Lumen® SIP Trunking

E911 fundamentals for SIP Trunking and required hardware configuration for customer premise equipment

Understanding groups/sites and PPU’s

- Each PPU (primary place of use), also known as a group or site, has one number identified as the phone number that will be utilized as the calling line ID for any 911 calls from that PPU.
- Customer can have one phone number per location at no charge that can be identified as the 911 Group Calling line ID.
- If the phone number is ported to Lumen, the customer admin is required to do a follow up reconfiguration in the portal to swap out any temporary phone numbers configured by the system designer.
- Each telephone number associated with a particular location (PPU/group) will use that same identified 911 Group Calling line ID number.
- Any additional numbers the customer wishes to have Lumen register for mobility or adding location level data (limit of 20 characters) with our 911 carrier will have a nominal monthly fee per number.
- Out-pulsing a toll-free number or a non-Lumen number not inventoried in the same enterprise/tenant is considered an alien telephone number. When calling 911 from an alien number, the customer will be charged a minimum fee of $75 per call since these calls are sent to the National Call Center with no address information.
- **This is a contract violation and a public safety risk** since these alien telephone numbers route to a national 911 call center with no address identifier.

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Critical Public Safety Bulletin: Required 911 settings for SIP Trunking

Customer premise hardware settings for 911

For 911 to work properly, several headers in the customer premise hardware must be formatted with the proper information.

- Each ‘group’ (PPU/site) is configured with one phone number to be used for 911 services. This number is called the “Group Caller ID.” All member telephone numbers share the 911 address assigned to that one phone number.
- If the Invite is not formatted properly, the Group Caller ID will not be delivered to the 911 operator and the address information will not automatically be provided. **This creates a public safety concern as the caller will not get routed automatically to the correct end office and the callers’ location will not be shown.** As such, the caller must be able to speak to tell the operator where they are located. At that point, the operator will be transferred to the correct end office resulting in a delay in responding.
- SIP traffic for each trunk (including 911) must use the pilot TN (telephone number) as the PAI.
- The source IP + port of the registration must be used as the source IP + port of all subsequent SIP traffic for that trunk.
- The host portion of SIP URI for the from, to and PAI SIP invite headers must be the customer domain provisioned in the VoIP portal.

See the next slide for detailed instructions.

Customer premise hardware settings for 911

SIP Header	Userinfo or username	Host portion of SIP URI or realm
From	**SIP Trunking user TN**	**Customer domain provisioned by Lumen**
Example:	4785553710	voip.centurylink.com
From:	<sip:4785553710@voip.centurylink.com>;tag=4247F498-864	
To	**Dialed TN**	**Customer domain provisioned by Lumen**
Example:	911	voip.centurylink.com
To:	<sip:911@voip.centurylink.com>	
Contact		**Source IP & port (Customer IP & port)**
Example:		63.157.48.130:5060
Contact:	<sip:4785553710@63.157.48.130:5060>	
Authorization	**Trunk authentication username provisioned by Lumen**	**Customer domain provisioned by Lumen**
Example:	Digest username="257389-9132756079"	realm="voip.centurylink.com"
Authorization:	Digest username="257389-9132756079",realm="voip.centurylink.com",uri=sip:911@voip.centurylink.com:5100", response="167fa3a0ab7669b23426620895a6c2ae",nonce="BroadWorksXiok8lubtTf9vnl8BW",cnonce="6DA143DD",qop=auth,algorithm=MD5,nc=00000001	
P-Asserted-Identity	**Pilot TN**	**Customer domain provisioned by Lumen**
Example:	9132756079	voip.centurylink.com
P-Asserted-Identity:	<sip:9132756079@voip.centurylink.com>	

TN = telephone number

911 example: Correct format – SIP invite to Lumen

INVITE: sip:911@voip.centurylink.com:5100 SIP/2.0
Via: SIP/2.0/UDP 63.157.48.130:5060;branch=z9hG4bK40368BD39C
From: <sip: 4785553710 @voip.centurylink.com>;tag=4247F498-864
To: <sip:911@voip.centurylink.com>
Date: Mon, 23 May 2016 16:38:29 GMT
Call-ID: 9D25A89A-203B11E6-B17ADA0D-8951F420@63.157.48.130
Supported: rel100,timer,resource-priority,replaces,histinfo,sdp-anat
Min-SE: 1800
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 102 INVITE
Timestamp: 1464021509
Contact: <sip:4785553710@ 63.157.48.130:5060 >
History-info: <sip:911@voip.centurylink.com:5100>;index=1, <sip:911@voip.centurylink.com:5100>;index=2
Expires: 180
Allow-Events: telephone-event
Authorization: Digest username=“ 257389-9132756079 ",realm="voip.centurylink.com",uri="sip:911@voip.centurylink.com:5100",response="167fa3a0ab7669b23426620895a6c2ae",nonce="BroadWorksXiok8lubtTf9vnl8BW",cnonce="6DA143DD",qop=auth,algorithm=MD5,nc=00000001
Max-Forwards: 68
P-Asserted-Identity: <sip:9132756079@voip.centurylink.com>
Session-Expires: 1800
Content Disposition: session;handling=required
Content-Type: application/sdpContent-Length: 300
Content Length: 285
Note: SDP was removed from this invite

SIP Trunking user TN

Customer domain

Source IP & port

Unique trunk username

911 example: Incorrect format – SIP invite to Lumen

INVITE: sip:911@voip.centurylink.com:5100 SIP/2.0
Via: SIP/2.0/UDP 63.157.48.130:5060;branch=z9hG4bK40368BD39C
From: <sip:4785553710@ 67.14.90.84 >;tag=4247F498-864
To: <sip:911@voip.centurylink.com>
Date: Mon, 23 May 2016 16:38:29 GMT
Call-ID: 9D25A89A-203B11E6-B17ADA0D-8951F420@63.157.48.130
Supported: rel100,timer,resource-priority,replaces,histinfo,sdp-anat
Min-SE: 1800
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 102 INVITE
Timestamp: 1464021509
Contact: <sip:4785553710@ 63.157.48.130:5060 >
History-info: <sip:911@voip.centurylink.com:5100>;index=1, <sip:911@voip.centurylink.com:5100>;index=2
Expires: 180
Allow-Events: telephone-event
Authorization: Digest username=“ 257389-9132756079 ",realm="voip.centurylink.com",uri=”sip:911@voip.centurylink.com:5100”,response="167fa3a0ab7669b 23426620895a6c2ae",nonce="BroadWorksXiok8lubtTf9vnl8BW",cnonce="6DA143DD",qop=auth,algorithm=MD5,nc=00000001
Max-Forwards: 68
P-Asserted-Identity: <sip:9132756079@voip.centurylink.com>
Session-Expires: 1800
Content-Type: application/sdpContent-Length: 300
Content Disposition: session;handling=required
Content Length: 285
Note: SDP was removed from this invite

The **from header** must match the format of the user information in the portal:
TN@CustomerDomain

In this example, that should be:
4785553710@voip.centurylink.com

This invite will fail with the **from header**:
sip:4785553710@67.14.90.84

Example: Group caller ID

The group name and group caller ID for the example shown on the previous two slides:

- TN/calling party: 478-272-3710
- Group name: Customer Example 0225 (a-18005)
- Group caller ID: 478-488-3751

Once the invite from SIP header is corrected, the group caller ID for each group will be sent to the 911 operator upon receiving a call. This will allow the 911 operator to receive the correct information and address.

Example: Cisco CUBE – Voice class SIP-profile

An example of a voice class SIP-profile CUBE configuration for modifying the host portion of the SIP URI in the **from header** to utilize the CenturyLink domain:

> Voice class sip-profiles 100 request REGISTER sip-header from modify "198.36.149.80" "voip.centurylink.com”