



## IP Office Technical Tip

**Tip no:** 200

**Release Date:** January 23, 2008

**Region:** GLOBAL

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### **IP Office Session Initiation Protocol (SIP) Configuration Primer**

There are many Internet Telephony Service Providers (ITSP) offering SIP services and the programming and operation of SIP can vary between them. As a new protocol for deployment, mis-configuration is a leading cause of IP Office SIP issues reported to Avaya Support. IP Office SIP in release 4.0.10 and greater can be programmed to use options found in several Internet Engineering Task Force (IETF) SIP related Request For Comment (RFC) documents. With this document as reference the user can adjust the IP Office SIP configuration to match their respective ITSP's SIP messaging format.

Individuals using this document must have an understanding of SIP protocols and standards found in the IETF RFC's. Where applicable throughout this document the IETF RFC related to the SIP format discussed or displayed is listed. This serves to validate IP Office operation per the specified IETF RFC and allow further review of the subject matter presented. A basic understanding and knowledge of IP Office SIP configuration can be found in Global IP Office Technical Tip 188 (Tech Tip 188).

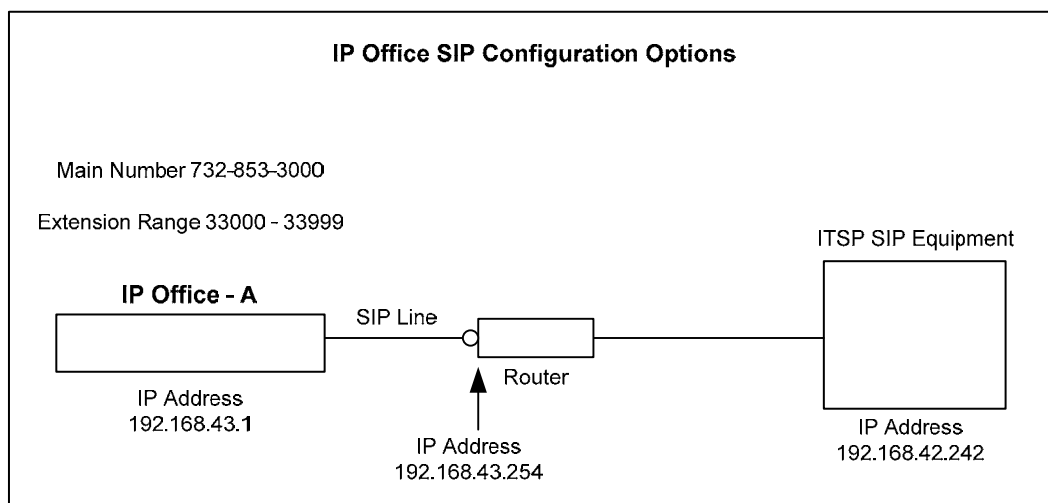
**NOTE 1** - *Ensure your SIP ITSP is listed in the "ITSP Interoperability" table found in IP Office Technical Bulletin 80 - General Availability of IP Office 4.0 Software and IP Office 500 page 48. Updates of this list may be found at <http://marketingtools.example.com/knowledgebase/>.*

**NOTE 2** - *Throughout this document the IP Office Monitor is used to view the SIP message format after the IP Office SIP configuration changes. Monitor is not an IP Office application supported for field use. The operation, function and display of Monitor can change without notice or documentation to the field. The ITSP equipment may have a capability similar to Monitor, inquire and use this ability during deployment if possible. SIP connections can also be monitored using commercial or freeware packet capture software.*

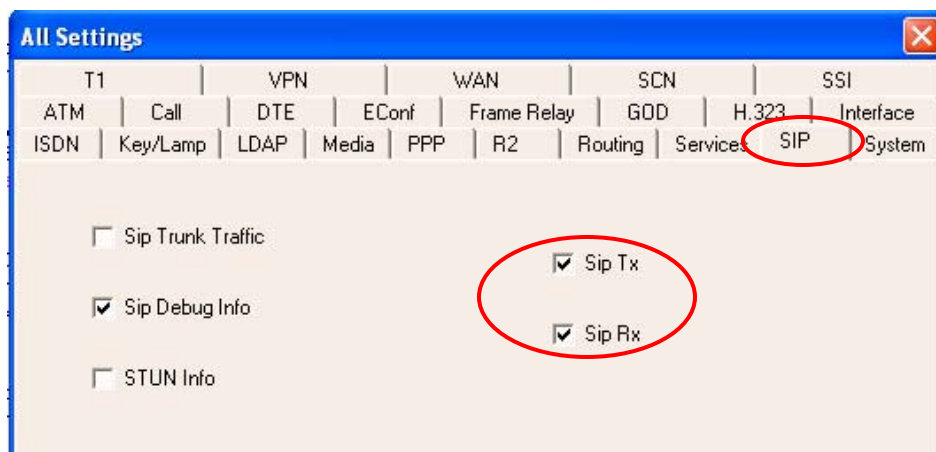
#### **Forward**

The following information, captures and displays are demonstrated using IP Office release 4.0.10. The diagram on the next page represents a typical network configuration connecting IP Office – A and the ITSP's SIP Equipment.

Throughout this document all example calls are initiated from IP Office – A extension 33302 (IP Office > User > SIP tab = 17328533302). All calls from the originating extension will be dialed to (called number) number 1-732-852-2203 located within, or available through, the ITSP network.



The IP Office Monitor SIP tab is set per the screen shot below and connect to IP Office – A (IP Address = 192.168.43.1)



**IP Office SIP Tel INVITE**

As a starting reference point the IP Office SIP configuration to the ISTP SIP service is configured per Tech Tip 188. With Monitor connected to IP Office – A, a call is initiated (dialed) from extension 33302 to 8 (designating the SIP Line in the short codes and ARS) and number 17328522203. After the call is completed and disconnected, pause or “file > save as” the Monitor to review the SIP information. The below represents the SIP INVITE sent by IP Office – A, extension 33302 (identified within the User > SIP tab as number 17328522203) to dialed number 17328522203 on remote IP Office – B. Search for and identify the first “INVITE” originating from IP Office – A.

## Display / Saved from Monitor

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**SIP Tx:** UDP 192.168.43.1:5060 -> 192.168.42.242:5060

- 1) **INVITE Tel:+17328522203** SIP/2.0  
Via: SIP/2.0/UDP 192.168.43.1:5060;rport;branch=z9hG4bKfe625964dd867f9a89f19fafcc65d335
  - 2) **From: 17328533302** <sip:17328533302@example.com>;tag=a083c01064cbeccf
  - 3) **To: Tel:+17328522203**  
Call-ID: c1de74d5f65d53f7779166eb1a428bee@192.168.43.1  
CSeq: 1962617718 INVITE
  - 4) **Contact: 17328533302** <sip:17328533302@192.168.43.1:5060;transport=udp>  
Max-Forwards: 70  
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE  
Content-Type: application/sdp  
Content-Length: 299  
  
v=0  
o=UserA 310281170 981016607 IN IP4 192.168.43.1
  - 5) s=Session SDP  
c=IN IP4 192.168.43.1  
t=0 0
  - 6) m=audio 49152 RTP/AVP 18 4 8 0 101
  - 7) a=rtpmap:18 G729/8000  
a=rtpmap:4 G723/8000  
a=rtpmap:8 PCMA/8000  
a=rtpmap:0 PCMU/8000  
a=fmtp:18 annexb = no
  - 8) a=rtpmap:101 telephone-event/8000  
a=fmtp:101 0-15
- 

## SIP Tel INVITE Review

**SIP Tx:** - This line identifies SIP Transmit (**Tx**) information from the IP Office the Monitor is connected to. Received SIP information will be displayed in the trace beginning with “**SIP Rx:**”

1) The INVITE request. It is “Tel” formatted and contains only the dialed number with the SIP version Identifier. The called number (17328522203) is prefixed with a “+ “ (plus sign) denoting a Global E.164 number format per IETF RFC 3966. Note the IP Office default to the SIP Tel format.

2) The “From” field displays the defined SIP > User tab information, in this case number 17328522203, in addition to “17328522203@example.com”. The User number and the fully qualified domain name as entered into the SIP Line tab “ITSP Domain Name” entry. The “From” number can be designated as an E.164 number if desired by placing the “ + “ in front of the number in the User > SIP tab field in the call originating system. See “User > SIP Tab” section on next page.

3) The “To” field is “Tel” formatted, displaying only the dialed number (17328522203) prefixed with a “+ “ (plus sign) denoting a Global E.164 number format (IETF RFC 3966).

4) Contact information

The User > SIP tab data is displayed (1732853302) by selecting “Use User Data” in the SIP URI “Local URI” definition.

The IP address of the device sending the INVITE – 192.168.43.1

The port the INVITE sent from – 5060 – set in the SIP Line “Network Configuration”

The port type (TCP or UDP) – udp – set in the SIP Line “Network Configuration”

5) The SDP session being setup.

6) The type of SDP session being setup denoted by the “m=audio” (IETF RFC 4566).

The port being negotiated for the media stream – 49152

The type of application/media – RTP standard AVP (Audio, Video, Protocol).

7) A list of the media attributes denoted by the “a=” entries (IETF RFC 2833).

Available codec’s for audio negotiation (the codec setting in the SIP Line “Network Configuration” set for “Automatic”).

“telephone-event/8000” denotes the RTP payload type and the default sampling rate (8000).

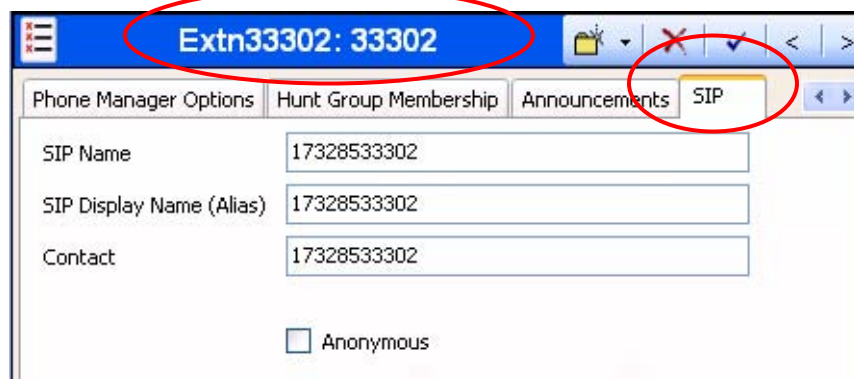
8) “0-15” denotes the DTMF events supported within the “telephone-event” payload (DTMF digits 0 to 15). Instead of sending actual DTMF tones, representations of the DTMF tones are sent (what key pressed and for how long) and the receiving site is expected to reproduce the DTMF tones locally (RFC 2833).

**NOTE 3** - The SIP “Tel” URI described above is a valid SIP format per IETF RFC 3966. It is possible some ITSP’s may not support the Tel format at this time. If a different SIP URI format is desired, changing the Tel URI format is the responsibility of the first network proxy receiving the INVITE (the ITSP provider equipment) per RFC 3966.

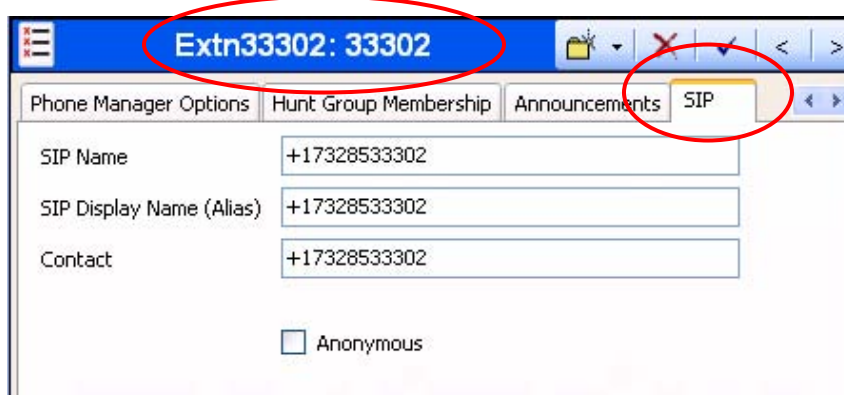
**User > SIP Tab - Calling Number E.164 designation**

By setting Local URI, Contact, Display Name to ‘Use User Data’ in the SIP URI programming each User now displays a tab for a SIP “User Data” entry. These entries can be individually programmed and are then be used to populate the SIP URI fields. By default this field is populated with the “Name” entry of the User. The example below displays the addition of the global E.164 number designator “ + “ (plus sign) to the calling number (IETF RFC 3966).

**Configured WITHOUT the “ + “ (plus sign) E.164 number designator**



## Configured WITH the “ + “ (plus sign) E.164 number designator



After configuring the User > SIP tab local number with the “ + “ sign save the configuration. Connect / reconnect Monitor to IP Office – A, a test call is repeated. After successfully completing the call and disconnecting, pause, or “file > save as” the Monitor to review the SIP information. The below represents the SIP INVITE sent by IP Office – A, extension 33302 (identified within the User > SIP tab as number +17328533302) to dialed number 17328522203. Search for and identify the “INVITE” originating from IP Office – A.

### Displayed / Saved from Monitor

SIP Tx: UDP 192.168.43.1:5060 -> 192.168.42.242:5060

- ```
INVITE Tel:+17328522203 SIP/2.0
Via: SIP/2.0/UDP 192.168.43.1:5060;rport;branch=z9hG4bKfe625964dd867f9a89f19fafcc65d335
1) From: +17328533302 <sip:+17328533302@example.com>;tag=a083c01064cbecf
   To: Tel:+17328522203
   Call-ID: c1de74d5f65d53f7779166eb1a428bee@192.168.43.1
   CSeq: 1962617718 INVITE
2) Contact: +17328533302 <sip:+17328533302@192.168.43.1:5060;transport=udp>
   Max-Forwards: 70
   Allow: INVITE, ACK, CANCEL, OPTIONS, BYE
   Content-Type: application/sdp
   Content-Length: 299
```

**NOTE 4** - from this point forward only the Monitor fields changed or affected will be displayed for review.

Note that the “From” and “Contact” fields of the call originator now include the Global E.164 number designation. (Items #1 and #2).

### IP Office SIP INVITE - no Domain Name

Removing the entry in the SIP Line “ITSP Domain Name” field will remove the Domain Name from the INVITE and will automatically insert the IP address of the calling unit in this location in the transmitted INVITE.

## Domain Name Removed

| SIP Line                        | SIP URI                             |
|---------------------------------|-------------------------------------|
| Line Number                     | 16                                  |
| ITSP Domain Name                |                                     |
| ITSP IP Address                 | 192 . 168 . 43 . 1                  |
| Primary Authentication Name     |                                     |
| Primary Authentication Password |                                     |
| Registration Required           | <input type="checkbox"/>            |
| In Service                      | <input checked="" type="checkbox"/> |
| Use Tel URI                     | <input type="checkbox"/>            |
| VoIP Silence Suppression        | <input type="checkbox"/>            |
| Out Of Band DTMF                | <input type="checkbox"/>            |

### Display / Saved from Monitor

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**SIP Tx:** UDP 192.168.43.1:5060 -> 192.168.42.242:5060

1) INVITE Tel:+17328522203 SIP/2.0  
Via: SIP/2.0/UDP 192.168.43.1:5060;rport;branch=z9hG4bKfe625964dd867f9a89f19fafcc65d335  
**From:** +17328533302 <sip:+17328533302@192.168.43.1>;tag=a083c01064cbeccf  
To: Tel:+17328522203  
Call-ID: c1de74d5f65d53f7779166eb1a428bee@192.168.43.1  
CSeq: 1962617718 INVITE  
Contact: +17328533302 <sip:+17328533302@192.168.43.1:5060;transport=udp>  
Max-Forwards: 70  
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE  
Content-Type: application/sdp  
Content-Length: 299

---

1) The "From" field now displays the SIP > User tab information and the IP Address of the transmitting IP Office "+17328533302 @192.168.43.1". The domain name is replaced by the IP address. This can be compared to the two previous Monitor displays were the IP Office was configured with the domain name entry earlier in this document.

### **IP Office SIP INVITE (non-Tel format)**

As previously reviewed, when configured per Tech Tip 188 the IP Office will default to the SIP "Tel" format, a valid format per the SIP standards (RFC 3966). It is possible that the ITSP currently does not support this format or requests a different SIP format described in RFC 3261. To change the IP Office SIP format to a non-Tel format add the below text to the SIP Line ARS Short Code.

### Non-Tel format - SIP Line ARS Short Code

**Edit Short Code**

Code: N;

Feature: Dial

Telephone Number: N"@192.168.42.242;user=phone"

Line Group Id: 16

Locale: United States (US English)

Force Account Code:

Buttons: OK, Cancel

Note the addition of the text after the Telephone Number "N" entry – "@192.168.42.242;user=phone". Add this text as listed – inserting the ITSP's IP address of their SIP equipment and save the IP Office configuration.

### Non-Tel format - SIP Line ARS Short Code

**ARS**

ARS Route Id: 52

Route Name: [8]

Dial Delay Time: System

In Service:

Time Profile: <None>

Secondary Dial tone:  SystemTone

Check User Call Barring:

Out of Service Route: <None>

Out of Hours Route: <None>

| Code | Telephone Number              | Feature        | Line Group Id |
|------|-------------------------------|----------------|---------------|
| 11   | 911                           | Dial Emergency | 0             |
| 911  | 911                           | Dial Emergency | 0             |
| N;   | N"@192.168.42.242;user=phone" | Dial           | 16            |

With the above text added to the SIP Line ARS Short Code a call is initiated from extension 33302 to number 17328522203. After the call is completed and disconnected, pause or "file > save as" the Monitor to review the SIP information.

## Display / Saved from Monitor

---

SIP Tx: UDP 192.168.43.1:5060 -> 192.168.42.242:5060

- 1) **INVITE sip:17328522203@192.168.42.242;user=phone SIP/2.0**  
Via: SIP/2.0/UDP 192.168.43.1:5060;rport;branch=z9hG4bKa083c01064cbeccf170c59d70bb9a58e  
From: 17328533302 <sip:17328533302@example.com>;tag=6e4d1c7bb5cc7a9c
  - 2) **To: <sip:17328522203@192.168.42.242;user=phone>**  
Call-ID: 1d852fc75e04499646a8b960e4ffbee2@192.168.43.1  
CSeq: 1265249445 INVITE  
Contact: 17328533302 <sip:17328533302@192.168.43.1:5060;transport=udp>  
Max-Forwards: 70  
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE  
Content-Type: application/sdp  
Content-Length: 301
- 

The “INVITE” and the “To:” fields (1 and 2 above) have been changed from the SIP Tel format. These fields are now formatted similar to the SIP URI formats described in RFC 3261 section 19.1.3. Additional changes can be made in the SIP Line ARS Short Code to adjust the IP Office SIP INVITE to match that expected by the SIP ITSP per RFC 3261. Coordinate these changes with the SIP ITSP.

Examples:

“@192.168.42.242”

“@example.com”

### **IP Office SIP INVITE (no Tel format) Global E.164 numbering**

When changing the IP Office SIP INVITE to the non-Tel format note that the E.164 number designator “+” (plus sign) is not automatically appended to the outgoing (called) number. If required to use the non-Tel SIP format and the E.164 designator by the ITSP this can be accommodated. Modify the IP Office SIP Line ARS Short Code by inserting the designator at the beginning of the Telephone Number “N” as seen below.

+N“@192.168.42.242;user=phone”

**Non-Tel SIP format – Global E.164 numbering**

| Edit Short Code    |                               |
|--------------------|-------------------------------|
| Code               | N:                            |
| Feature            | Dial                          |
| Telephone Number   | +N"@192.168.42.242;user=phone |
| Line Group Id      | 16                            |
| Locale             | United States (US English)    |
| Force Account Code | <input type="checkbox"/>      |

With the above “ + “ designator added to the SIP Line ARS Short Code a call is initiated from extension 33302 to number 17328522203. After the call is completed and disconnected, pause or “file > save as” the Monitor to review the SIP information.

### Display / Saved from Monitor

---

SIP Tx: UDP 192.168.43.1:5060 -> 192.168.42.242:5060

- 1) INVITE sip:0017328522203@192.168.42.242;user=phone SIP/2.0  
Via: SIP/2.0/UDP 192.168.43.1:5060;rport;branch=z9hG4bK35175e7b81bd07d91694a8b06af07a6b  
From: 17328533302 <sip:17328533302@example.com>;tag=de12aa85559051d7
  - 2) To: <sip:0017328522203@192.168.42.242;user=phone>  
Call-ID: 398855d9b9dfcd268ab5285cefef05d4@192.168.43.1  
CSeq: 859474496 INVITE  
Contact: 17328533302 <sip:17328533302@192.168.43.1:5060;transport=udp>  
Max-Forwards: 70  
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE  
Content-Type: application/sdp  
Content-Length: 301
- 

Note in the above trace that the called number located in the “INVITE” (1) and the “To:” (2) fields have been modified. In each field / number the “+” has been replaced with the equivalent “00” (zero, zero) - 00 being the prefix for international calls (RFC 3966).

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