this function are still being developed.

Reason for Revision of Specification

Issue Number	Date of Issuance	Reason for Subsequent Issuance	Page numbers
2.0	01/27/12	Added Table of Contents	2 - 3
		Added TCP Protocol	6
		Updated Call Information example	20-21
		Combined Obtaining ALI (optional) and Location Updates sections, updated the text and renamed the section Obtaining ALI and Location Updates.	20
		Added Obtaining ALI and Location Updates when AQS is Provided.	20
		Expanded language to clarify listServicesByLocation and findService queries	22
		Added Reason for Revision of Specification	24
		Added Appendix 1: TCP References	25

Appendix 1: TCP References

IETF - RFC-3261

Excerpt: "If a request is within 200 bytes of the path MTU, or if it is larger than 1300 bytes and the path MTU is unknown, the request MUST be sent using an RFC 2914 [43] congestion controlled transport protocol, such as TCP.

ATIS - ATIS-0500019.2010

Excerpt: "Signaling 0200-0100 Signaling shall be supportable over UDP and TCP with or without TLS security. PSAP policy shall govern which of these transport mechanisms are acceptable." Rationale: Further NENA recommendations have stated TCP as the primary delivery mechanism for SIP with a fall back to UDP. Also, the various ATIS NGN standards recommend TCP. Therefore it is appropriate for the RFAI to support both TCP and UDP."

NENA - NENAi3TechnicalRequirementsDocument

Excerpt: "4.1.12 Transport

SIP signaling within the ESInet must be TCP with TLS. Fallback to UDP is allowed. However emergency call messages have many large elements, for example, a PIDF-LO, and are more likely to be fragmented when carried in UDP."