

# Setting Up Bridge Line Appearance Lines for Small and Medium Businesses

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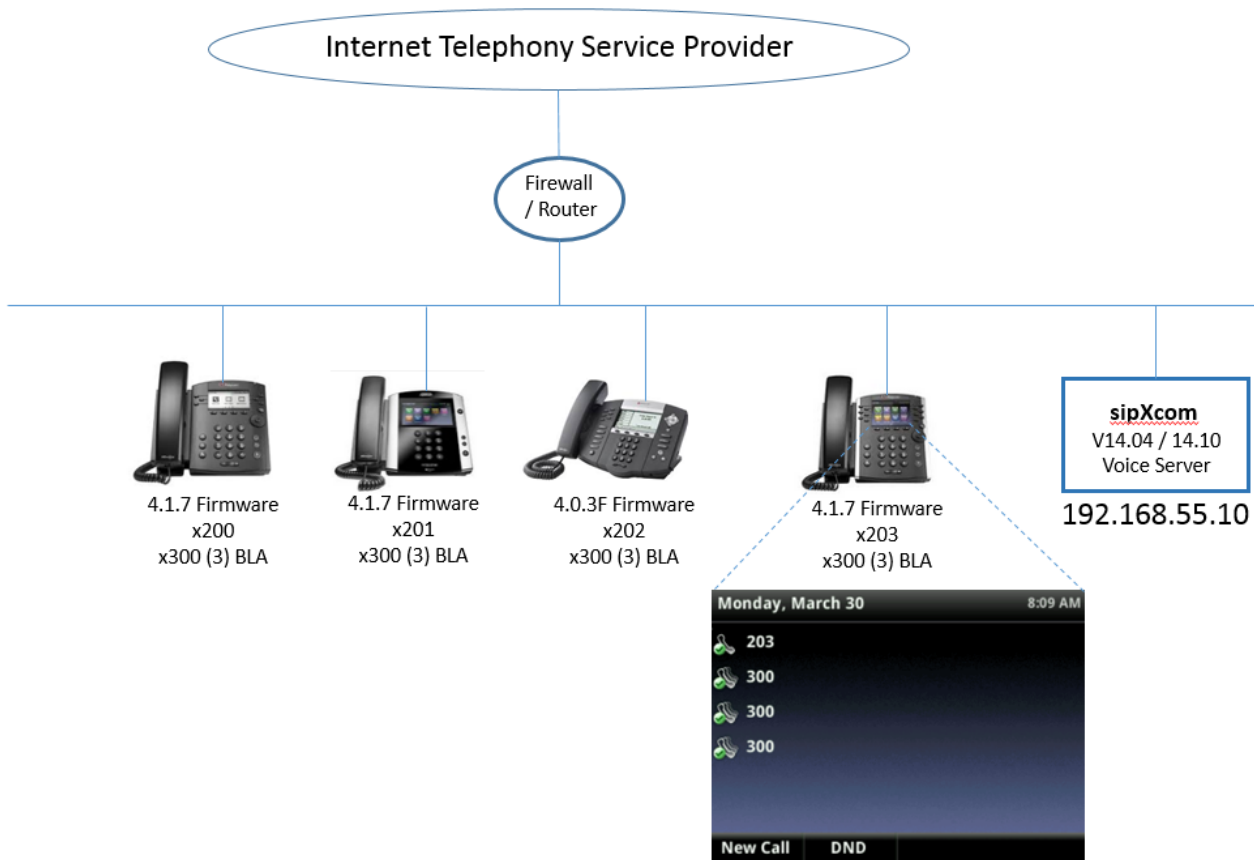
## Introduction

In the small and medium business market, legacy TDM-based PBXs are often configured with **shared call appearance** lines for incoming calls. Shared call appearance lines are lines that appear on multiple phones and deliver the following functionality:

- When an incoming call is mapped to a shared call appearance line, the lines ring simultaneously on multiple phones.
- When the call is answered by one phone, the status light on the answered phone goes green while the shared call appearance lines on the other phones goes red, signifying that the line has been answered by one phone.
- If the caller is placed on hold, all the shared call appearance lines flash red, indicating that the caller has been placed on hold.
- Any phone with shared call appearance line can take an 'on-hold' caller off of hold and make the call active again.
- When the call is completed and the line is placed 'on-hold', the status light for a shared call appearance line across all phones is turned off.

Small and medium businesses using shared call appearance on their existing TDM-based phone systems often request this capability when upgrading to a new phone system. Many medical, legal, and financial practices with one or more receptionists leverage the shared call appearance capability in their phone systems to answer calls centrally, and simplify the process of transferring calls from the reception area by placing the caller on hold and requesting another person to 'pick up the line on hold'.

With **sipXcom**, this capability is called **Bridged Line Appearance (BLA)** - the remainder of this document describes how to enable this feature with Polycom VVX Business Media and Polycom Soundpoint IP phones. sipXcom releases 14.04 and 14.10 have been used to validate BLA functionality in the following lab configuration:



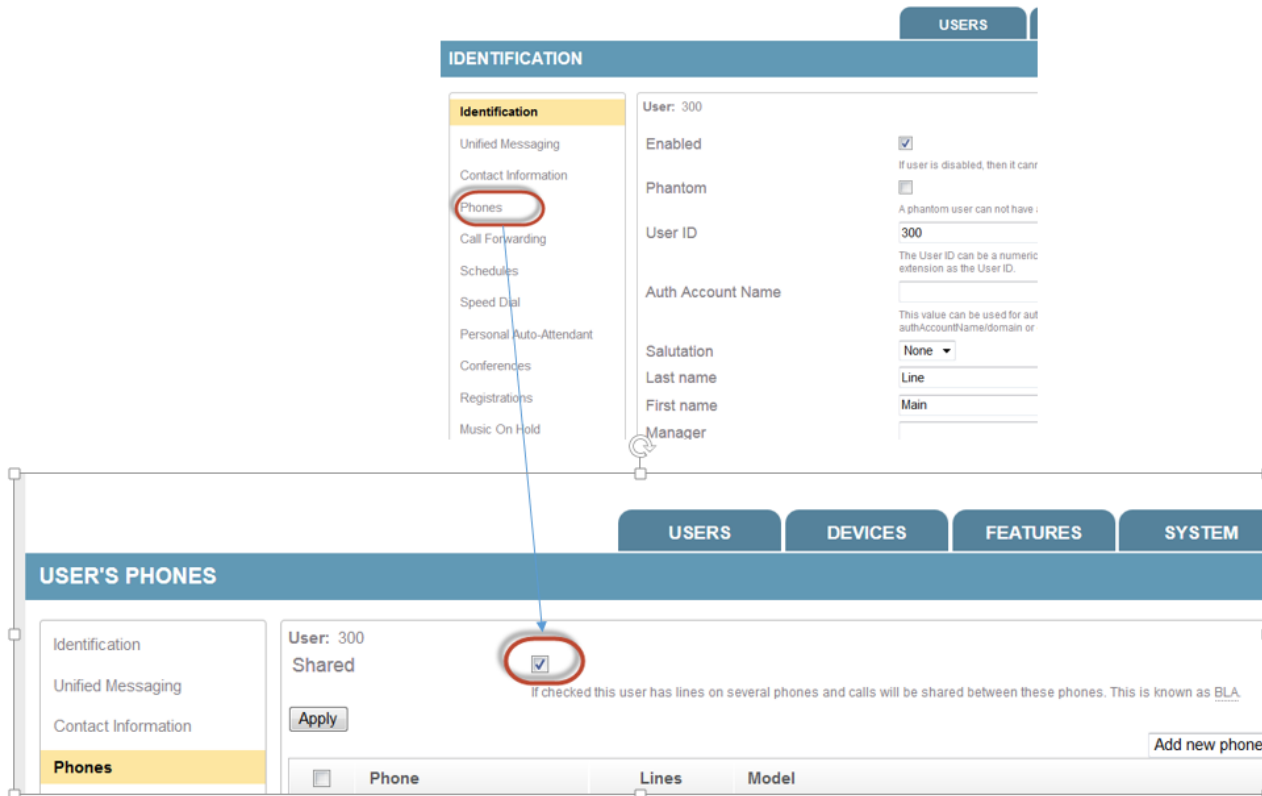
Each phone has three appearances of a BLA-enabled 'Main Line' x300 on their phones. Incoming calls to the main line will ring the first available main line as follows:

- If all main lines are available, then sipXcom will ring the first appearance of x300.
- If main line 1 is busy, the sipXcom will ring the second appearance of x300. If main lines 1 and 2 are busy, then sipXcom will ring the third appearance of x300.
- If main line 1 and / or main line 2 lines become available and a new incoming call arrives, sipXcom will ring the first available appearance of x300.
- If all main lines are busy and a fourth incoming call comes in, then sipXcom can be configured to either return a busy signal, go to voicemail, or trigger an auto-attendant.

⚠ The 'rollover' main line behavior only works properly when the same number of BLA-enabled lines appears on each phone. BLA lights will not appear in all cases when there is an asymmetrical number of BLA-enabled line appearances across the phones.

## Step 1 - Define x300 and Enable Bridged Line Appearance

After x300 has been defined, go to **Users->Phone** section and click on the **Shared** button - this enables the BLA capability on the line.



## Step 2 - Load the Polycom Firmware onto the sipXcom Voice Server

Load the Polycom Firmware and Updater onto the sipXcom voice server. Assign each device file to the correct version (4.0.x or 4.1.7). Map the phones to the correct firmware versions in the **Phones->Mac Address->Firmware Version** field - push the phone profiles.

⚠ Caution - BLA does not work properly with Polycom 4.1.8 and higher firmware loads on VVX phones and Polycom 4.0.6 and higher on Soundpoint IP phones. A 'ghost' line with red light appears on the phone answering the BLA-enabled line. This issue was resolved with Polycom 5.2.5 firmware on VVX Business Media Phones and verified with Sipxcom release 15.10. Polycom has not yet applied 'ghost line' fix for the Soundpoint IP phones - this issue continues to appear for every firmware release starting with Polycom 4.0.6 and higher.

Name	Active	Device Type
Polycm 4.0.3F	Active	Polycm SoundPoint IP/SoundStation IP/VVX
Polycm 4.1.7	Active	Polycm SoundPoint IP/SoundStation IP/VVX
Screen Capture	Active	Unmanaged (T)FTP Files

- Add files...
- Add files...
  - AudioCodes MP 11x
  - AudioCodes MP128
  - AudioCodes Mediant
  - AudioCodes TP260
  - AudioCodes HD IP Phone
  - Cisco 7940/60
  - Grandstream
  - LG Nortel IP Phone 68xx
  - Avaya 1210/1220/1230 IP Deskphone
  - Nortel IP Phone 1120/40
  - Nortel IP Phone 1535
  - Polycm SoundPoint IP/SoundStation IP/VVX
  - Unmanaged (T)FTP Files
  - Yealink Phones

**Phone Firmware Files**

Polycm phone firmware can be downloaded from the [Polycm support portal](#).

SIP Application (zip): UC\_Software\_4\_0\_3F\_release\_sig\_split.zip [Download](#) [Delete](#)

Version: 4.0.X

Each firmware version will be deployed in separate directories.

BootROM (zip): SoundPoint\_IP\_BootROM\_4\_4\_0B\_Upgrader\_release\_sig.zip [Download](#) [Delete](#)

The BootROM is updated less frequently than the SIP firmware. Upgrading BootROM is usually not a reversible operation.

**Phone Firmware Files**

Polycm phone firmware can be downloaded from the [Polycm support portal](#).

SIP Application (zip): Polycm\_UC\_Software\_4\_1\_7\_release\_sig\_split.zip [Download](#) [Delete](#)

Version: 4.1.7

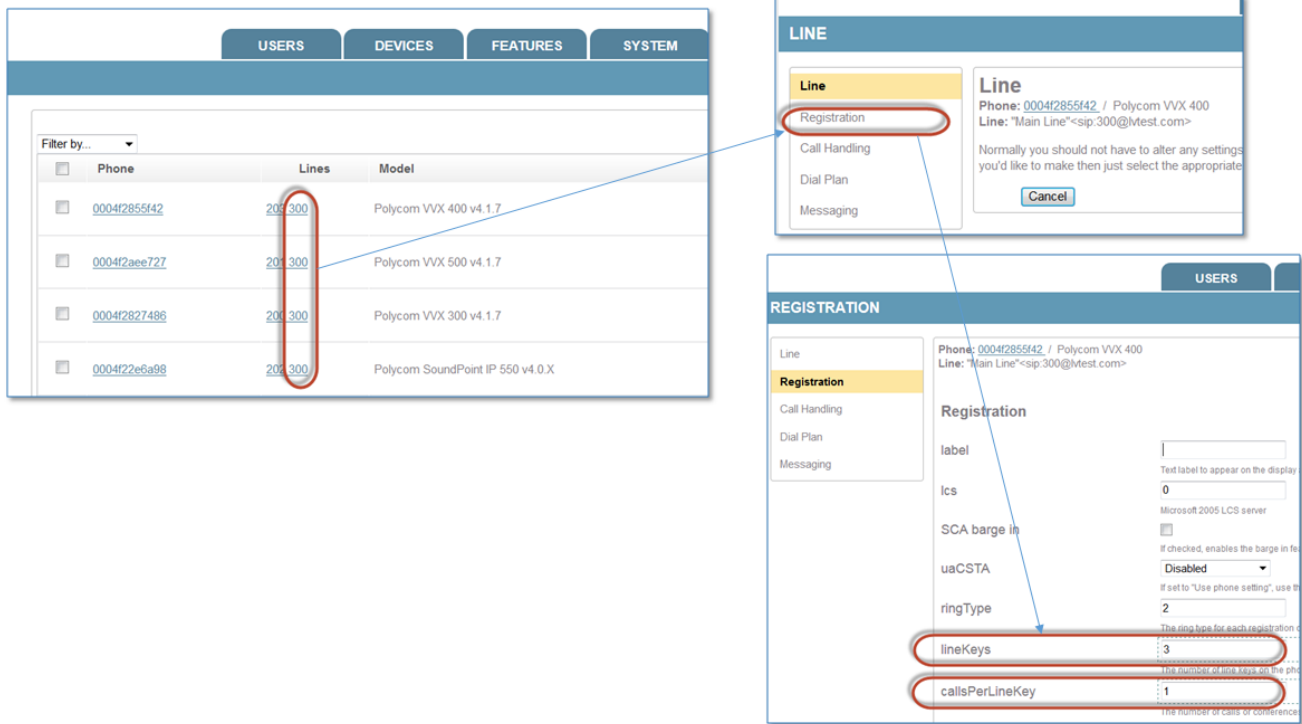
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BootROM (zip): SoundPoint\_IP\_BootROM\_4\_4\_0B\_Upgrader\_release\_sig.zip [Download](#) [Delete](#)

The BootROM is updated less frequently than the SIP firmware. Upgrading BootROM is usually not a reversible operation.

### Step 3 - Assign the Lines to the Phones

Assign the personal line as the first line on each phone, and the BLA-enabled main line as the second line. For the BLA-enabled line on each phone, go to the **Line->Registration** section, and change the number of line keys on each phone to 3 from 1, and the Calls Per Line Key from 24 to 1. This step is necessary for creating the roll-over effect on the main line.



## Step 4 - Between sipXcom 14.10 and 18.12, Provision SIP Domain Name in the Registration Outbound Proxy Field

From release 14.10 and 18.12, the SIP domain name of the sipXcom server is not automatically provisioned in the Outbound Proxy field in the **Phone->Line->Registration** menu - this has been corrected in release 19.04. Go to the **Phone->Line->Registration** menu for each phone where there is a shared line defined, and provision the **Phone->Line->Registration** menu field with the SIP Domain Name of the voice server.

**REGISTRATION**

Line: 0004f2ab8745 / Polycm VVX 500  
Line: "Main LV Line"<sip:300@hprod.com>

**Registration**

Auto Label Generation  (Default: unchecked)  
Defaults to false. By default, if unchecked and label field is empty, the user extension is automatically generated on the phone. When checked, the First Name/Last Name (if available and "Use Uri as Line Label" is ON and label field empty) will automatically be generated.

label

label Text label to appear on the display adjacent to the associated line key. If omitted, the label will be derived from the user part of 'Registration Server'

lcs  (Default: 0)  
Microsoft 2005 LCS server

SCA barge in  (Default: unchecked)  
If checked, enables the barge in feature for SCAs.

uaCSTA  (Default: Disabled)  
If set to "Use phone setting", use the phone's uaCSTA setting. If set to "Enabled" or "Disabled", override phone's uaCSTA setting for this line.

ringType  (Default: 2)  
The ring type for each registration can be configured. Note: ring type number 1 is silent ring.

lineKeys  (Default: 1)  
The number of line keys on the phone to be associated with registration. Only set to maximum number of lines. If not specified, value is 1

callsPerLineKey  (Default: 24)  
The number of calls per conference which may be active on a half-line key for a specific registration on the phone.

**Outbound Proxy**

Outbound Proxy Port  (Default: 0)  
Only required if both the following conditions are true. First, you want to direct all outgoing SIP traffic to a server other than the address you specified for the registration server. Second, you want the outbound proxy only for this line. All other lines will continue to send traffic to the registrar. NOTE: If you want to set the outbound proxy for all lines, goto to the phone settings for "Sip Server"

## Step 5 - Disable Alert-Info For External Calls

Go to the **System->Sipxproxy** menu and disable the **Enable for external calls** Alert-Info options. This capability is normally used to map calls to specific call waiting types and (beep, ring, and silent) - Sipxproxy inserts Alert-info values and class fields in the SIP header when this option is enabled that informs the phones what ring types are used. With BLA lines and the Enable for External Calls Alert-Info option disabled, Sipxproxy will check each appearance of a BLA-enabled line on the phone whether it is available to handle an incoming call. With the Alert-info option enabled, Sipxproxy will not check each appearance of a BLA-enabled line on the phone. With the **CallisperLineKey** option set to 1, Sipxproxy will send subsequent calls to voicemail or the AA when the first main line is active and **Alert-Info** options are enabled.

### Alert-Info

The Alert-Info header field can trigger alternate ringer sounds on many telephones, it is used to mark external phonecalls as such and trigger a different ringer sound if the phone is configured for it.

Enable for internal calls

(Default: unchecked)

Enable for external calls

(Default: checked)



## Step 6 - If Using sipXcom 14.10 and Higher with BLA, Change Firewall Settings for the SAA/BLA Services

This step is required when using sipXcom release 14.10 and higher with the BLA feature enabled. The issue has been pinpointed to default SIP server IP address not being passed to the phones. Go to the **System->Firewall->Rules** menu, and change the firewall settings for the **SAA/BLA** TCP and UDP services from **CLUSTER** to **PUBLIC**. Bridged Line Appearance services works for both standalone and high availability configurations. This issue has been corrected in sipXcom 18.12.

Regional Database	Disabled	CLUSTER	<input type="checkbox"/>
Regional Database Arbitrator	Disabled	CLUSTER	<input type="checkbox"/>
SAA/BLA TCP	10.20.2.31 : 5170	PUBLIC	<input type="checkbox"/>
SAA/BLA UDP	10.20.2.31 : 5170	PUBLIC	<input type="checkbox"/>
SIP Proxy - TCP	10.20.2.32 : 5060 10.20.2.31 : 5060 10.20.2.33 : 5060	PUBLIC	<input type="checkbox"/>
SIP Proxy - TLS	10.20.2.32 : 5061 10.20.2.31 : 5061 10.20.2.33 : 5061	PUBLIC	<input type="checkbox"/>
SIP Proxy - UDP	10.20.2.32 : 5060 10.20.2.31 : 5060 10.20.2.33 : 5060	PUBLIC	<input type="checkbox"/>
SIP Registry - TCP	10.20.2.32 : 5070 10.20.2.31 : 5070 10.20.2.33 : 5070	CLUSTER	<input type="checkbox"/>
SIP Registry - UDP	10.20.2.32 : 5070 10.20.2.31 : 5070	CLUSTER	<input type="checkbox"/>

## Step 7 - Suggested Options When Main Line Incoming Call is not Answered

When using BLA on the main line, a suggested best practice is to place the main in a hunt group - this provides the greatest flexibility for implementing three options.

1. Busy - no answer response
2. Send the call to x300 voicemail
3. Send the call to an auto-attendant

**Busy Signal**

Enabled:

Name: 914-417-2292

Extension: 600

DID Number: 9144172292

Description:

Call Sequence

Sequence	User	Aliases
Initially call	300	30

Use Voicemail:

Fallback Destination:

**X300 Voicemail**

Enabled:

Name: 914-417-2292

Extension: 600

DID Number: 9144172292

Description:

Call Sequence

Sequence	User	Aliases
Initially call	300	

Use Voicemail:

Allow Call Forwarding:

**Auto-Attendant**

Enabled:

Name: 914-417-2292

Extension: 600

DID Number: 9144172292

Description:

Call Sequence

Sequence	User	Aliases
Initially call	300	30

Use Voicemail:

Fallback Destination: 100