POLYCOM®

SoundPoint® IP Family Technical Bulletin – TB 16157

SoundPoint® IP audio issues experienced on handset and headset calls when used with certain types and versions of audio gateways.

This information applies to:

- All SoundPoint® IP phones used in situations that involve access to a PSTN system using certain SIP/PSTN audio gateways.

SYMPTOMS

Phone users will typically report the following reproducible[1] symptoms:

- The problem occurs only when making a VoIP/PSTN call (call through a PSTN gateway), but not when making a VoIP to VoIP call (between extensions within the enterprise).
- The problem occurs regardless of whether the SIP phone places the call or answers the call. IP phones running the MGCP application are not affected.
- The problem occurs regardless of the model of SoundPoint® IP desktop phone in use.
- The problem occurs when the handset or headset is used, but not when using hands-free (namely, the speakerphone).

The problem is experienced by the SoundPoint® IP phone user as "clipping" in the audio that they hear. Typical descriptions are:

- "The other side drops out when I talk."
- "The other side can't talk when I am talking."
- "The background noise changes when I talk."
- "There are strange low-level noises right after I stop talking."
- "The other side sounds choppy, broken up, rough."
- "The other side says that they can hear me perfectly."

[1] There is some evidence that the severity of the problem can vary from call to call or by time of day.

CAUSE

Certain audio gateway products are introducing audio distortion when used with some IP phones, including SoundPoint® IP products. This problem has been experienced with Cisco IOS® software 12.3 and 12.4 running on the Cisco AS5350, AS5400, and AS5850 universal gateways.

Although the SoundPoint® IP Handset Transmit signal is within the limits defined by the widely accepted ANSI/TIA/EIA-810 specification for narrowband VoIP transmission, it was found that certain characteristics of that Tx signal such as level and DC bias interact in a certain fashion with the gateway to give the user poor audio signal quality.

Detailed investigation of the issue by Polycom, Inc. has resulted in the implementation of a workaround, detailed in the next section.

WORKAROUND

To limit the effect of the issue, two steps need to be taken:

Step 1: ***Upgrade Software***.

Upgrade your SoundPoint IP phones to the latest SIP application. Refer to the VoIP SIP Software Release Matrix on the Polycom Customer Support web site.

Ensure that you are using the default configuration files, especially the default audio parameters.

Step 2: ***Reduce Tx Signal Level.***

The Tx signal level can be adjusted by a setting in the **sip.cfg** file:

> ***Implementation:*** In **sip.cfg**, reduce the value of voice.gain.tx.analog.handset by 6. As of SIP 3.0, voice.gain.tx.analog.handset = 6.
>
> **Note:** *For headset users, the gain may need to be similarly reduced utilizing the voice.gain.tx.analog.headset parameter in the* ***sip.cfg*** *file*.
>
> ***Side Effect:*** The loudness of the Tx signal will be reduced by 6dB and phone users may sound ‘quiet’ to the parties they are talking to on the far end.

STATUS

Polycom is working with partners to set up a technical interaction with the particular gateway manufacturers to establish a proper long-term resolution to the issue. This will allow SoundPoint® IP phones to be used with their default ANSI/TIA/EIA-810 compliant configuration with good audio quality.