# AT&T IP Flex Reach Configuration Guide

## IC 4.0 with ACME SBC

**Version 1.0**

**12/05/2012**

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# SIP Carriers

# 1 AT&T


## 1.1 Introduction

This document is intended for the setup and configuration of options for the Interactive Intelligence Interaction Center (IC) server and ACME SBC for use with the AT&T IP Flex Reach offering. This document covers setup and configuration relative only to the IC server and AT&T. Other configurations (stations, permissions, QoS, etc…) are outside the scope of this document. The interoperability test consisted of the following products:

| AT&T Service Tested | Status |
|---|---|
| AT&T IP Flex Reach | 100% |

| Product Tested | Version Tested |
|---|---|
| Interaction Center (xIC) | 4.0 SU2 |
| Interaction Media Server | 4.0 SU2 |
| IP Phones | |
| Polycom IP 430, 450 | 3.2.5.0643 |
| ACME SBC Net-Net 4250 | 5.0.0 Patch 13 |

## 1.2 Product Descriptions

**Interaction Center (xIC)** – Interaction Center delivers an innovative pre-integrated application suite to manage all business communications on one platform. xIC's powerful contact center applications and PBX / IP PBX call processing, voicemail, fax server and unified messaging capabilities extend its reach throughout the enterprise - connecting and empowering agents, supervisors and business users, to elevate productivity, performance and customer service.

**ACME SBC** – The ACME Session Border Controller is used to bridge the customer network to the AT&T IP network to support the services provided (IP Toll Free and Flex Reach). The Border Controller ensures that the session, consisting of SIP signaling and RTP, is converted, measured, and/or modified for calls to traverse both the customer and remote networks successfully.

**Interaction Media Server** – A standalone server used with a xIC integration that provides a dramatic increase in system scalability and reliability by processing the majority of all media RTP audio flows. The audio processing is offloaded from the xIC server to the dedicated media server or servers registered to the xIC servers.

# 2 Special Notes

**Transport Protocol Support**
UDP is the only supported transport protocol available with the AT&T IP Flex Reach service. The default protocol for IC 4.0 is UDP.

**T.38 Faxing –** Requires that the call originate as a G.729 call. An inbound G.711 fax cannot be re-invited to T.38.

Interactive Intelligence does not support T.38 SG3 faxing at the time this document was created. It does however, support G3 faxing.

**Multiple SIP Trunk Connections**
If remote sites have independent AT&T IP Flex Reach service, refer to section 5 for specific requirements.

**xIC Version**
AT&T IP Flex Reach was certified using Interactive Intelligence IC server 4.0. This integration requires a minimum of SU2; which includes all resolutions uncovered during testing.

**Remote Priority Codecs**
The IC server does not accept remote priority codecs. The codec that will be selected for use is the highest one in the IC server list that matches one offered by the carrier. Please put the carrier desired codec at the top of the list to avoid any issues.

**G.729 Annex B Support**
The IC server can use G.729 with the Annex B option, however it is not dynamically configurable per call. It is enabled on a given line (as demonstrated in the appropriate configuration section below), and is only an always on, or always off option.

**G.726 Support**
The IC Server does not support the G.726 codec that is sent by ATT.

**Failover Support**
The ACME Packet SBC is required for redundant IC server support. The SBC maintains the route availability to both the IC server pair as well as any number of IP Border Elements provided by AT&T.

**IP Flex Reach Phone Numbers**
A customer may receive one of 2 types of DIDs from AT&T: Virtual TNs and non-virtual TNs.

A Virtual TN is one that has an NPA that is different from the NPA at the customer site to which it is being routed. For a virtual TN, AT&T will pass 10 digits to the PBX. For example, if a PBX telephone is associated with a VTN, the number received from AT&T would be 10 digits (i.e. 732-216-2700).

A non-virtual TN has an NPA that is the same NPA as the customer site. For a non virtual TN, AT&T will pass the length of the phone extension plus some prefix if needed (typically a 4 digit extension without a prefix). If a PBX telephone is associated with a non virtual TN, the number received from AT&T would be 4 digits (i.e. 2701 for a 908-216-2701 TN).

However, when originating calls to AT&T, the calling party number must be a 10 digit number regardless of the type of TN associated with the phone that is originating the call. On the PBX, a specific 10 digit AT&T TN will always be used as the calling number for PBX to AT&T calls.

**Emergency 911/E911 Services Limitations**

While AT&T IP Flex Reach Services support E911/911 calling capabilities in certain circumstances, there are significant limitations on how these capabilities are delivered. Please review the AT&T IP Flexible Reach Service Guide in detail to understand these limitations and restrictions.

# 3 Overview

## 3.1 Network Diagram

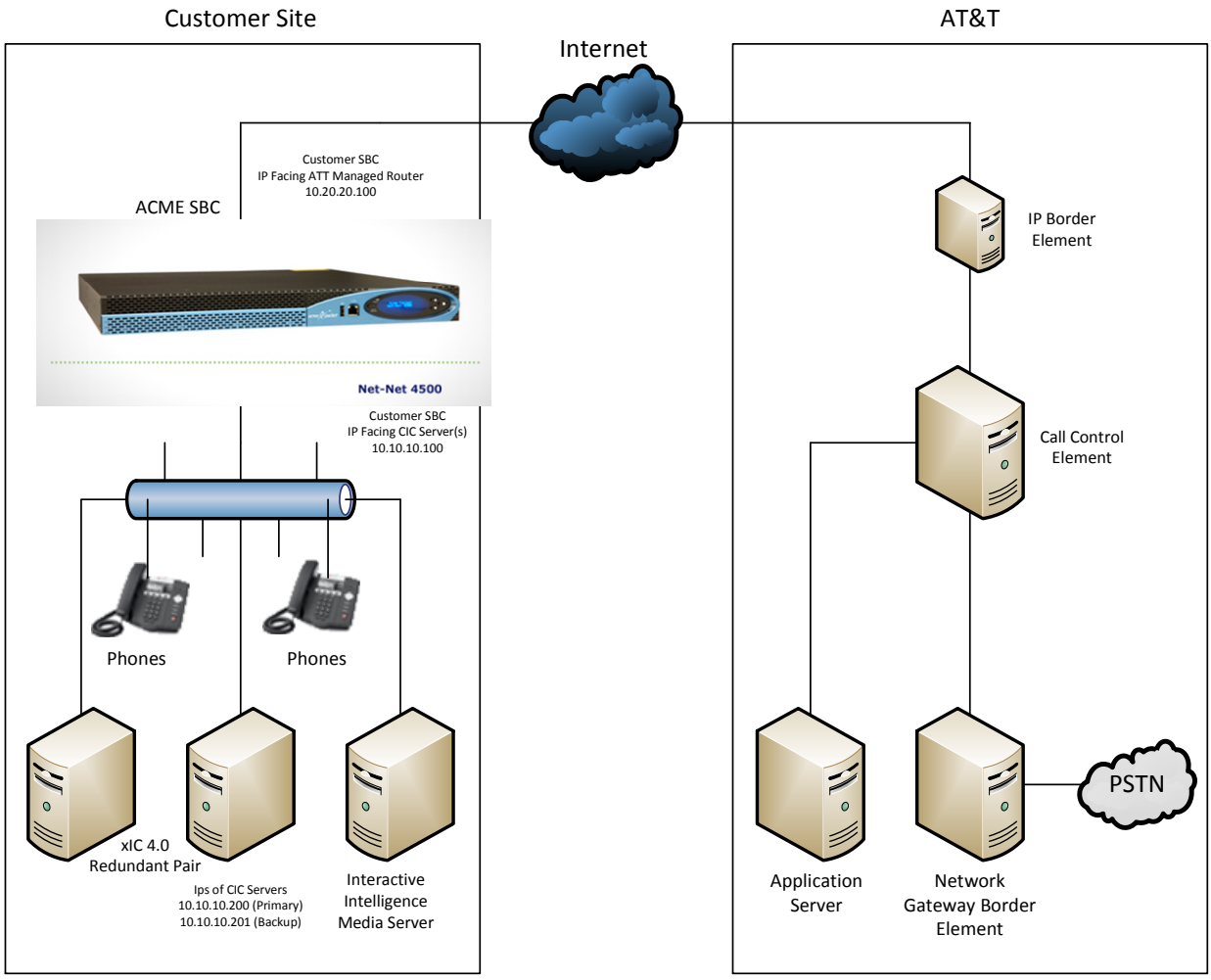


**Figure 1: Diagram of General Network Setup**

## 3.2 Proxy Description Overview

### 3.2.1 Customer Site

**ACME SBC:** Configuration of the ACME SBC will be in a separate configuration guide provided by ACME Packet.

**IC Server(s):** The outbound proxy of the AT&T SIP line will be the ACME Packet SBC.

**Refer to *section 4.1.3.4* for the dialog menu.**

**IP Border Element:** The destination Proxy address of the IP Border Element will be directed to the ACME Packet SBC IP address.

### 3.2.2 AT&T Network

AT&T will request the IP address of the customer's border element interface connected to the AT&T Managed Router. This is the ACME Packet SBC shown in **Figure 1** above.

# 4 IC Configuration Guide

## 4.1 Line Configuration

The line page has a vast majority of the configuration options required for SIP carrier setup. This is the section that configures the connection to the carrier's servers, any authentication or registration information, and basic configuration needs.

As stated before, two lines must be created. These lines are required, one for the AT&T connection and one for the stations. Each portion of the lines page will be explained as it relates to the AT&T Service. For this document, the AT&T connection line will be referred to as *AT&T SIP Line*, and the station line will be referred to as *stations*. Also, any reference to a menu, while talking about the line configuration, will refer to the options on the left side of the line configuration page, and tabs will refer to the standard tab interface across the top of the line configuration page.

### 4.1.1 Line Menu

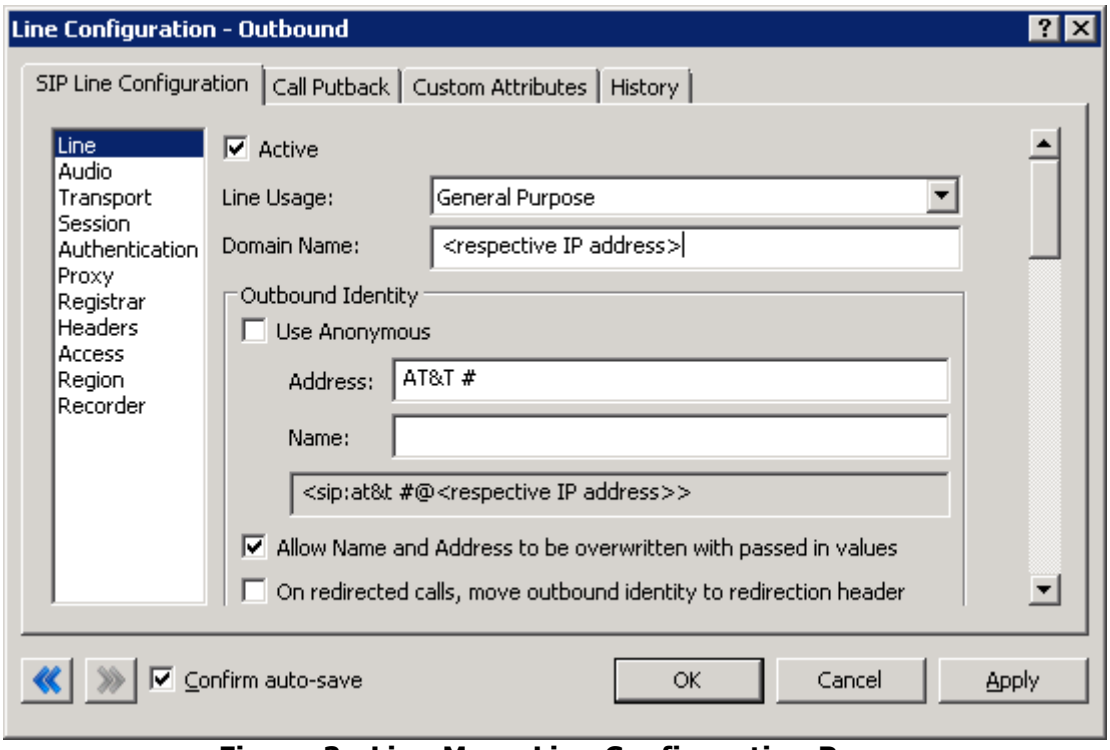


**Figure 2: Line Menu Line Configuration Page**

#### 4.1.1.1 Active

The active box should be checked. This activates the line. If this box is not checked, the line will not be available for any function. This can also be affected by right clicking on the line in Interaction Administrator, dropping to the *Set Active* menu option, and selecting *Yes*.

#### 4.1.1.2 Domain Name

This box should contain the ACME Packet SBC IP address.

#### 4.1.1.3 Address

The phone number provided by the SIP carrier should be entered into this box. The number entered is used in the "From" header in outbound SIP calls. This is critical to be accurate as AT&T validation is done based on this number. Incorrect numbers can lead to some functionality (e.g. international calling, etc…) not working as expected or not working at all.

#### 4.1.1.4 Enable T.38 Faxing

AT&T's SIP carrier service supports the T.38 faxing protocol by default. Uncheck this option if you do not have (or wish to use) an analog to SIP capable FXS type device to connect an analog fax machine to the system.

#### 4.1.1.5 Remainder of Line Menu Options

These have no major direct impact on the SIP carrier configuration, and should be addressed according to business needs.

### 4.1.2 Audio Menu

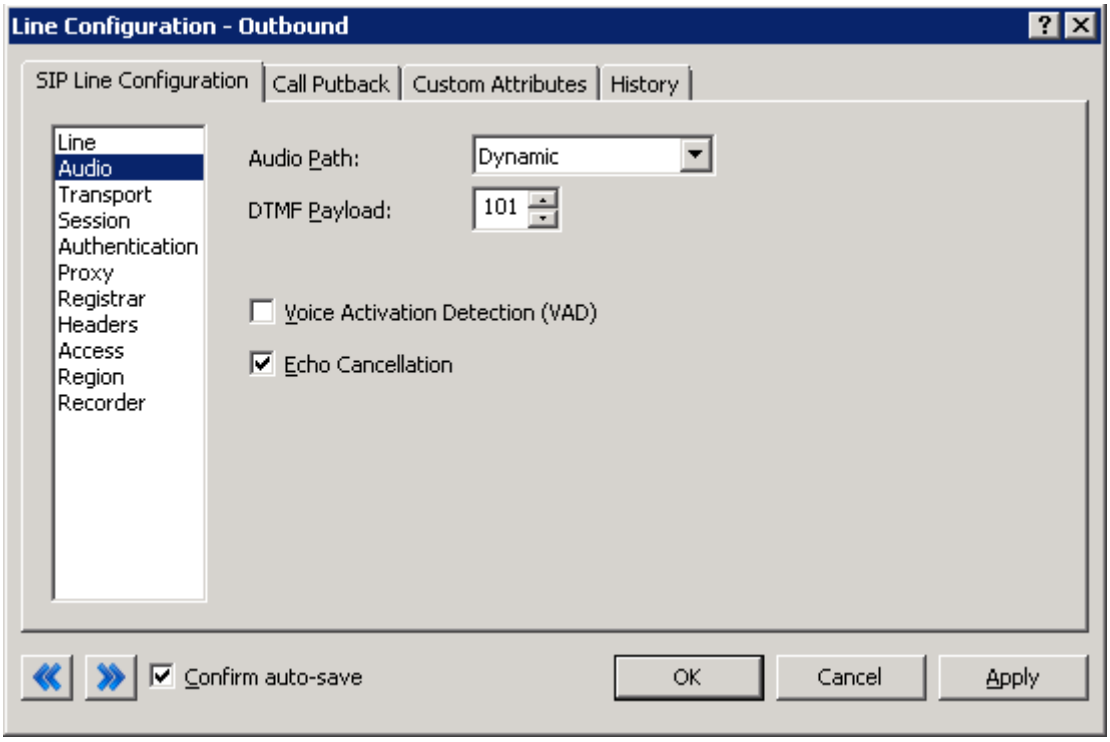


**Figure 3: Audio Menu Line Configuration Page for IP Flex Reach**

#### 4.1.2.1 Audio Path

This is for the most part, the choice of the client with respect to the business being done on the server. However, there are **several important caveats**.

1. *Dynamic* audio for SIP carriers has significantly less delay as compared to *Always In* audio (~100ms).
2. The audio will be brought into the IC server when set to *Dynamic Audio* for any call that is recorded (just for that call, not permanently). If using a Media Server recorded calls will not travel through the IC server.

#### 4.1.2.2 DTMF Payload

To use DTMF options AT&T and the IC server must negotiate the same payload. IP Flex Reach uses a Payload type of 100 by default .

Normal Media should be configured on the *station* connection Line Configuration to avoid any compatibility issues with DTMF negotiation. See media timing in Figure 5, Session Menu Line Configuration settings.

### 4.1.2.3 Voice Activation Detection (VAD)

This checkbox controls the Annex B option when using G.729. The IC server will *not* dynamically negotiate G.729 with annexb=yes. If Annex B is desired, this box must be checked, otherwise it will always use the annexb=no option. If it is required to have both, another line can be set up with some differentiating factor, one with Annex B enabled, and one without, then use the difference to select between the two. The reseller or an Interactive Intelligence support option can give more information on how this can be configured for the desired result.

### 4.1.2.4 Remainder of Audio Menu Options

These have no major direct impact on the SIP carrier configuration, and should be addressed according to business needs.

## Transport Menu

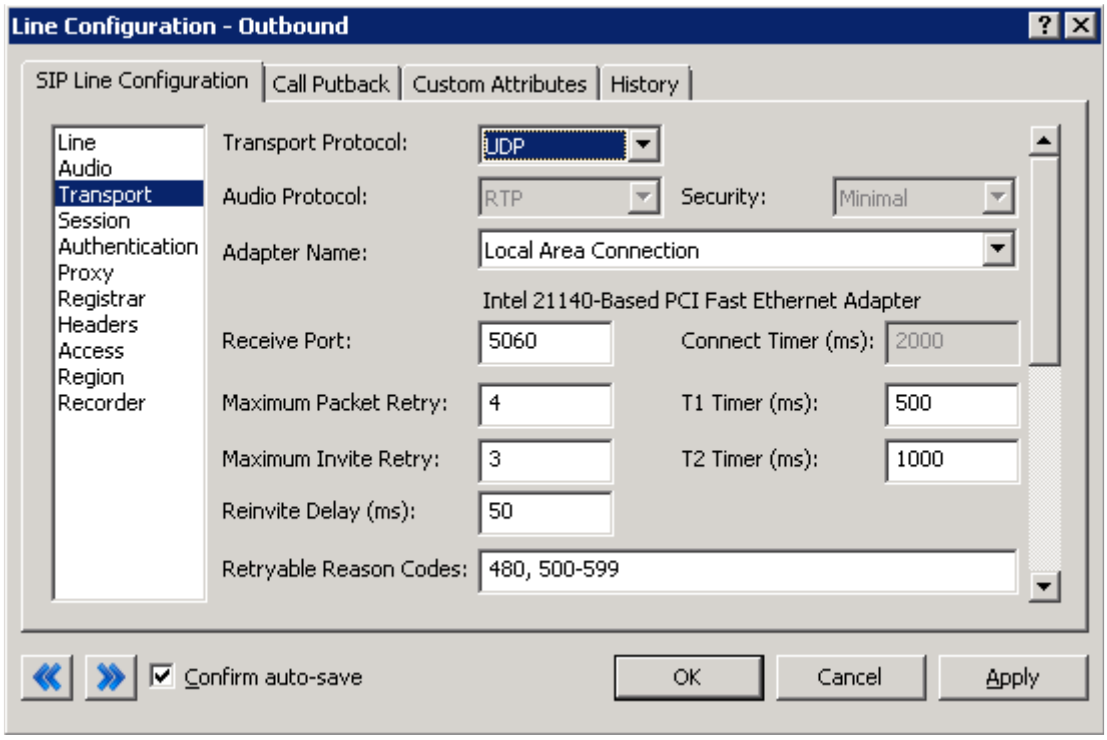


**Figure 4: Transport Menu Line Configuration Page**

### 4.1.2.5 Transport Protocol

This option should be set to UDP. As of March 25, 2009 UDP is the only available protocol. TCP and TLS are not currently supported.

### 4.1.2.6 Receive Port

This option should be set to 5060 (the standard SIP port), unless an agreement for an alternative port has been agreed upon with the AT&T.

### 4.1.2.7 Remainder of Transport Menu Options

These have no major direct impact on the SIP carrier configuration, and should be addressed according to business needs.

## 4.1.3 Session Menu

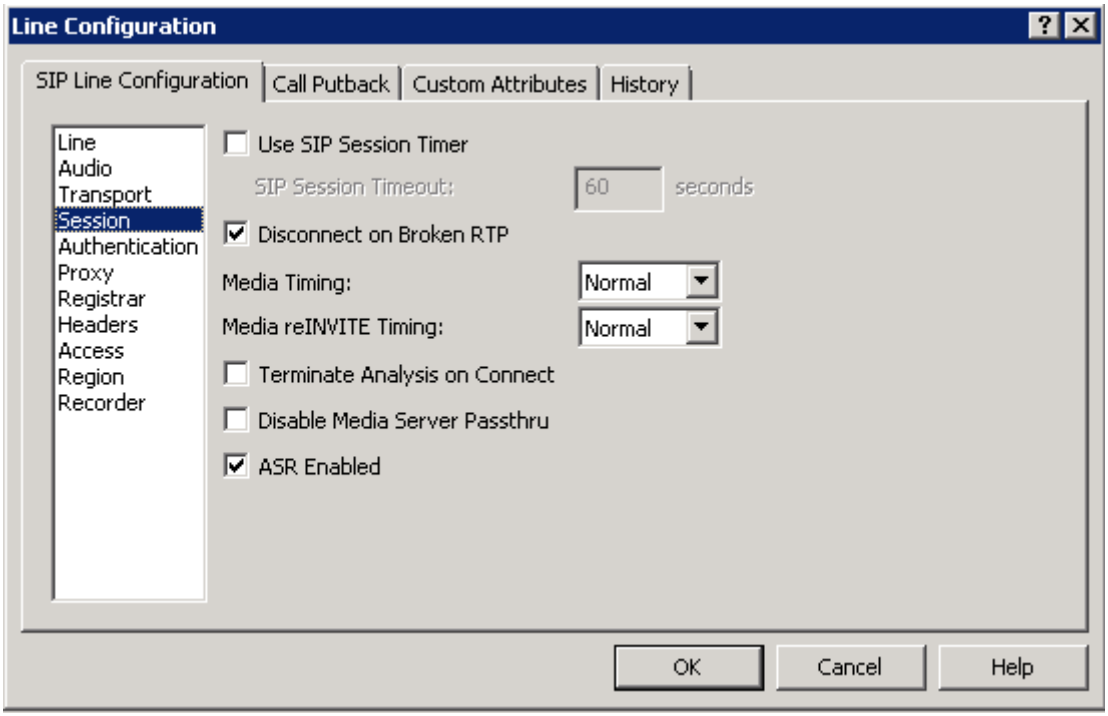


**Figure 5: Session Menu Line Configuration Page**

### 4.1.3.1 Media Timing

This must be set to normal to allow RFC2833 DTMF tones to work, as stated above (**Error! Reference source not found.** DTMF Type). Setting this to normal is the recommend method by Interactive Intelligence for all SIP Carriers, and is required for the AT&T service to function properly.

### 4.1.3.2 Remainder of Session Menu Options

These have no major direct impact on the SIP carrier configuration, and should be addressed according to business needs.

### 4.1.3.3 Authentication Menu

This box must be checked to enable authentication to the SIP Carrier. At the moment, AT&T uses a static IP model with no authentication, so nothing should be done with this page. However, were they to require authentication, the *User Name* and *Password* fields should be filled out with the appropriate information provided by the SIP Carrier.

### 4.1.3.4 Proxy Menu

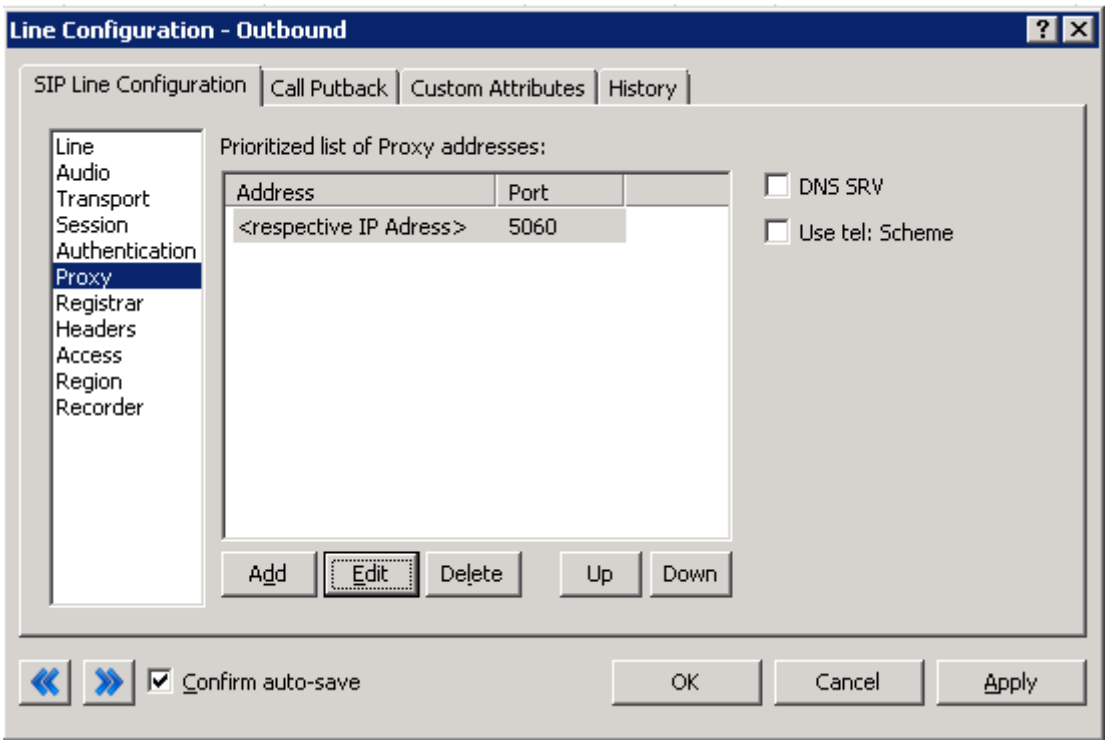


**Figure 6: Proxy Menu Line Configuration Page**

### 4.1.3.5 Prioritized list of Proxy IP addresses

In the case of AT&T, the proxy entry should be configured to send information directly to the ACME Packet SBC.

#### 4.1.3.6 Remainder of Proxy Menu Options

These have no major direct impact on the SIP carrier configuration, and should be addressed according to business needs.

### 4.1.4 Registrar Menu

#### 4.1.4.1 External Phone Numbers

This box is not currently used by AT&T's configuration. In most cases it would be used to register multiple numbers to the same IC server. However as AT&T uses a static IP method, they do the registration/routing setup on their end and do not require the IC server to request the various numbers itself.

#### 4.1.4.2 Prioritized list of Registrar IP addresses

This box is not used in AT&T's current configuration. The current system of providing a static IP or FQDN makes registration messages unnecessary.

### 4.1.5 Access Menu (Access Control lists)

If your business needs require that your endpoints (i.e. phones) use port 5060, Access Control lists are recommended. The 3.0 and higher versions of the IC server come with default station lines that are set to 8060. If using these default station lines for your endpoints, and not requiring multiple lines that are using the same protocol and port, this section can be skipped. These lists are recommended if not using the default station lines because separate lines allow better tracking of resource utilization.

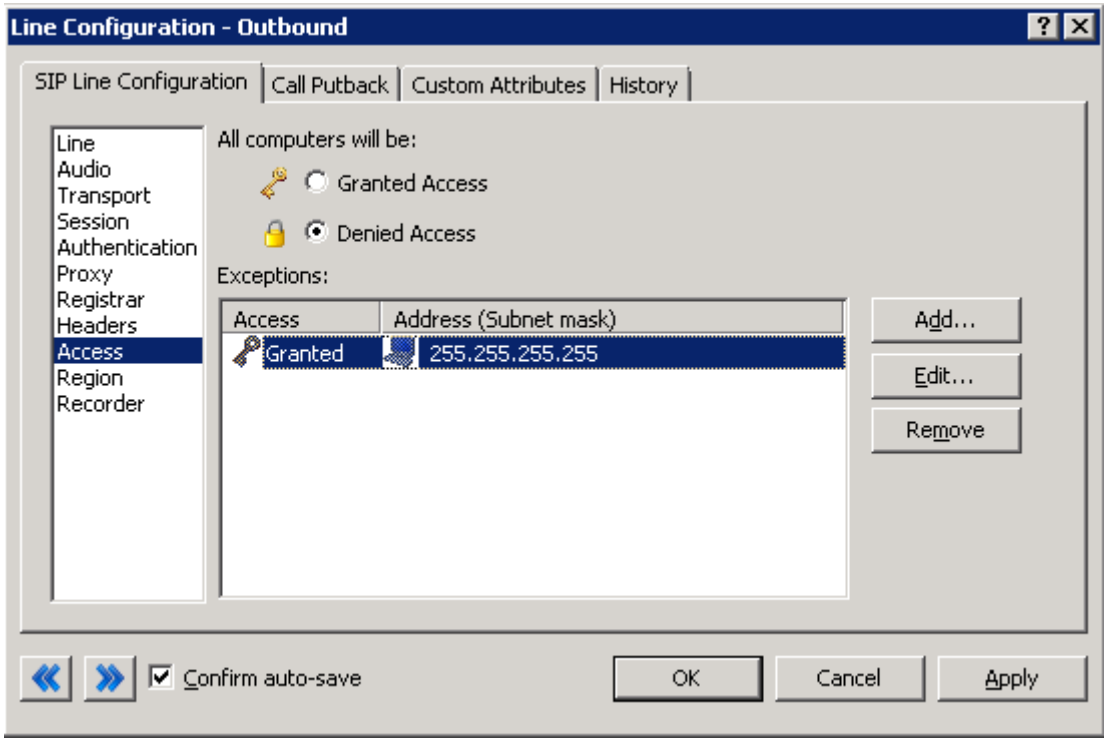


**Figure 7: Access Menu Line Configuration Page (note the 255.255.255.255 address is a sample, and the actual number should be respective to the customer needs).**

### 4.1.5.1 AT&T SIP Line

For the access menu, the radio button should be shifted to the value:

*By default, all computers will be:* ***Denied Access***.

In the access list below the radio button, the resolved IP address for each proxy server **MUST** be added. The "add menu" has a DNS lookup option if the only information provided by the carrier were FQDNs. This allows the IC server to talk to all the required elements of the SIP carrier.

### 4.1.5.2 Stations Line

In the case of the stations line, this is up to the discretion of the user. It is possible to enter in single IPs, IP groups (using subnet masks), or allow everything. The user has several options based on business needs and security requirements. However, note that only one line can be selected to "*Granted Access"* per port per IC server.

The reason why the SIP Carrier Line was selected to be ***Denied Access*** was because it has far fewer and less complicated entries than the line that will be supporting all the local endpoints.

#### 4.1.5.3 Region Menu

This should be set at the user discretion; however the user should take care to assure the location supports the proper codecs supported by the SIP Carrier.

In the case of AT&T, only G.711 (mu-law and A-law), G.726 and G.729 are supported, so selecting a location that does not have any of these as an option would cause the line not to function properly. Given bandwidth and situational information, AT&T typically recommends using G.729 (this can be the default by moving it to the top of the list, if it is not supported by the other end device, then it should fall back to the second in the list and so on).

**Important Note:** The IC server does not support G.726. It should not even be included in the supported list in a 4.0 GA IC server, as it could cause issues.

# 5 Multiple SIP Trunk Connections

## 5.1 Remote Site Architecture

Some integrations require that each remote site have a IP Flex Reach termination. For this to be supported, a local media server is required at each of these locations. This is to ensure an available route for RTP traffic back to the origination site. Each remote site in this architecture would include an ACME SBC, Media Server, and phone endpoints.

# 6 Fax Caveats

AT&T supports useable and functioning T.38 faxing. However if the customer would like to use an analog fax machine connected to the network, or if T.38 faxing is not an option, the way to circumvent this problem is with an analog to SIP FXS device connecting an analog fax machine to the IP network. The FXS device will pass the SIP information on allowing for G.711 pass-through (which is the carrying of the fax signal through the voice packets on the network). This has been tested using an AudioCodes Media Pack and a Cisco FXS card on its SIP Gateway.

**Note:** In the case of AT&T, it may be possible to use G.729 to do the pass-through faxing, however due to the compression used by the codec, and the sensitivity of fax communications, it is not recommended (and not tested) by Interactive Intelligence.

**Note:** Interactive Intelligence does not support T.38 SG3 faxing at the time this document was created. It does however, support G3 faxing.

# 7 Session Border Controller (SBC) configuration

*Taken from Acme's configuration document*

## 7.1 Support

Questions or help concerning this Acme Packet application note can be sent to:
support@acmepacket.com

## 7.2 Disclaimer

The content in this document is for informational purposes only and is subject to change by Acme Packet without notice. While reasonable efforts have been made in the preparation of this publication to assure its accuracy, Acme Packet assumes no liability resulting from technical or editorial errors or omissions, or for any damages resulting from the use of this information. Unless specifically included in a written agreement with Acme Packet, Acme Packet has no obligation to develop or deliver any future release or upgrade or any feature, enhancement or function.

**Emergency 911/E911 Services Limitations**

While AT&T IP Flexible Reach services support E911/911 calling capabilities in certain circumstances, there are significant limitations on how these capabilities are delivered. Please review the AT&T IP Flexible Reach Service Guide in detail to understand these limitations and restrictions.

## 7.3 Design Goals

The reference configuration represents the most common SIP to SIP deployment models: Originating SIP traffic and terminating to a SIP provider via the Net-Net SD. The config also supports bi-directional call-flows via Local-Policy routes.

This document will annotate the configuration with information on its general applicability. The intent is to:

- Minimize SIP to SIP interoperability issue's by standardizing field configurations
- Provide guidelines for new users for the Session Director
- Provide a configuration template, baselining the SIP to SIP configuration (with accompanying diagram)
- Flexibility: how resilient the configuration is and how adaptable the configuration is when turning up new SIP to SIP networks
- Performance: minimize the use of unnecessary configuration objects

## 7.4 Call Scenario1

This section includes a reference SIP signaling ladder diagram, where the Session Director is integrated as a Core/Enterprise Session Border Controller, performing SIP between Interactive Intelligence's equipment in the Core/Enterprise and AT&T's IP FlexReach service as the SIP Peer. Below is the representative call-flow.

SIP Signaling Device/IP PBX Call Setup via Acme SBC

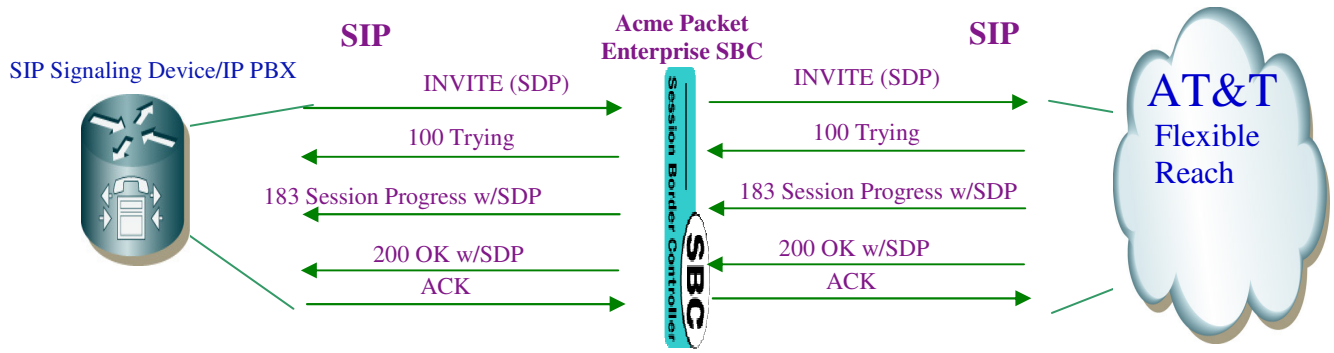

**Diagram 1: Representative Call-Flow for Acme Packet SD Interop with Interactive Intelligence**

## 7.5 Notes on Reference Configuration

This section includes a reference architecture diagram, where the Session Director is integrated as a Core/Enterprise Session Border Controller, performing SIP between Interactive Intelligence's equipment in the Core/Enterprise and AT&T's IP FlexReach service as the SIP Peer.

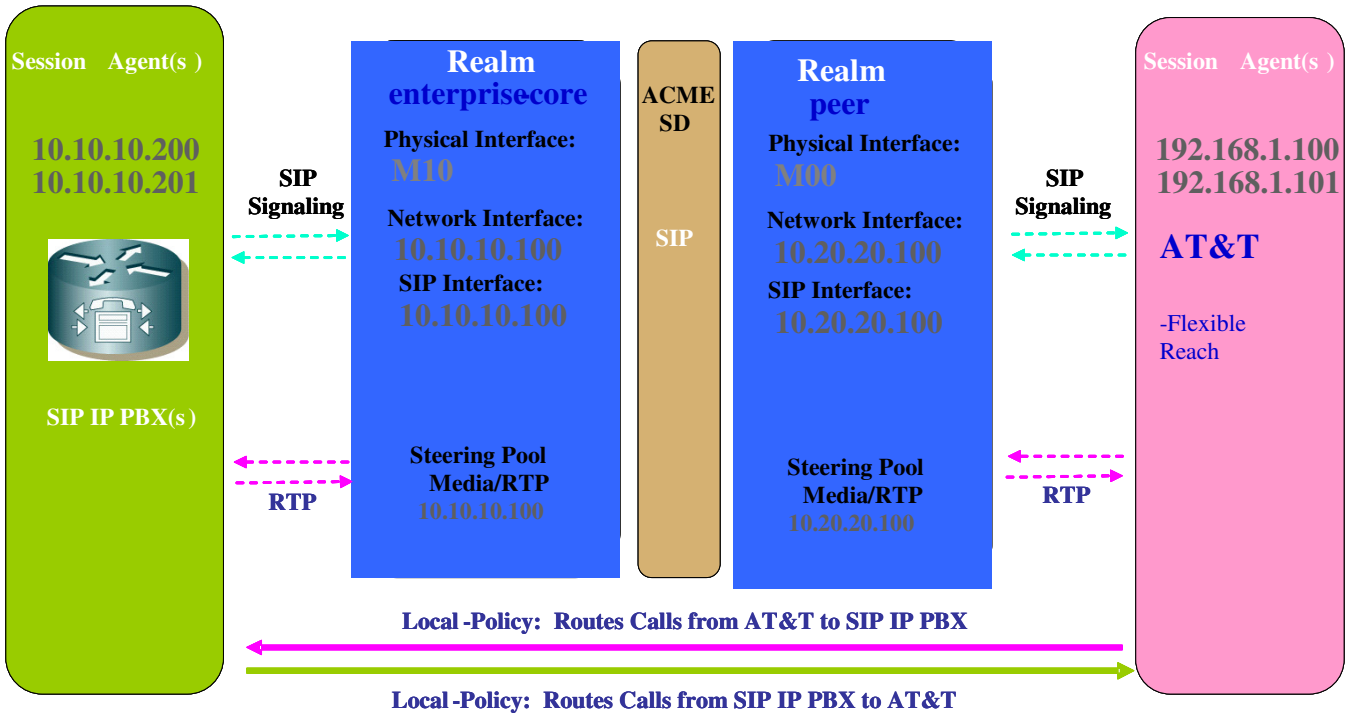


**Diagram 3: Example IP addresses for Reference Configuration**

## 7.6 Full Copyright Statement

## 7.7 Reference Configuration

```
local-policy
        from-address          *
        to-address            *
        source-realm          peer
        activate-time         N/A
        deactivate-time       N/A
        state                 enabled
        policy-priority       none
        policy-attribute
                next-hop            SAG:CPE
                realm               enterprise-core
                action              none
                terminate-recursion disabled
                carrier
                start-time          0000
                end-time            2400
                days-of-week        U-S
                cost                0
                app-protocol        SIP
                state               enabled
                media-profiles

local-policy
        from-address          *
        to-address            *
        source-realm          enterprise-core
        activate-time         N/A
        deactivate-time       N/A
        state                 enabled
        policy-priority       none
        policy-attribute
                next-hop            SAG:ATT
                realm               peer
                action              none
                terminate-recursion disabled
                carrier
                start-time          0000
                end-time            2400
                days-of-week        U-S
                cost                0
                app-protocol        SIP
                state               enabled
                media-profiles

media-manager
      state                          enabled
      latching                       disabled
      flow-time-limit                86400
      initial-guard-timer            43200
      subsq-guard-timer              43200
      tcp-flow-time-limit            86400
      tcp-initial-guard-timer        300
      tcp-subsq-guard-timer          300
      tcp-number-of-ports-per-flow   2
      hnt-rtcp                       disabled
      algd-log-level                 NOTICE
      mbcd-log-level                 NOTICE
      red-flow-port                  1985
```

```
        red-mgcp-port                   1986
        red-max-trans                   10000
        red-sync-start-time             5000
        red-sync-comp-time              1000
        media-policing                  enabled
        max-signaling-bandwidth         10000000
        max-untrusted-signaling         100
        min-untrusted-signaling         30
        app-signaling-bandwidth         0
        tolerance-window                30
        rtcp-rate-limit                 0
        min-media-allocation            32000
        min-trusted-allocation          1000
        deny-allocation                 1000
        anonymous-sdp                   disabled
        arp-msg-bandwidth               32000
        fragment-msg-bandwidth          0
        rfc2833-timestamp               enabled
        default-2833-duration           100
        rfc2833-end-pkts-only-for-non-sig disabled
        translate-non-rfc2833-event     disabled
        last-modified-date

network-interface
        name                            M00 (at&t/peer Facing )
        sub-port-id                     0
        hostname
        ip-address                      10.20.20.100(example address)
        pri-utility-addr
        sec-utility-addr
        netmask                         255.255.255.0
        gateway                         10.20.20.1
        sec-gateway
        gw-heartbeat
                state                           disabled
                heartbeat                       0
                retry-count                     0
                retry-timeout                   1
                health-score                    0
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout                     11
        hip-ip-list                     10.20.20.100
        ftp-address
        icmp-address
        snmp-address
        telnet-address


network-interface
        name                            M10 (enterprise/core facing)
        sub-port-id                     0
        hostname
        ip-address                      10.10.10.100 (example address)
        pri-utility-addr
        sec-utility-addr
        netmask                         255.255.255.0
        gateway                         10.10.10.1
        sec-gateway
```

```
        gw-heartbeat
                state                           disabled
                heartbeat                       0
                retry-count                     0
                retry-timeout                   1
                health-score                    0
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout                     11
        hip-ip-list                     10.10.10.100
        ftp-address
        icmp-address                    10.10.10.100
        snmp-address
        telnet-address


phy-interface
      name                            enterprise-core
      operation-type                  Media
      port                            0
      slot                            0
      virtual-mac
      admin-state                     enabled
      auto-negotiation                enabled
      duplex-mode
      speed

phy-interface
      name                            peer
      operation-type                  Media
      port                            0
      slot                            1
      virtual-mac
      admin-state                     enabled
      auto-negotiation                enabled
      duplex-mode
      speed



realm-config
        identifier                      peer
        addr-prefix                     0.0.0.0
        network-interfaces
                                        M00:0
        mm-in-realm                     enabled
        mm-in-network                   enabled
        mm-same-ip                      enabled
        mm-in-system                    enabled
        bw-cac-non-mm                   disabled
        msm-release                     disabled
        qos-enable                      disabled
        max-bandwidth                   0
        ext-policy-svr
        max-latency                     0
        max-jitter                      0
        max-packet-loss                 0
```

```
        observ-window-size              0
        parent-realm
        dns-realm
        media-policy
        in-translationid
        out-translationid
        in-manipulationid
        out-manipulationid
        class-profile
        average-rate-limit              0
        access-control-trust-level
        invalid-signal-threshold        0
        maximum-signal-threshold        0
        untrusted-signal-threshold      0
        deny-period                     30
        symmetric-latching              disabled
        pai-strip                       disabled
        trunk-context
        early-media-allow
        additional-prefixes
        restricted-latching             none
        restriction-mask                32
        accounting-enable               enabled
        user-cac-mode                   none
        user-cac-bandwidth              0
        user-cac-sessions               0
        net-management-control          disabled
        delay-media-update              disabled


realm-config
        identifier                      enterprise-core
        addr-prefix                     0.0.0.0
        network-interfaces
                                        M10:0
        mm-in-realm                     enabled
        mm-in-network                   enabled
        mm-same-ip                      enabled
        mm-in-system                    enabled
        bw-cac-non-mm                   disabled
        msm-release                     disabled
        qos-enable                      disabled
        max-bandwidth                   0
        ext-policy-svr
        max-latency                     0
        max-jitter                      0
        max-packet-loss                 0
        observ-window-size              0
        parent-realm
        dns-realm
        media-policy
        in-translationid
        out-translationid
        in-manipulationid
        out-manipulationid
        class-profile
        average-rate-limit              0
        access-control-trust-level
        invalid-signal-threshold        0
        maximum-signal-threshold        0
        untrusted-signal-threshold      0
```

```
        deny-period                   30
        symmetric-latching            disabled
        pai-strip                     disabled
        trunk-context
        early-media-allow
        additional-prefixes
        restricted-latching           none
        restriction-mask              32
        accounting-enable             enabled
        user-cac-mode                 none
        user-cac-bandwidth            0
        user-cac-sessions             0
        net-management-control        disabled
        delay-media-update            disabled

session-agent
        hostname                        192.168.1.200
        ip-address
        port                            5060
        state                           enabled
        app-protocol                    SIP
        app-type
        transport-method                UDP
        realm-id                        peer
        description                     at&t Session Agent Primary
        carriers
        allow-next-hop-lp               enabled
        constraints                     enabled
        max-sessions                    0
        max-inbound-sessions            0
        max-outbound-sessions           0
        max-burst-rate                  0
        max-inbound-burst-rate          0
        max-outbound-burst-rate         0
        max-sustain-rate                0
        max-inbound-sustain-rate        0
        max-outbound-sustain-rate       0
        min-seizures                    5
        min-asr                         0
        time-to-resume              0
        ttr-no-response                 300
        in-service-period               0
        burst-rate-window               0
        sustain-rate-window             0
        req-uri-carrier-mode            None
        proxy-mode
        redirect-action
        loose-routing                   enabled
        send-media-session              enabled
        response-map
        ping-method                     OPTIONS
        ping-interval                   300
        ping-in-service-response-codes
        out-service-response-codes
        media-profiles
        in-translationid
        out-translationid
        trust-me                        enabled
        request-uri-headers
        stop-recurse
        local-response-map
```

```
        ping-to-user-part
        ping-from-user-part
        li-trust-me                     disabled
        in-manipulationid
        out-manipulationid
        p-asserted-id
        trunk-group
        max-register-sustain-rate       0
        early-media-allow
        invalidate-registrations        disabled
        rfc2833-mode                    none
        rfc2833-payload                 0
        codec-policy


session-agent
        hostname                        192.168.1.201
        ip-address
        port                            5060
        state                           enabled
        app-protocol                    SIP
        app-type
        transport-method                UDP
        realm-id                        peer
        description                     at&t Session Agent Secondary
        carriers
        allow-next-hop-lp               enabled
        constraints                     enabled
        max-sessions                    0
        max-inbound-sessions            0
        max-outbound-sessions           0
        max-burst-rate                  0
        max-inbound-burst-rate          0
        max-outbound-burst-rate         0
        max-sustain-rate                0
        max-inbound-sustain-rate        0
        max-outbound-sustain-rate       0
        min-seizures                    5
        min-asr                         0
        time-to-resume              0
        ttr-no-response                 300
        in-service-period               0
        burst-rate-window               0
        sustain-rate-window             0
        req-uri-carrier-mode            None
        proxy-mode
        redirect-action
        loose-routing                   enabled
        send-media-session              enabled
        response-map
        ping-method                     OPTIONS
        ping-interval                   300
        ping-in-service-response-codes
        out-service-response-codes
        media-profiles
        in-translationid
        out-translationid
        trust-me                        enabled
        request-uri-headers
        stop-recurse
        local-response-map
```

```
        ping-to-user-part
        ping-from-user-part
        li-trust-me                     disabled
        in-manipulationid
        out-manipulationid
        p-asserted-id
        trunk-group
        max-register-sustain-rate       0
        early-media-allow
        invalidate-registrations        disabled
        rfc2833-mode                    none
        rfc2833-payload                 0
        codec-policy



session-agent
        hostname                        10.10.10.200
        ip-address
        port                            5060
        state                           enabled
        app-protocol                    SIP
        app-type
        transport-method                UDP
        realm-id                        *
        description                     enterprise Session Agent SIP
        carriers
        allow-next-hop-lp               enabled
        constraints                     disabled
        max-sessions                    0
        max-inbound-sessions            0
        max-outbound-sessions           0
        max-burst-rate                  0
        max-inbound-burst-rate          0
        max-outbound-burst-rate         0
        max-sustain-rate                0
        max-inbound-sustain-rate        0
        max-outbound-sustain-rate       0
        min-seizures                    5
        min-asr                         0
        time-to-resume              0
        ttr-no-response                 0
        in-service-period               0
        burst-rate-window               0
        sustain-rate-window             0
        req-uri-carrier-mode            None
        proxy-mode
        redirect-action
        loose-routing                   enabled
        send-media-session              enabled
        response-map
        ping-method                     OPTIONS
        ping-interval                   300
        ping-in-service-response-codes
        out-service-response-codes
        media-profiles
        in-translationid
        out-translationid
        trust-me                        disabled
        request-uri-headers
        stop-recurse
```

```
        local-response-map
        ping-to-user-part
        ping-from-user-part
        li-trust-me                     disabled
        in-manipulationid
        out-manipulationid
        p-asserted-id
        trunk-group
        max-register-sustain-rate       0
        early-media-allow
        invalidate-registrations        disabled
        rfc2833-mode                    none
        rfc2833-payload                 0
        codec-policy

session-agent
        hostname                        10.10.10.201
        ip-address
        port                            5060
        state                           enabled
        app-protocol                    SIP
        app-type
        transport-method                UDP
        realm-id                        *
        description                     enterprise Session Agent SIP
        carriers
        allow-next-hop-lp               enabled
        constraints                     disabled
        max-sessions                    0
        max-inbound-sessions            0
        max-outbound-sessions           0
        max-burst-rate                  0
        max-inbound-burst-rate          0
        max-outbound-burst-rate         0
        max-sustain-rate                0
        max-inbound-sustain-rate        0
        max-outbound-sustain-rate       0
        min-seizures                    5
        min-asr                         0
        time-to-resume              0
        ttr-no-response                 0
        in-service-period               0
        burst-rate-window               0
        sustain-rate-window             0
        req-uri-carrier-mode            None
        proxy-mode
        redirect-action
        loose-routing                   enabled
        send-media-session              enabled
        response-map
        ping-method                     OPTIONS
        ping-interval                   300
        ping-in-service-response-codes
        out-service-response-codes
        media-profiles
        in-translationid
        out-translationid
        trust-me                        disabled
        request-uri-headers
        stop-recurse
        local-response-map
```

```
        ping-to-user-part
        ping-from-user-part
        li-trust-me                     disabled
        in-manipulationid
        out-manipulationid
        p-asserted-id
        trunk-group
        max-register-sustain-rate       0
        early-media-allow
        invalidate-registrations        disabled
        rfc2833-mode                    none
        rfc2833-payload                 0
        codec-policy


session-group
      group-name                        ATT
      description
      state                             enabled
      app-protocol                      SIP
      strategy                          RoundRobin
      dest
                                        192.168.1.200
                                        192.168.1.201
      trunk-group
      sag-recursion                     enabled
      stop-sag-recurse                  401,407

session-group
      group-name                        CPE
      description
      state                             enabled
      app-protocol                      SIP
      strategy                          RoundRobin
      dest
                                        10.10.10.200
                                        10.10.10.201
      trunk-group
      sag-recursion                     enabled
      stop-sag-recurse                  401,407



sip-config
      state                             enabled
      operation-mode                    dialog
      dialog-transparency               enabled
      home-realm-id                     peer
      egress-realm-id                   enterprise-core
      nat-mode
      registrar-domain
      registrar-host
      registrar-port                    0
      register-service-route            always
      init-timer                        500
      max-timer                         4000
      trans-expire                      32
      invite-expire                     180
      inactive-dynamic-conn             32
      enforcement-profile
```

```
      pac-method
      pac-interval                  10
      pac-strategy                  PropDist
      pac-load-weight               1
      pac-session-weight            1
      pac-route-weight              1
      pac-callid-lifetime           600
      pac-user-lifetime             3600
      red-sip-port                  1988
      red-max-trans                 10000
      red-sync-start-time           5000
      red-sync-comp-time            1000
      add-reason-header             disabled
      sip-message-len               0
      enum-sag-match                disabled
      extra-method-stats            disabled
      rph-feature                   disabled
      nsep-user-sessions-rate       0
      registration-cache-limit      0
      options                       add-prov-to-tag=no
                                      insert-arp-header
                                      max-udp-length=0
                                      set-inv-exp-at-100-resp


sip-interface
      state                         enabled
      realm-id                      peer
      description
      sip-port
            address                       10.20.20.100
            port                          5060
            transport-protocol            UDP
            tls-profile
            allow-anonymous               agents-only
      carriers
      trans-expire                  0
      invite-expire                 0
      max-redirect-contacts         0
      proxy-mode
      redirect-action
      contact-mode                  none
      nat-traversal                 none
      nat-interval                  30
      tcp-nat-interval              30
      registration-caching          disabled
      min-reg-expire                300
      registration-interval         3600
      route-to-registrar            disabled
      secured-network               disabled
      teluri-scheme                 disabled
      uri-fqdn-domain
      trust-mode                    all
      max-nat-interval              3600
      nat-int-increment             10
      nat-test-increment            30
      sip-dynamic-hnt               disabled
      stop-recurse                  401,407
      port-map-start                0
      port-map-end                  0
      in-manipulationid
```

```
      out-manipulationid             Privacy
      sip-ims-feature                disabled
      operator-identifier
      anonymous-priority             none
      max-incoming-conns             0
      per-src-ip-max-incoming-conns  0
      inactive-conn-timeout          0
      untrusted-conn-timeout         0
      network-id
      ext-policy-server
      default-location-string
      charging-vector-mode           none
      charging-function-address-mode none
      ccf-address
      ecf-address
      term-tgrp-mode                 none
      implicit-service-route         disabled
      rfc2833-payload                101
      rfc2833-mode                   transparent
      constraint-name
      response-map
      local-response-map
      enforcement-profile
      refer-call-transfer            disabled
      route-unauthorized-calls
      tcp-keepalive                  none
      add-sdp-invite                 disabled
      add-sdp-profiles

sip-interface
      state                          enabled
      realm-id                       enterprise-core
      description
      sip-port
            address                        10.10.10.100
            port                           5060
            transport-protocol             UDP
            tls-profile
            allow-anonymous                all
      carriers
      trans-expire                   0
      invite-expire                  0
      max-redirect-contacts          0
      proxy-mode
      redirect-action
      contact-mode                   none
      nat-traversal                  none
      nat-interval                   30
      tcp-nat-interval               90
      registration-caching           disabled
      min-reg-expire                 300
      registration-interval          3600
      route-to-registrar             disabled
      secured-network                disabled
      teluri-scheme                  disabled
      uri-fqdn-domain
      trust-mode                     all
      max-nat-interval               3600
      nat-int-increment              10
      nat-test-increment             30
      sip-dynamic-hnt                disabled
```

```
      stop-recurse                    401,407
      port-map-start                  0
      port-map-end                    0
      in-manipulationid
      out-manipulationid              Privacy
      sip-ims-feature                 disabled
      operator-identifier
      anonymous-priority              none
      max-incoming-conns              0
      per-src-ip-max-incoming-conns   0
      inactive-conn-timeout           0
      untrusted-conn-timeout          0
      network-id
      ext-policy-server
      default-location-string
      charging-vector-mode            pass
      charging-function-address-mode  pass
      ccf-address
      ecf-address
      term-tgrp-mode
      implicit-service-route          disabled
      rfc2833-payload                 101
      rfc2833-mode                    transparent
      constraint-name
      response-map
      local-response-map
      enforcement-profile
      refer-call-transfer             disabled
      route-unauthorized-calls
      tcp-keepalive                   none
      add-sdp-invite                  disabled
      add-sdp-profiles

sip-manipulation
      name                            Privacy
      description                     changing ip
      header-rule
            name                            PAI_Header
            header-name                     P-Asserted-Identity
            action                          manipulate
            comparison-type                 case-sensitive
            match-value
            msg-type                        any
            new-value
            methods
            element-rule
                  name                            PAI_Local_IP
                  parameter-name
                  type                            uri-host
                  action                          replace
                  match-val-type                  any
                  comparison-type                 case-sensitive
                  match-value
                  new-value                       $LOCAL_IP
      header-rule
            name                            PPI_Header
            header-name                     P-Preferred-Identity
            action                          manipulate
            comparison-type                 case-sensitive
            match-value
            msg-type                        any
```

```
      new-value
      methods
      element-rule
            name                              PPI_Local_IP
            parameter-name
            type                              uri-host
            action                            replace
            match-val-type                    any
            comparison-type                   case-sensitive
            match-value
            new-value                         $LOCAL_IP
header-rule
      name                              From_Header
      header-name                       From
      action                            manipulate
      comparison-type                   case-sensitive
      match-value
      msg-type                          request
      new-value
      methods
      element-rule
            name                              From_header
            parameter-name
            type                              uri-host
            action                            replace
            match-val-type                    any
            comparison-type                   case-sensitive
            match-value
            new-value                         $LOCAL_IP
header-rule
      name                              To_Header
      header-name                       To
      action                            manipulate
      comparison-type                   case-sensitive
      match-value
      msg-type                          request
      new-value
      methods
      element-rule
            name                              To_header
            parameter-name
            type                              uri-host
            action                            replace
            match-val-type                    any
            comparison-type                   case-sensitive
            match-value
            new-value                         $REMOTE_IP
header-rule
      name                              RPI_Header
      header-name                       Remote-Party-ID
      action                            manipulate
      comparison-type                   case-sensitive
      match-value
      msg-type                          any
      new-value
      methods
      element-rule
            name                              RPI_header
            parameter-name
            type                              uri-host
            action                            replace
```

```
            match-val-type                  any
            comparison-type                 case-sensitive
            match-value
            new-value                       $LOCAL_IP
header-rule
      name                              Refer_header
      header-name                       Referred-By
      action                            manipulate
      comparison-type                   case-sensitive
      match-value
      msg-type                          any
      new-value
      methods
      element-rule
            name                              referredbyhdr
            parameter-name
            type                              uri-host
            action                            replace
            match-val-type                    any
            comparison-type                   case-sensitive
            match-value
            new-value                         $LOCAL_IP
header-rule
      name                              ReferredTo
      header-name                       Refer-To
      action                            manipulate
      comparison-type                   case-sensitive
      match-value
      msg-type                          any
      new-value
      methods
      element-rule
            name                              refertohdr
            parameter-name
            type                              uri-host
            action                            replace
            match-val-type                    any
            comparison-type                   case-sensitive
            match-value
            new-value                         $REMOTE_IP

header-rule
      name                              ContactHdr
      header-name                       Contact
      action                            manipulate
      comparison-type                   case-sensitive
      match-value
      msg-type                          any
      new-value
      methods
      element-rule
            name                              ContactHostReplace
            parameter-name
            type                              uri-host
            action                            replace
            match-val-type                    any
            comparison-type                   case-sensitive
            match-value
            new-value                         $LOCAL_IP
last-modified-by
```

```
steering-pool
        ip-address                      10.20.20.100
        start-port                      16384
        end-port                        32767
        realm-id                        peer
        network-interface

steering-pool
        ip-address                      10.10.10.100
        start-port                      16384
        end-port                        32767
        realm-id                        enterprise-core
        network-interface


system-config
        hostname                        enterprise
        description                     enterpriseSBC
        location
        mib-system-contact
        mib-system-name
        mib-system-location
        snmp-enabled                    enabled
        enable-snmp-auth-traps          disabled
        enable-snmp-syslog-notify       enabled
        enable-snmp-monitor-traps       enabled
        enable-env-monitor-traps        disabled
        snmp-syslog-his-table-length    1
        snmp-syslog-level               WARNING
        system-log-level                WARNING
        process-log-level               NOTICE
        process-log-ip-address          0.0.0.0
        process-log-port                0
        call-trace                      disabled
        internal-trace                  disabled
        log-filter                      all
        default-gateway                 10.20.20.1 (example address)
        restart                         enabled
        exceptions
        telnet-timeout                  0
        console-timeout                 0
        remote-control                  enabled
        link-redundancy-state           disabled
```

# 8 E911 Support

AT&T currently supports E911 support via linking provided phone numbers to internal databases. This allows for dynamic updates, and accurate routing of emergency calls based on originating location. Essentially, it is how a standard phone line would be expected to function.

# 9 Troubleshooting

Depending on how the service was contracted, support is available for the Interactive Intelligence solution set. If any Interactive Intelligence related materials were obtained via direct sale from Interactive Intelligence then a support contact procedure should have been established at that time. If Interactive Intelligence related materials were obtained through a reseller or partner, then that party should be contacted for support options.

Testlab.inin.com
E-Mail Support: support@ININ.com
Support Phone: 1-800-267-1364