# Tennessee Emergency Communications Board
# Reimbursement Requirements For
# NG911 CONTROLLER AND/OR ADJUNCT EQUIPMENT

*Effective January 3, 2012*

The Tennessee Emergency Communications Board ("TECB") was created "for the purpose of assisting emergency communications district boards of directors in the area of management, operations, and accountability, and establishing emergency communications for all citizens of the state."[1] To those ends, the TECB is authorized to "administer the deployment of 911 service for emerging communications technologies, including but not limited to IP-enabled service, that are capable of connecting users dialing or entering the digits 911 to public safety answering points and other non-wireline services."[2] The TECB also has the statutory authority to establish technical operating standards for 911 service.[3]

To further its statutory purposes, the TECB has the power and authority to:

> Respond to requests from emergency communications districts, commercial mobile radio service (CMRS) providers or other parties and subject to availability of funds, review and approve requests for reimbursements for expenditures or payment of obligations incurred to implement, operate, maintain, or enhance statewide wireless enhanced 911 service in conformance with any rules or orders of the FCC, and other federal and state requirements that pertain to wireless enhanced 911 service.[4]

Pursuant to such authority, the TECB hereby establishes a reimbursement program with selected concomitant standards for NG911 controller and adjunct equipment. ECDs that receive, share, or contribute facilities, resources or income with other governmental entities will be required to submit a written interlocal agreement with such governmental entities memorializing their relationship. By accepting this funding, the ECD confirms that it is in compliance with, and will continue to comply with, all laws, rules, policies and orders applicable to the ECD and the provision of 911 or wireless enhanced 911 service.

Reimbursement may be either retroactive or prospective, meaning that it is available for reimbursement of purchases of approved, eligible equipment and for advance payments for proposed or planned purchases of approved, eligible equipment. The process is the same except where differences are described in the following procedures.

[1] Tenn. Code Ann. § 7-86-302(a).
[2] Tenn. Code Ann. § 7-86-306(a)(8).
[3] Tenn. Code Ann. § 7-86-306(a)(9)
[4] Tenn. Code Ann. § 7-86-306(a)(11).

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NG911 CONTROLLER AND/OR ADJUNCT EQUIPMENT
*Effective February 18, 2010*

## PROGRAM GUIDELINES APPROVED BY TECB

On February 18, 2010, the TECB voted to provide funding up to a maximum allowable amount for each ECD for the purchase and installation of controllers and/or adjunct equipment needed to implement the Next Generation 911 (NG911) project. The objective as stated at the meeting and approved by the TECB is to provide ECDs with sufficient funds to connect to the NG911 Internet Protocol (IP) infrastructure – for most ECDs that involves purchase an IP capable controller or other related IP components. Basic components of the reimbursement program include:

1. Funding of $25 million, which was set aside in the budget for this purpose;
2. Reimbursement to ECDs that submit approved technical specifications for qualified equipment and other documents when appropriate;
3. Preapproval by TECB may be obtained to assure that reimbursement is available for the purchase of specific equipment;
4. Pre-payment by TECB for specific equipment anticipated costs is available, upon request and under justifiable circumstances;
5. In order to provide ECDs with maximum autonomy to satisfy their individual needs, ECDs are responsible for bidding, vendor and equipment selection, and implementing the funding processes available under this program. Funding is subject to TECB review and approval of equipment;
6. In situations where ECDs choose to lease qualifying equipment, lease payment costs are eligible for reimbursement;
7. Reimbursements will be retroactive for eligible costs incurred since January 1, 2008;
8. Each ECD has an allocation based on two components: (1) a base amount of $120,000 plus (2) an additional amount based on the proportion of population in the ECD compared to that of the State;
9. The remaining balances in the $40,000 controller funding program and the $50,000 GIS mapping system reimbursement program (active since July 1, 2003) are closed out and the leftover funding in each ECD's account is rolled into each individual ECD's NG911 available allocation.

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## PROCEDURES

**Written Request:** Submit a written request to initiate action. Instructions for the required contents of the written request are provided below in items **1** through **6**. Requests may be:

- Mailed to: TECB; Davy Crockett Tower, 500 James Robertson Parkway; Nashville, TN 37243-0582;
- Hand delivered to: TECB; Davy Crockett Tower, 11th Floor; 500 James Robertson Parkway; Nashville, TN 37243-0582;
- E-mailed to: TN.ECB@tn.gov **or** James.Barnes@tn.gov;
- Faxed to: (615) 401-7642 (preferred) **or** (615) 253-2180.

The following is a checklist of information that must be submitted or available for TECB review. You do not have to submit a copy of this checklist.

- ☐ a cover letter providing necessary information
- ☐ detailed equipment specifications, if not already provided
- ☐ vendor or TECB certification that required minimum standards are met
- ☐ a copy of vendor invoice(s), or proposed costs
- ☐ indication of payment, or commitment to purchase, if requesting funds
- ☐ a completed and signed request for reimbursement form, if applicable

**1. Cover Letter:** Requests should include a cover letter from an authorized representative of the ECD. Each cover letter should include the following:

A. A statement identifying the purpose of the request. The purpose should be for one of the following reasons:
   - Retroactive reimbursement; **or**
   - Pre-approval of the commitment of TECB to reimburse the cost (This is primarily to assure your Board that you have the financial support of the TECB before you enter into any purchasing commitments); **or**

- Prospective advance payment. This option must include information and documentation of your Board's commitment to purchase. Documentation may include copies of Board minutes authorizing the purchase, purchase orders, signed contract agreements with vendors, etc.;

B. The address of the physical location of the equipment, so that it may be examined, if needed;

C. A description of the relationship between your ECD and any other local governmental entity that shares, jointly uses, contributes, or obtains any facilities, equipment, resources or income of any kind with the ECD to enable TECB to determine if an Interlocal Agreement is required;

D. In the event that you are unable to obtain the vendor's signature on the Vendor Certification form, you must explain the circumstances and request that the TECB Director of Technical Services review and approve the Vendor Certification, if appropriate;

E. Any special requests or circumstances.

**2. <u>Equipment Identification</u>:** The request must include a detailed description of the equipment sufficient to enable technical evaluation of the suitability and function adequacy (include specifications, if available). Because the purpose of this funding program is to connect PSAPs to the NG911 infrastructure, an NG911 controller must be on hand and IP platform connections must be installed sufficiently to make the controller IP capable by the relative "flip of a switch" before consideration will be given to processing any reimbursement for adjunct equipment.

**3. <u>Vendor Certification:</u>** Provide documentation from the vendor or TECB staff that minimum standards established are met or exceeded for identified equipment by:

- Having the vendor sign a certification form (Attachment B) and attach the form to your request for reimbursement.
- If the vendor does not sign the certification form, include an explanation in the cover letter (described in Section 1 above) as to the reasons the vendor could not comply, and request the Director of Technical Services of TECB to certify the equipment. This will be scheduled by the Director of Technical Services as time permits.

**4. <u>Cost:</u>** Obtain and attach an invoice or invoices from the vendor to identify and verify cost. (If pre-approval or prospective procedures are being followed, attach price or cost data from proposals, bids, etc.)

5. **Proof of Payment:** Provide documentation that the invoice or invoices have been paid and include with the reimbursement request. This documentation may include copies of cancelled checks, copies of accounting ledgers listing the applicable payment, or check numbers and dates written on the face of the invoice or invoices.
    - If you are requesting pre-approval and not requesting funds, proof of payment will be required when you request reimbursement.
    - If you are requesting prospective advance payment for funds before you make payment to the vendor, proof of payment is not required. Instead, documentation of your commitment to pay a specific amount (described in Section 1 above) is required.

6. **Request Reimbursement:** Submit to TECB a completed "*Request For Reimbursement*" form (a copy is provided as Attachment A).

**Additional Information:**

**Approval Notice For Prospective Purchase Request:** If you are requesting pre-approval and not requesting funds, the TECB shall provide a conditional notification of approval and describe the conditions under which the approval is granted. If a request for approval is rejected, the TECB shall provide justification for the rejection and the ECD may appeal the rejection to the TECB members at the next scheduled TECB meeting. Requests for appeal must be received no later than two (2) weeks prior to a TECB meeting to be placed on the agenda for consideration during that meeting.

**Inventory:** If applicable, the ECD shall record the item as an asset and maintain inventory records for audit purposes.

**Monitoring:** TECB staff or your auditor may monitor compliance with this program. Any discrepancies found may result in a request for repayment of inappropriately used funds, reductions in future funding to the extent of inappropriately used funds, or other action deemed authorized and appropriate. Monitoring efforts may include, but not be limited to:

- Checking equipment specifications,
- Matching invoices with equipment,
- Verifying location and use of equipment.

**ATTACHMENT A**

# STATE OF TENNESSEE
# TENNESSEE EMERGENCY COMMUNICATIONS BOARD
# DEPARTMENT OF COMMERCE AND INSURANCE

**500 JAMES ROBERTSON PARKWAY,**
**NASHVILLE, TENNESSEE 37243-0582**
**615-253-2164/FAX: 615-401-7642**

## REQUEST FOR REIMBURSEMENT
## NG-911 CONTROLLER AND/OR ADJUNCT EQUIPMENT

**District:**

**Contact:**

**Address:**

**Brief Description of Equipment Purchased: (Attach detailed information.)**

As an authorized representative of the Emergency Communications District named above, the undersigned submits this request for reimbursement costs for purchases of a NG-911 capable controller and/or adjunct equipment.

| Description | Total Request |
|---|---|
| **Reimbursement Request** | **$** |

## CERTIFICATION

I hereby certify that this request for cost reimbursement, which is submitted to the Tennessee Emergency Communications Board pursuant to Tenn. Code Ann. § 7-86-306(a)(11), is correct and valid. I further certify that the amount claimed was expended to implement, operate, maintain or enhance statewide wireless enhanced 911 service in conformance with all applicable orders and rules of the Federal Communications Commission and other federal and state requirements that pertain to wireless enhanced 911 service.

I further certify that if the ECD shares, jointly uses, contributes, or obtains any facilities, equipment, resources or income of any kind with another governmental entity, then an interlocal agreement memorializing the terms of joint operation as contemplated by Tenn. Code Ann. § 7-86-105(b)(7) exists between the parties and is attached to this document, or on file in TECB offices and available for review.

Signature of District Agent Date

# Tennessee Emergency Communications Board
# Reimbursement Requirements For
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*Effective January 30, 2012*

ATTACHMENT B

## VENDOR CERTIFICATION

I hereby certify that the NG911 controller and/or adjunct equipment identified on the attached documentation and/or associated invoice meets the i3 standards adopted by the National Emergency Number Association, as reflected in the attached technical specifications (Attachment C). I also certify that the licensing necessary for the NG911 controller to meet i3 requirements as implemented in the Tennessee 9-1-1 over NetTN network is included in this transaction.

_____

Vendor or Company Name

_____ _____

Authorized Vendor Representative Signature Date

**ATTACHMENT C**

***NOTE: This document was converted from a .pdf file to a Microsoft word file and may have some differences from the original document. For the original document go to the TECB website at www.tn.gov/commerce/911 and look under District Resources/Funding and Reimbursement Programs/NG911 Controller and/or Adjunct Equipment.***

# 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)'.

AT&T reserves the right to revise this document for any reason, including but not limited to, conformity with standards promulgated by various governmental or regulatory agencies, utilization of advances in the state of the technical arts, or the reflection of changes in the design of any equipment, techniques, or procedures described or referred to herein. Liability to anyone arising out of use or reliance upon any information set forth herein is expressly disclaimed, and no representations or warranties, expressed or implied, are made with respect to the accuracy or utility of any information set forth herein.

This document is not to be construed as a suggestion to any manufacturer to modify or change any of its products, nor does this document represent any commitment by AT&T to purchase any product
whether or not it provides the described characteristics.

Nothing contained herein shall be construed as conferring by implication, estoppel or otherwise, any license or right under any patent, whether or not the use of any information herein necessarily employs an invention of any existing or later issued patent.

## 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).

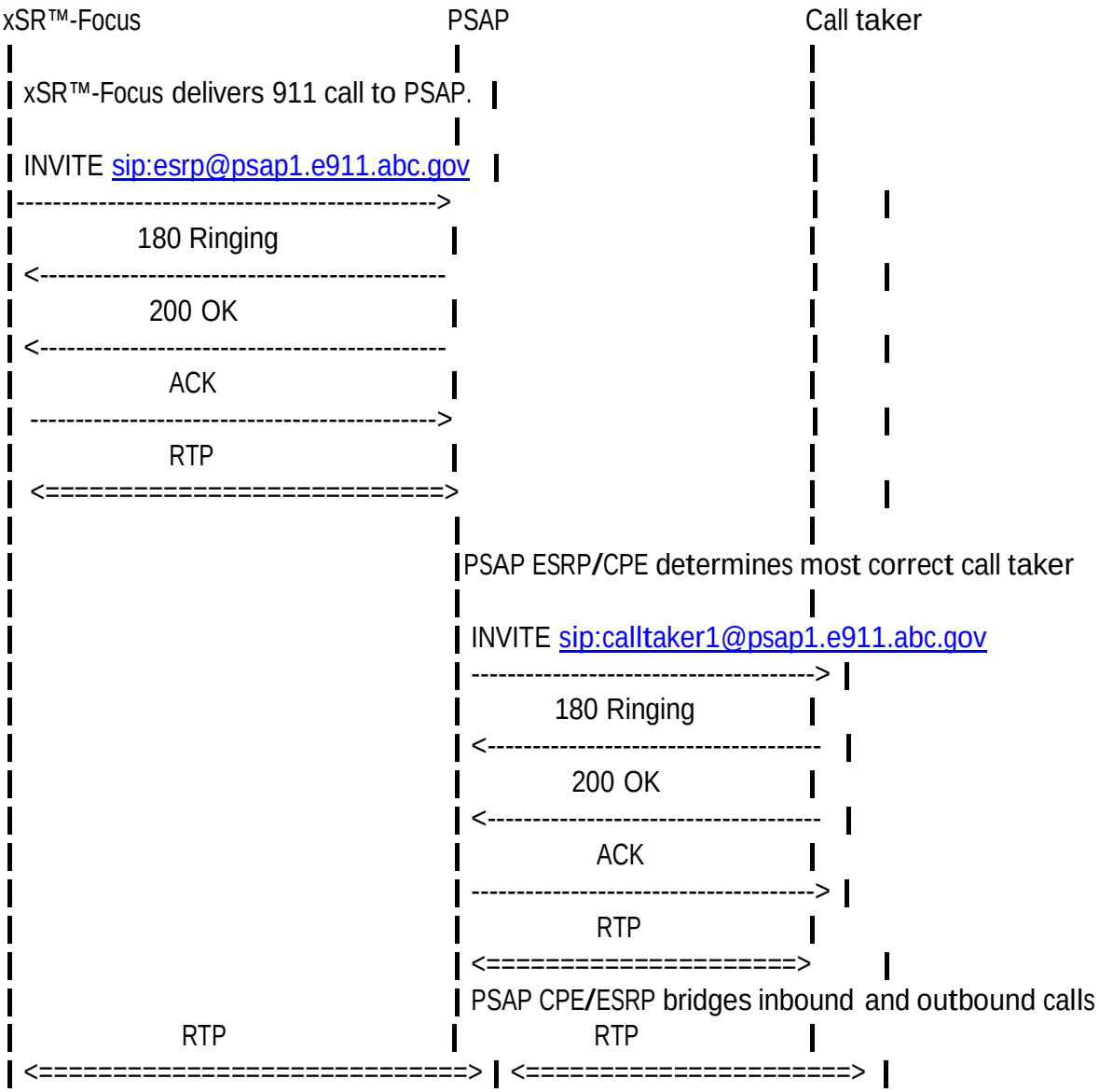
Here's an example PIDF-LO format created by xSR™ (complies with RFC 5139):

```
<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>
```

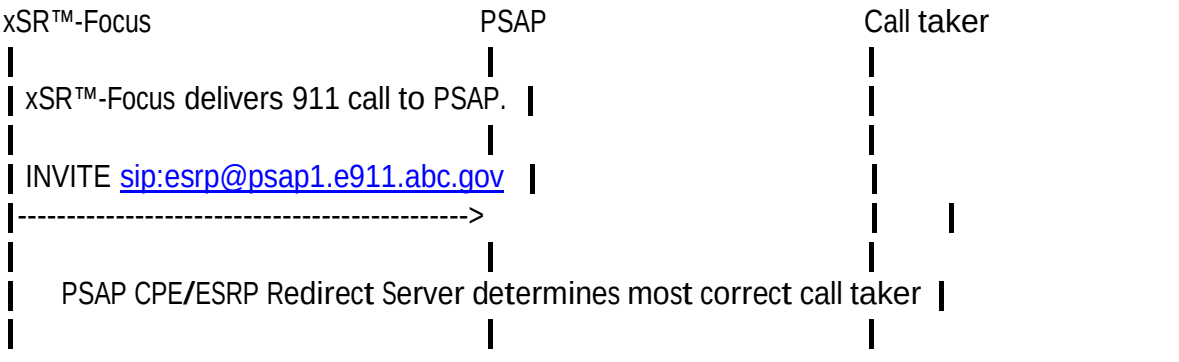
The PIDF-LO passed within the SIP INVITE to PSAPs can be provided by value (LbyV) or by reference (LbyR). For all ECRIT (IP) originated 911 calls, which format is provided depends entirely on the PIDF-LO originally sent by the 911 caller. xSR™ will always dereference a PIDF-LO received by reference and 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.

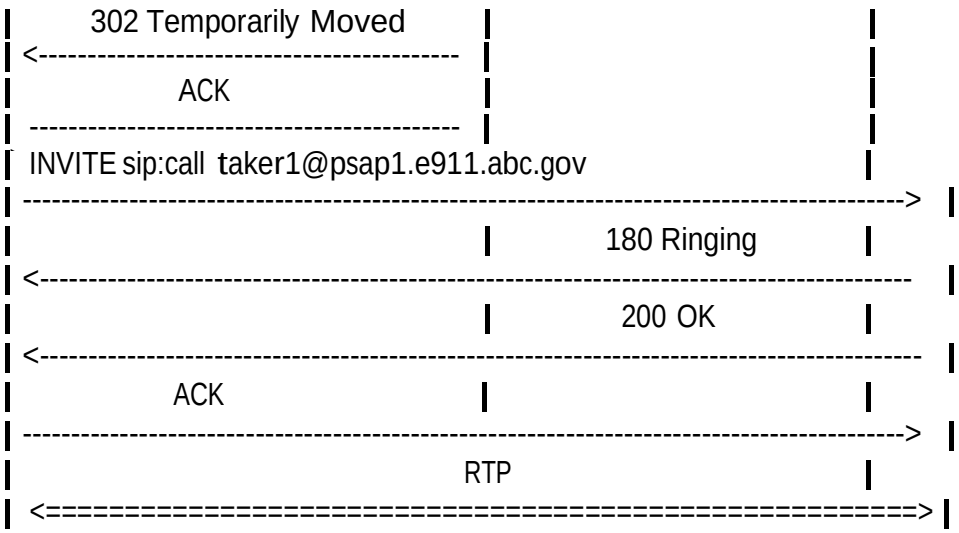
Since ultimately a call taker must answer all 911 calls originating from the xSR™, it is the responsibility of the PSAP CPE to deliver inbound calls from the xSR™ to the correct call taker. The xSR™ will simply deliver SIP calls to the predefined SIP URI of the PSAP. The PSAP SIP endpoint may answer 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

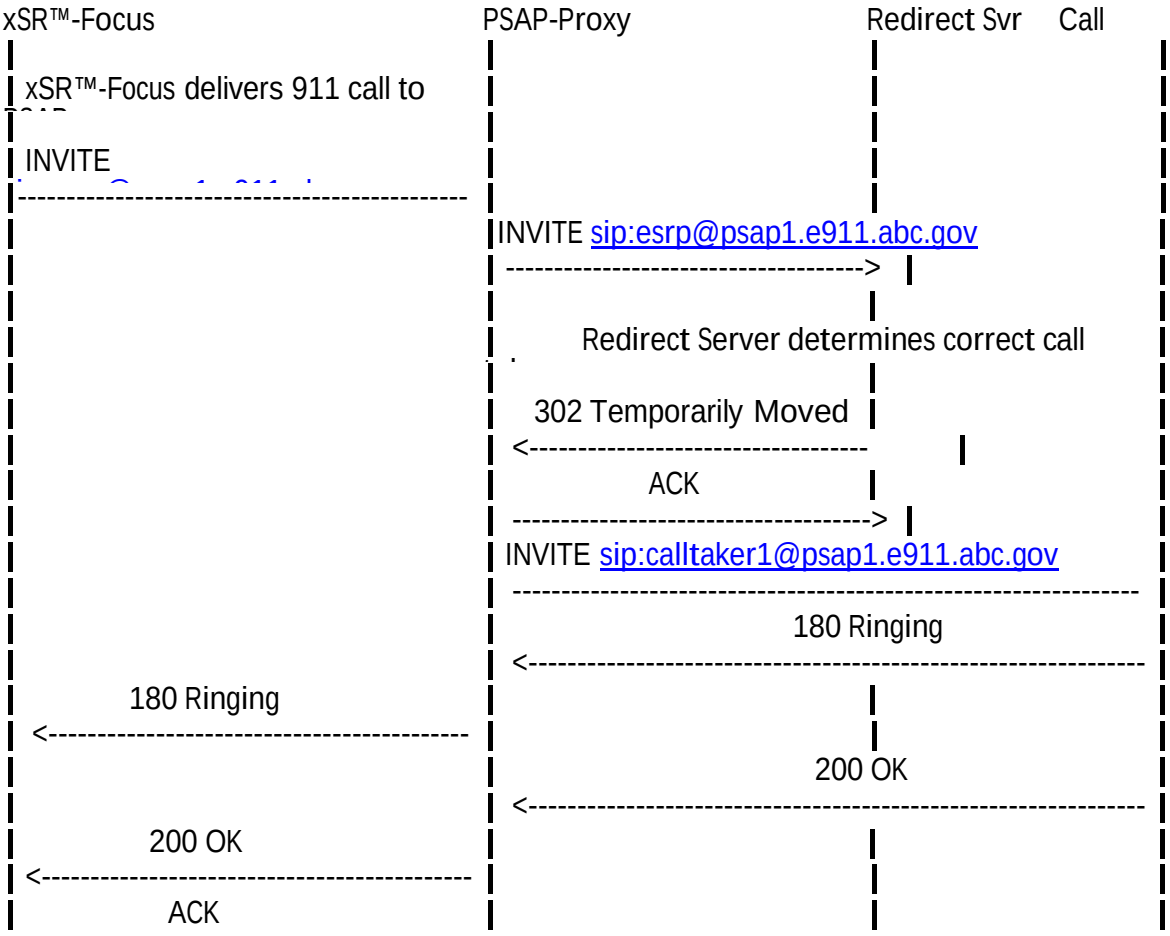


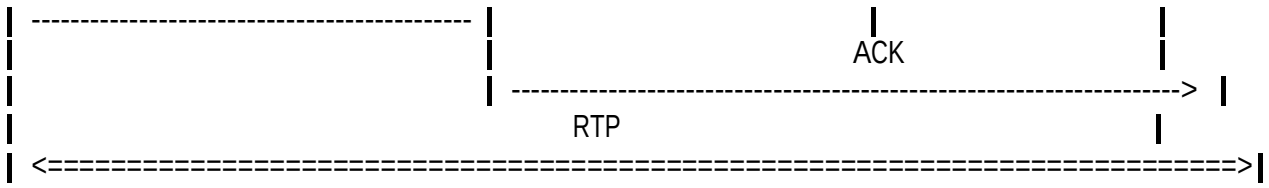

### 1ST Party Call Control Style PSAP CPE - Call Flow



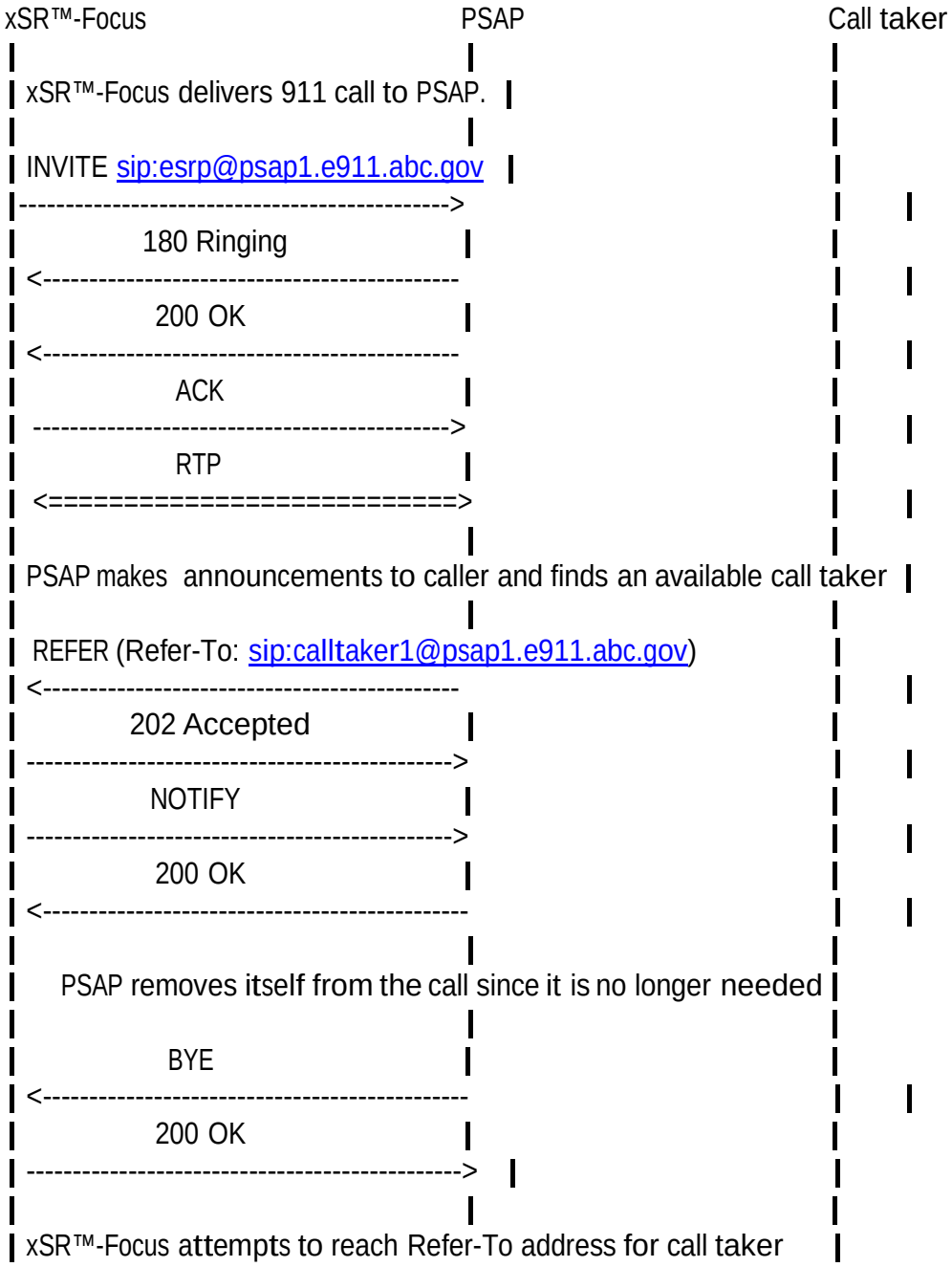


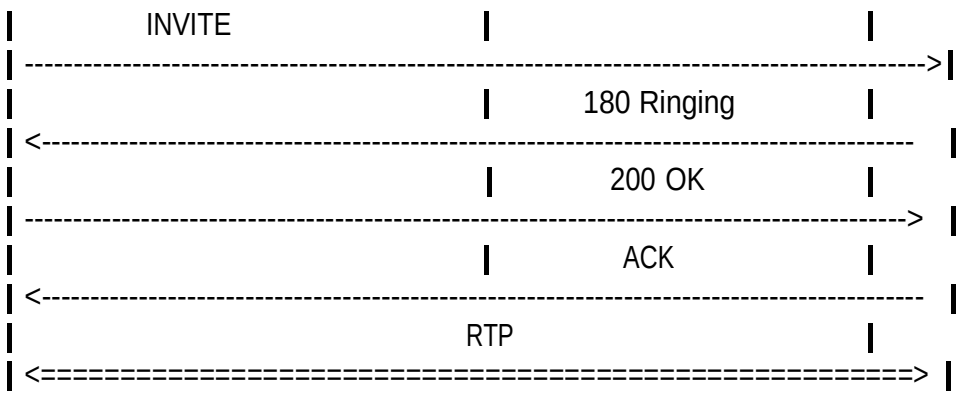

### 1st Party Call Control Style with Proxy - Call Flow






### 1ST Party Call Control Style with REFER - Call Flow






## 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™.

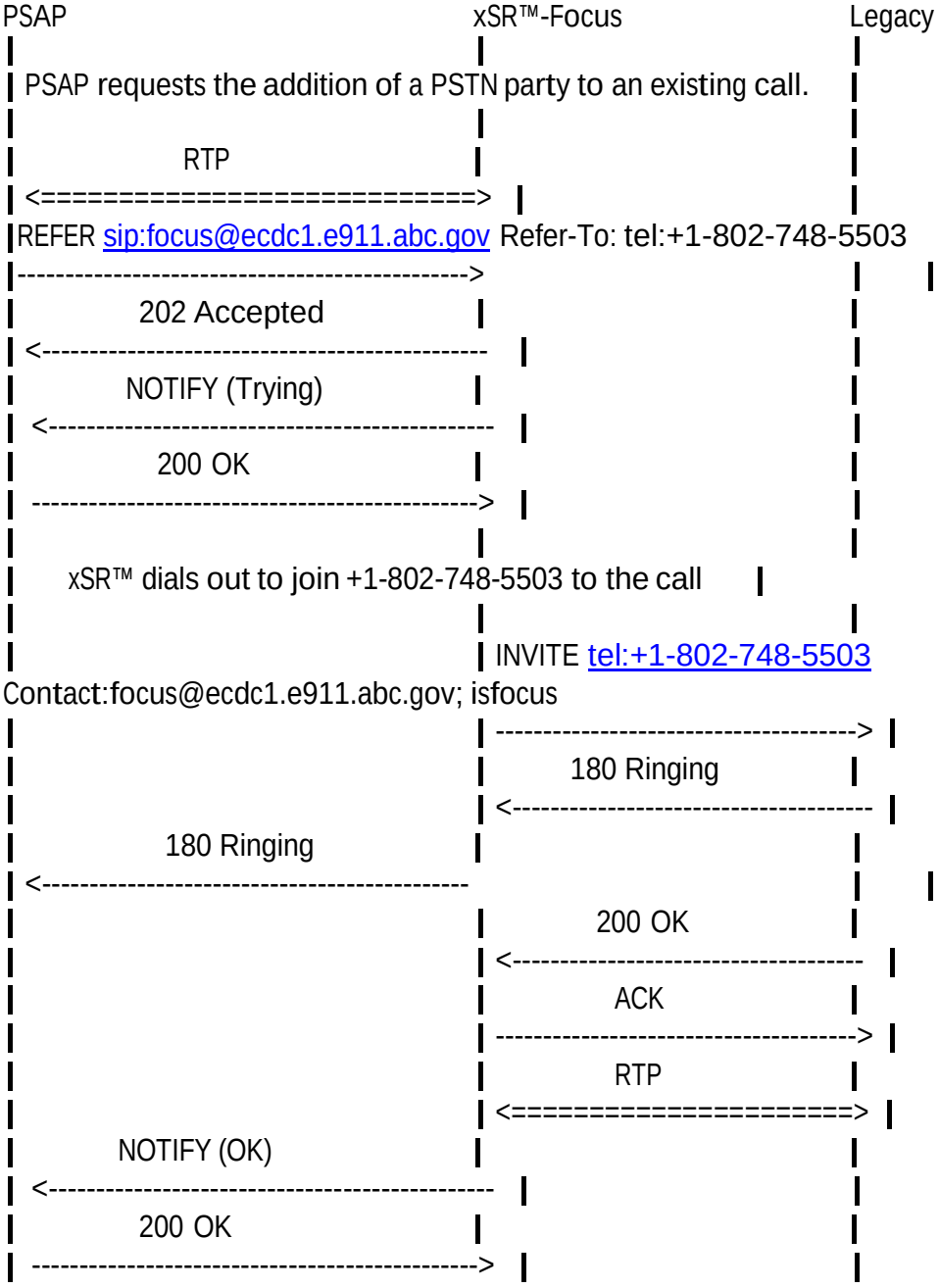
Focus Trunks are either Legacy PSTN trunks or SIP trunks terminated at the xSR™. These trunks are not directly accessible by PSAPs. They are, however, indirectly accessible for selective transfers. xSR™ is the endpoint for all selective transfer mixing. This guarantees an available outbound trunk and 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.

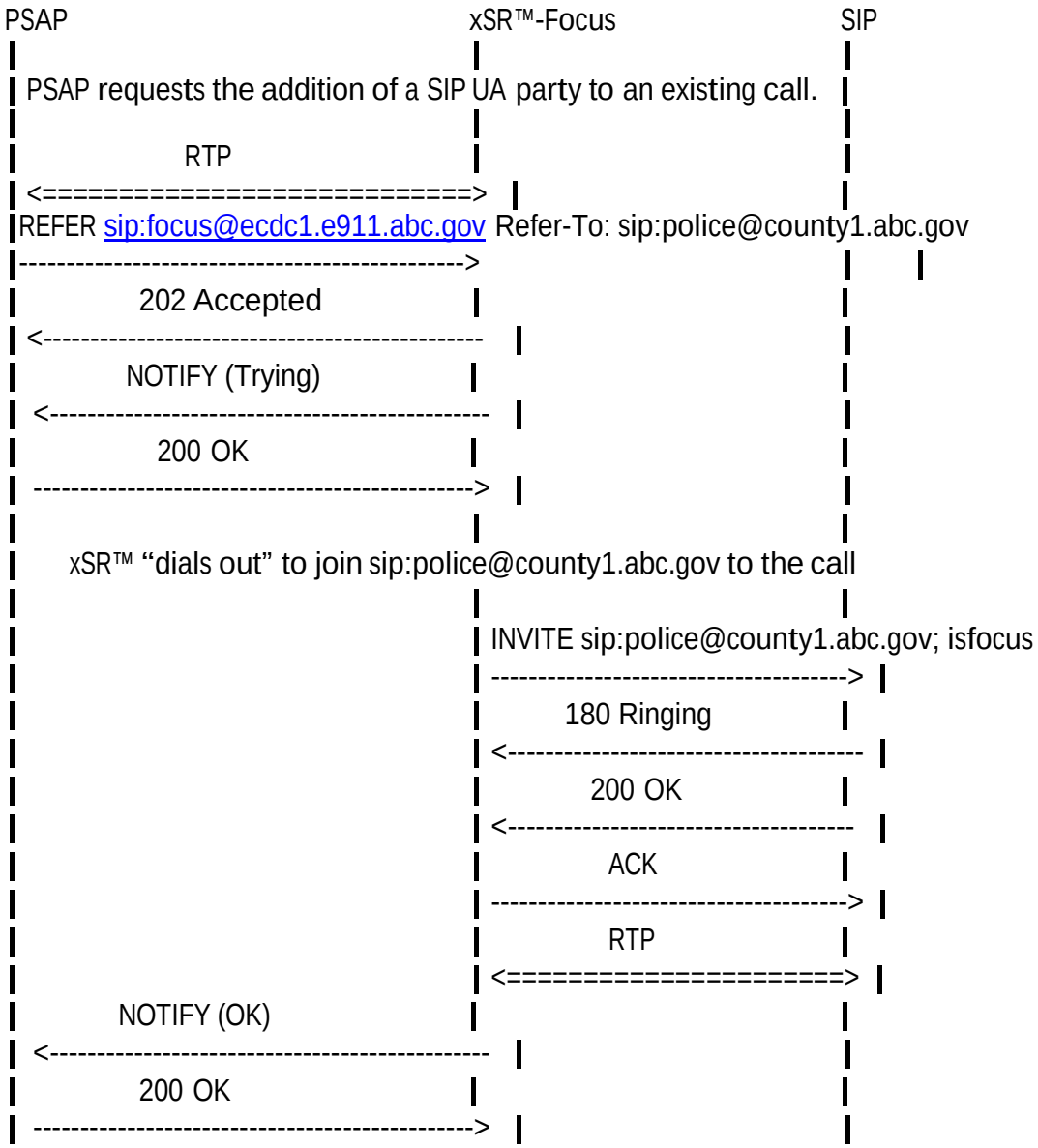
Focus Trunks can be added to existing 911 calls sent through the xSR™ through the SIP REFER request. RFC 4579 section 5.5 describes the procedure required to add a new resource to a conference. The

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

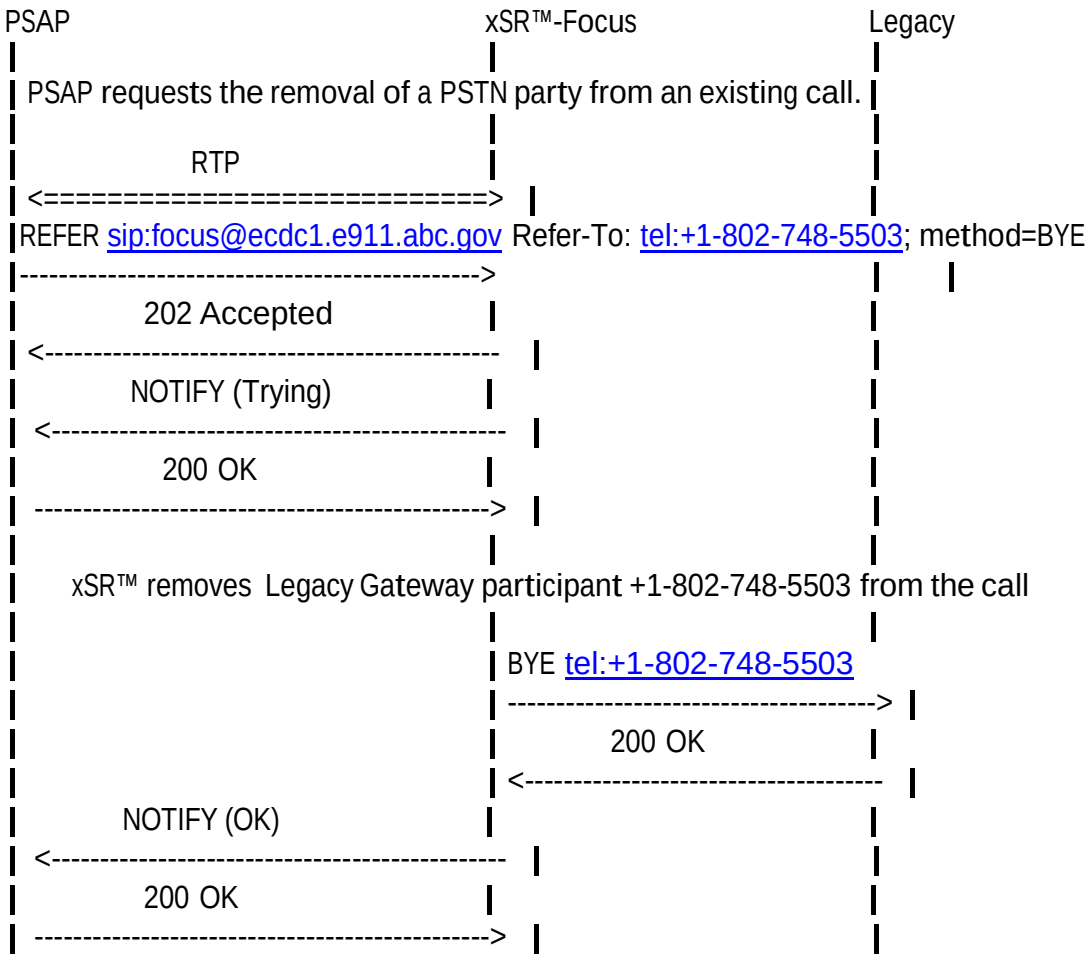


### Addition of SIP UA party – Call Flow




## 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:




## Knowing Call Information

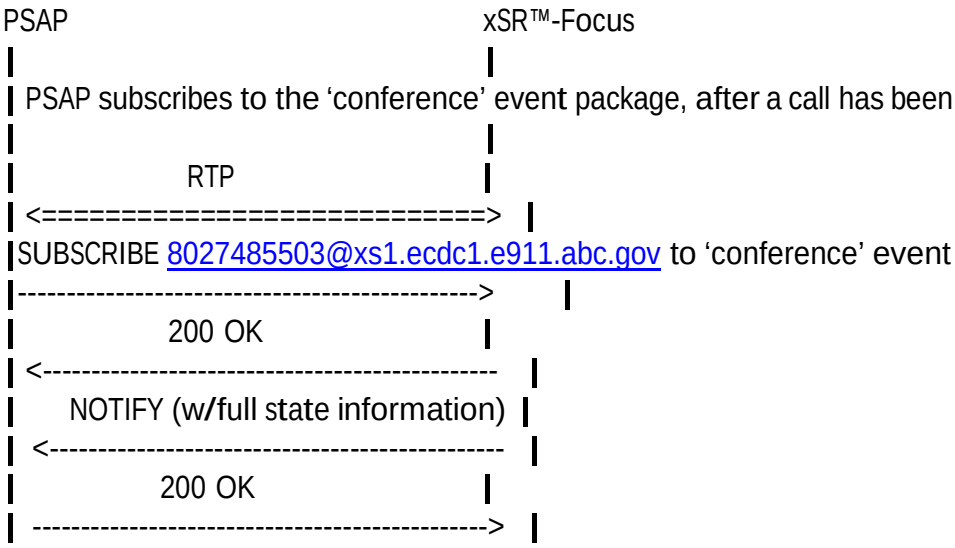
Call participants will change over the course of any call. Since xSR™ is a conference focus, it will be the first to recognize when participants have been successfully added or removed. As such, it is the most appropriate SIP UA to publish this call information. xSR™ has the ability to NOTIFY subscribers 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.




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

- 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."

Attachment D

## ADEQUATE EQUIPMENT INSURANCE CERTIFICATION

I hereby certify that, in accordance with Tennessee Emergency Communications Board Policy 40, the _____ Emergency Communications District will maintain comprehensive insurance on the equipment for which these funds are provided.

_____ _____

Director Date