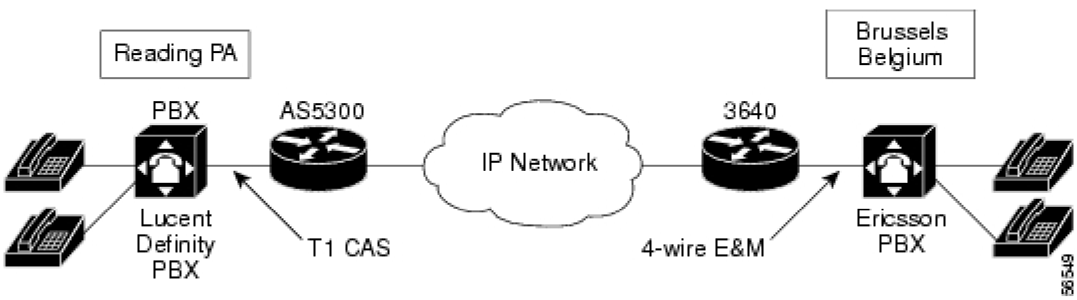
Echo Analysis Case Study

The following case study describes how Cisco worked with an enterprise customer to eliminate echo in a VoIP network. The customer is a large manufacturing company with headquarters in Reading, Pennsylvania, and several plants in the United States and overseas. One plant, located in Brussels, Belgium, previously used the PSTN for intersite calling, which resulted in high toll charges. Because the customer already had a data network in place, the logical choice was to implement a combined voice and data network. Because traffic at the headquarters was required to cross the Ethernet backbone to the PBX, the customer decided to use IP for voice traffic. It was calculated that the customer would save $3000 a month by installing three voice trunks across the data infrastructure.

Figure 1 shows the network topology between the headquarters and the remote site in Brussels.

Figure 1 Case Study Customer Topology



The Brussels site has 4-wire E&M trunks connected from an Ericsson PBX to the Cisco 3640 router. In Reading, a Cisco AS5300 access server is connected to a Lucent Definity GR3 PBX. All the proper QoS considerations and dial plan configurations were discussed and properly planned, and will not be discussed in this document.

Echo Problem Description

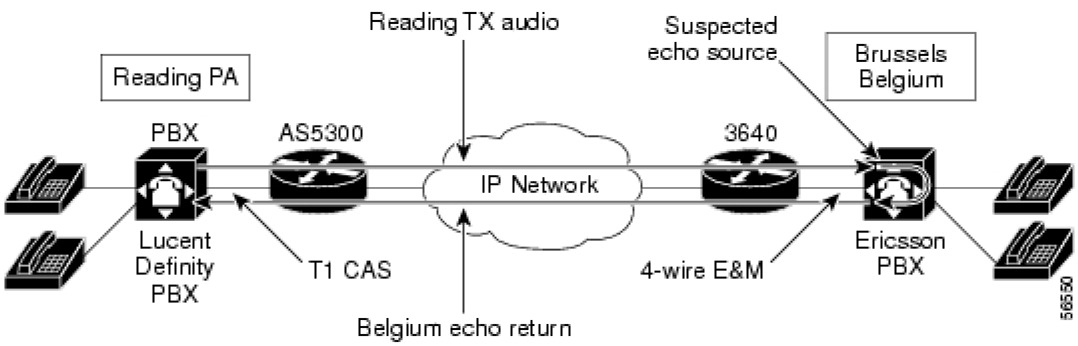
When the voice and data network was first implemented, users experienced substantial echos, so the customer reverted to using the PSTN for calls between the headquarters and the Brussels site. The customer initially believed that the Cisco routers were causing the echo, but we explained that our routers function like a 4-wire circuit and that it was not possible for leakage between the two voice paths to create echo.

After testing calls between headquarters and Brussels, we noticed large amounts of echo and determined that the echo was being heard only on the headquarters end of the calls; therefore, the source of the echo was in the Brussels tail circuit—between the Cisco 3640 router and the telephone in Brussels.

Initially, we thought this might be a case of loud echo, which means an echo caused by insufficient ERL in the tail circuit. We ruled out the possibility of a long echo—an echo delay longer than the echo canceler's coverage. The Cisco 3640 had echo cancelers active on the Brussels tail circuit, and the Brussels tail was connected only to the PBX. Long echo was not a possibility because the PBX would not cause a delay long enough to cause long echo. If calls from headquarters were dropping off the Brussels PBX or being routed to a third destination, long echo could then have been a possibility.

Eventually we discovered that a hybrid in the tail circuit was converting signals from 4-wire to 2-wire. Hybrids can be a common echo source. Figure 2 shows how the hybrid was deployed in the customer's network.

Figure 2 Echo in Customer Topology



We explained to the customer that the echo problem probably existed before VoIP was implemented but that it had not been perceivable because the PSTN delay was below the noticeable threshold. Packet-based networks create some small delays (as a result of packet encoding, queueing delays, and jitter buffers) that might unmask preexisting echo problems. This condition is normal and is characteristic of a packet-based network.

We set out to resolve the echo issue by proving that the problem was the PBX in Brussels and by proposing a solution to eliminate the echo problem. We looked at the following issues:

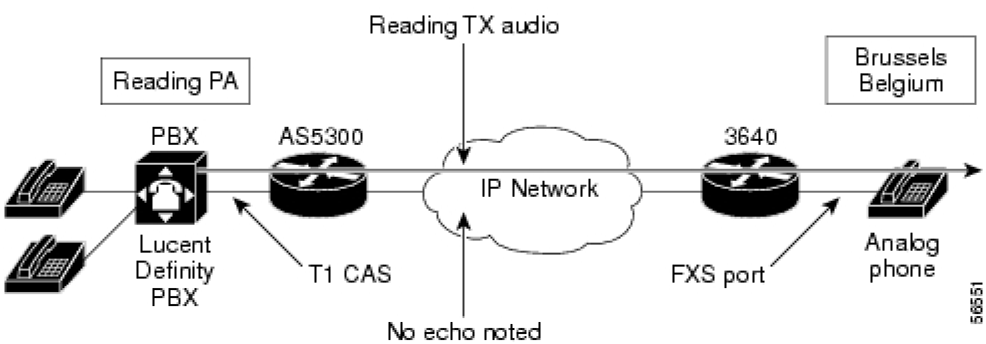
- Source of the echo
- Audio levels of the PBX
- ERL of the PBX
- Impedance settings

To thoroughly check the network, we ordered a commercial test set for the Brussels site. Before the test set was delivered, we ran a simpler preliminary test. We had an FXS module shipped to the customer site in Brussels from the local Cisco Technical

Assistance Center (TAC). We instructed the customer's on-site personnel to install and configure the FXS module in the existing Cisco 3640 to allow calls from the FXS port on the Brussels 3640 to the PBX in Reading. When we established calls between the Brussels 3640 and the PBX in Reading, there was no perceivable echo and the quality was very clear.

This test indicated that if the 4-wire to 2-wire conversion occurred on the router (as opposed to the Ericsson PBX), no echo was present. Therefore, the Ericsson PBX was most likely causing the echo. The simplest solution to such an echo problem would be to connect only FXS ports from the Cisco 3640 into the PBX. This configuration would allow the router to perform the 4-wire to 2-wire conversion, and the FXS ports would appear as central office (CO) trunks to the Ericsson PBX. Although this configuration would not provide as much flexibility as the 4-wire E&M trunks, it would not take away any functionality from the customers because they used an auto-attendant. Figure 3 shows this FXS test configuration.

Figure 3 FXS Test Configuration



Eliminating the Echo

After our test set arrived, we arranged to have a Cisco representative in Reading and an Ericsson representative on site in Brussels. The process to eliminate the echo was as follows:

1. Verify proper impedance levels on the Ericsson PBX in Brussels.

2. Verify proper audio levels.

3. Measure the ERL of the Ericsson PBX.

Verifying Proper Impedance Levels

The Ericsson representative verified that the impedance of the 4-wire E&M circuits was set for 600 ohms, which matched the configuration on the Cisco 3640.

Verifying Proper Audio Levels

We verified proper audio level settings from the Reading site to the Brussels site. The test set had the ability to connect to the Lucent PBX like any 2-wire analog phone; it also had a dial pad that allowed our test set to initiate a call to Brussels. After we established a call to Brussels, we injected a 1004-Hz tone at 0 dB into the Lucent PBX. We then measured the audio levels at various points along the voice path. These levels were verified in accordance with Cisco audio guidelines.

We entered a **show call active voice** EXEC command on the Reading router to verify the audio levels. The level on the Reading router measured -3 dB, which was the correct level according to the Cisco guidelines.

If the levels had needed to be adjusted, we would have entered the **input gain** voice-port configuration command. For example:

```
voice-port
  input gain 3
```

This command increases the level into the VoIP network by 3 dB. For these input gain changes to take effect, you need to hang up and reestablish the call.

After we verified the proper audio settings on the Reading router, we entered a **show call active voice** EXEC command on the Cisco 3640 in Brussels. This router displayed a -7 dB audio setting heading toward the Ericsson PBX. Even though the -7 dB level itself was acceptable, the optimal level is -12 dB at the phone on the PBX because different PBXs have different loss levels. Figure 4 and Table 1 depict the level adjustment configuration and the levels that were seen.

Figure 4 Audio Level and Echo Test Setup

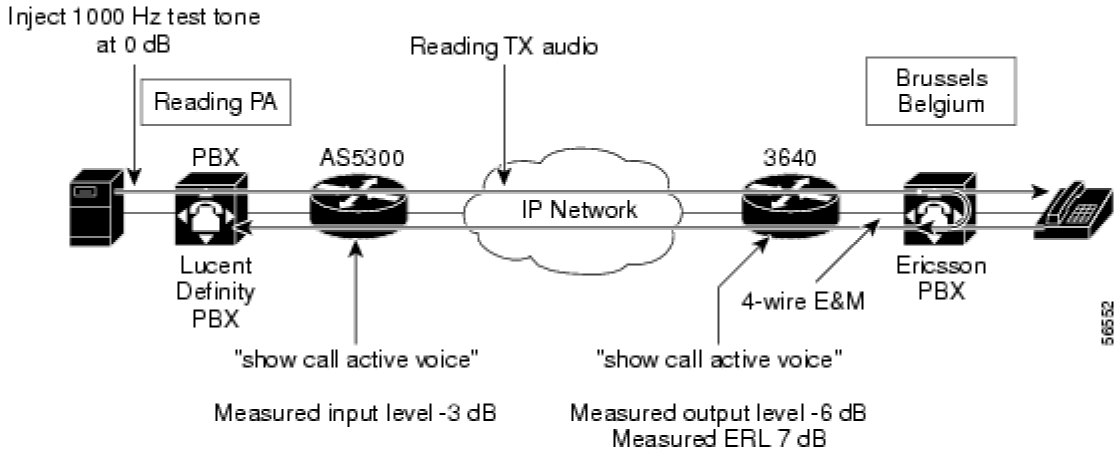


Table 1 show call active voice Command Output	
Reading AS5300	**Brussels 3640**

`Reading#` **`show call active voice`**
 	
`CoderTypeRate=g729r8`
`NoiseLevel=0`
`ACOMLevel=0`
`OutSignalLevel=-79`
`!This is the input level`
`InSignalLevel=-3`
	`Brussels#` **`show call active voice`**
	
	`CoderTypeRate=g729r8`
	`NoiseLevel=0`
	`ACOMLevel=0`
	`!This is the output level, R(out)`
	`OutSignalLevel=-7`
	`!This is the input level, S(in)`
	`InSignalLevel=-14`
	`InfoActivity=2`
	`ERLLevel=7`
	`!ERL = R(out) - S(in)`
	`!ERL = (-7) - (-14) = 7 dB`
	`!ERL should be > 15 dB`

Measuring ERL

Because the audio levels were acceptable to the customer, we did not adjust them. However, we did raise and lower the audio levels during the ERL test. We sourced a tone from Reading and measured the echo on the Cisco 3640 router in Brussels. (Note that you do not need an official test set for echo testing. You can use DTMF tones or your own voice to get an approximate indication of level mismatches.)

We applied the same 1004-Hz tone at 0 dB to the Reading PBX and then we again entered the **show call active voice** EXEC command to display the ERL level. The ERL represents the level of the echo coming out of the PBX in relation to the signal into the PBX. Notice that in Table 5 the ERL level is -14 dB, which means that, in relation to the signal going into the PBX, the echo is coming back at a level only *7 dB less* than what was going in.

The International Telecommunication Union Telecommunication Standardization Sector (ITU-T) recommendation G.131 states that the ERL of a PBX should be greater than 15 dB. The ERL was substantially higher than an echo canceler can effectively nullify; therefore, the echo problem was with the Brussels PBX. To further verify the problem location, we adjusted the audio level into the PBX up and down. When we adjusted the audio level, the ERL remained constant.

We ran the same test with the FXS port plugged into the Ericsson PBX, as shown in Figure 5. Note the measured ERL of 19 dB. Table 2 shows output from the **show caller active voice** EXEC command. That call exhibited no echo.

Figure 5 ERL Test Using the FXS Port in Brussels

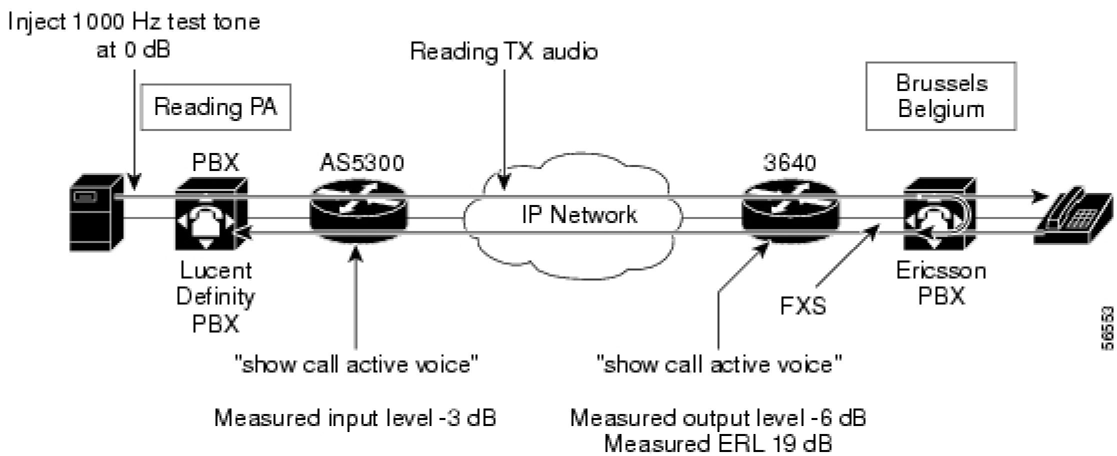


Table 2 show call active voice Command Output for FXS Test

Reading AS5300	Brussels 3640
Reading# **show call active voice** CoderTypeRate=g729r8 NoiseLevel=0 ACOMLevel=0 OutSignalLevel=-79 !This is the input level InSignalLevel=-3	Brussels# **show call active voice** CoderTypeRate=g729r8 NoiseLevel=0 ACOMLevel=0 !This is the output level, R(out) OutSignalLevel=-7 !This is the input level, S(in) InSignalLevel=-27 InfoActivity=2 ERLLevel=20 !ERL = R(out) - S(in) !ERL = (-7) - (-27) = 20 dB !ERL is > 15 dB

Case Study Summary

The customer was satisfied with our testing results and decided to use our suggested workaround of using FXS ports, which appeared as CO trunks to the Brussels PBX, out of the Brussels Cisco 3640 router. This solution reduced some of the inward dialing flexibility of the network, but because all inbound calls were handled by an auto-attendant, no functionality was lost.

This case study illustrates the importance of educating customers about proper expectations of packet-based networks. Specifically, you should stress that the normal characteristics of packet-based networks may unmask preexisting problems in the TDM-based voice infrastructures.

This particular kind of echo problem—where the echo is PBX-based—is the easiest to solve. It is much more difficult to solve a case where the tail circuit is the PSTN and calls to only some locations are being affected. Not only are such cases difficult to troubleshoot, but they also present the challenge of your convincing the customer that the problem is in the PSTN, not the VoIP network. In reality, this type of echo problem is not related just to VoIP. It is essentially media-independent, and can exist wherever added delays in the network might exist.

As a general recommendation, an end-to-end implementation of a fixed-loss plan should be undertaken first, and the appropriate values applied to the gateway interfaces. Then, the field measurements can be applied to the port configurations. This process is preferred instead of adjusting the levels based on the field measurements without the fixed-loss plan in place.

Related Documents

- *Cisco IOS Voice, Video and Fax Configuration Guide*, Release 12.2

- Shenoi, K. *Digital Signal Processing in Telecommunications*. Prentice Hall PTR; 1995

- *Voice over IP Fundamentals*, Cisco Press, 2000