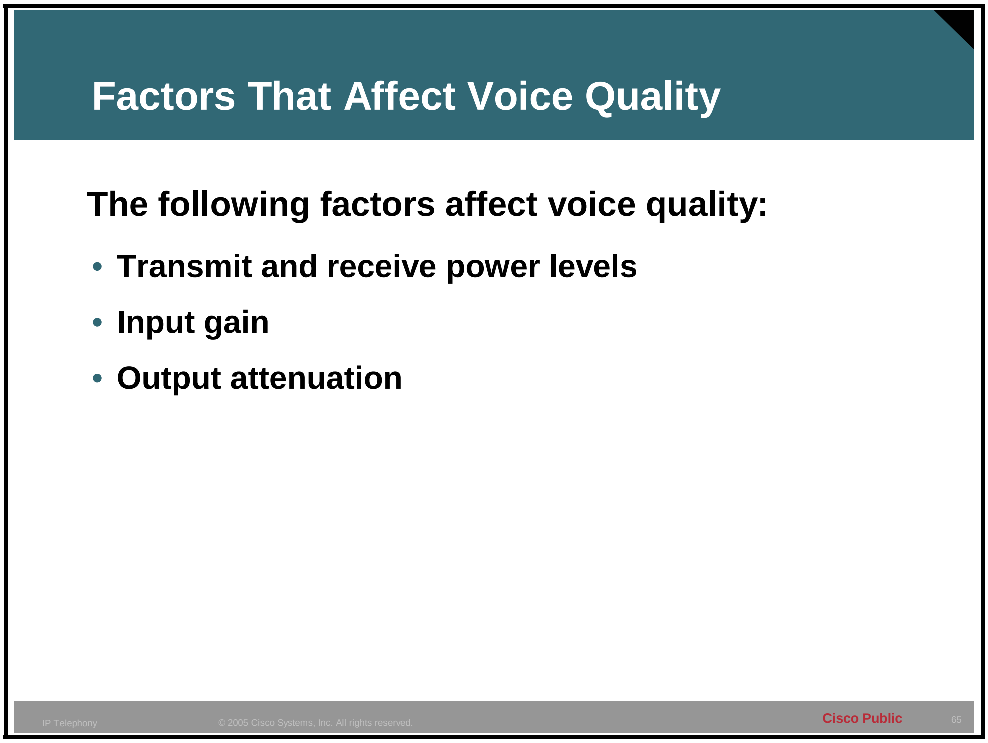
Adjusting Voice Quality

Electrical Characteristics

This topic describes the electrical characteristics of analog voice and the factors affecting voice quality.



Voice signal power in a long-distance connection must be tightly controlled. The delivered signal power must be high enough to be clearly understood, but not so strong that it leads to instabilities such as echo. In the traditional telephony network, telephone companies control the signal power levels at each analog device. Now that the IP network is carrying voice, it may be necessary to adjust signal power on a voice interface to fine-tune the voice quality.

Most initial voice signals enter the network through a two-wire local loop. Most switches connect to other switches through a four-wire connection. As voice travels through the network for delivery to the remote telephone, the voice signal must be passed from the two-wire local loop to the four-wire connection at the first switch, and from the four-wire connection at the switch to a two-wire local loop at the remote end. If the impedance at these two-wire to four-wire connections is not matched exactly, some of the voice signal reflects back in the direction of the source. As a result, originating callers hear their own voice reflected back. Sometimes, the reflected signal is reflected again, causing the destination to hear the same conversation twice.

In a traditional voice network, voice can reflect back; it usually goes unnoticed, however, because the delay is so low. In a VoIP network, echo is more noticeable because both packetization and compression contribute to delay.

Another problem is inconsistent volume at different points in the network. Both echo and volume inconsistency may be caused by a voice port that is generating a signal level that is too high or too low. You can adjust signal strength, either in the inbound direction from an edge telephone or switch into the voice port, or in the outbound direction from the voice port to the edge telephone or switch. Echo results from incorrect input or output levels, or from an impedance mismatch. Although these adjustments are available on the Cisco voice equipment, they are also adjustable on PBX equipment.

Too much input gain can cause clipped or fuzzy voice quality. If the output level is too high at the remote router voice port, the local caller hears echo. If the local router voice port input decibel level is too high, the remote side hears clipping. If the local router voice port input decibel level is too low, or the remote router output level is too low, the remote-side voice can become distorted at a very low volume and DTMF may be missed.

Calculating Decibel Levels

Change in signal strength is measured in decibels (dBs). You can either boost the signal or attenuate it by configuring the voice port for input gain or output attenuation. You must be aware of what a voice port connects to and know at what dB level that device works best.

Calculating network dB levels is often an exercise in simple number line arithmetic. The table provides common dB levels.

Calculating Decibel Levels

Source 1 Out/In	Router 1 Adjustment	Net at Router 1	WAN	Net at M Router 2	Router 2 Adjustment	Destination 1 In/Out
0 dB -->	–3 dB -->	–3 dB	—	–3 dB	± 6 dB -->	--> –9 dB
–9 dB <--	<--± 6 dB	–3 dB	—	–3 dB	–3 dB	<-- 0 dB

Baselining Input and Output Power Levels

Considerations for baselining input and output power levels are as follows:

- Analog voice routers operate best when the receive level from an analog source is set at approximately –3 dB.
- In the United States and most of Europe, the receive (transmit) level that is normally expected for an analog telephone is approximately –9 dB. In Asian and South American

countries, receive levels are closer to –14 dB. To accommodate these differences, the output levels to the router are set over a wide range.

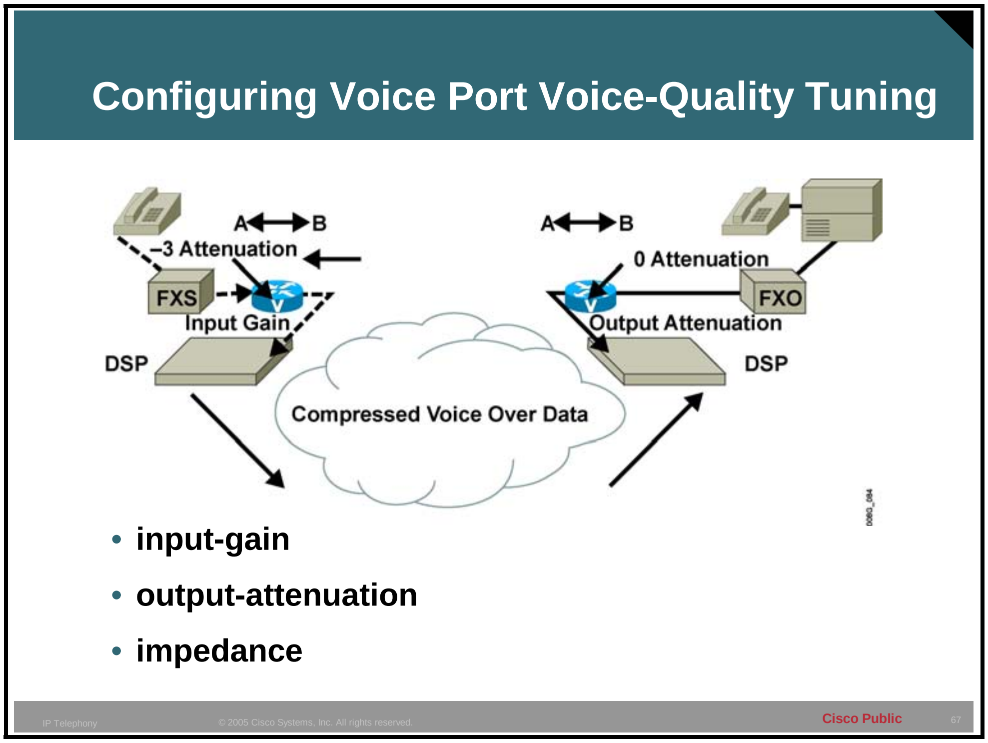
- Overdriving the circuit can cause analog clipping. Clipping occurs when the power level is above available pulse code modulation (PCM) codes, and a continuous repetition of the last PCM value is passed to the DSP.
- Echo occurs when impedance mismatches reflect power back to the source.

Example: Decibel Levels

Adjustment of decibel levels may be necessary throughout a voice network. A station connected to a PBX may experience one level of loudness when calling a local extension, a different level when dialing an outside line, and different levels when calling remote sites via VoIP. Adjustments may be necessary in this case.

Voice Quality Tuning

This topic describes voice-quality tuning configuration.



In an untuned network, a port configuration that delivers perceived good quality for a call between two dial peers might deliver perceived poor quality for a call between two other dial peers.

Voice quality adjustment is a defined, step-by-step procedure that is implemented after the network is up and running. It is ineffective for you to begin changing the default voice port configurations until full cross-network calls are established; a correctly implemented procedure results in a quality compromise between various sources that the customer accepts as good overall quality.

A variety of different factors, including input gain and output attenuation, can affect voice quality.

A loss plan looks at the required dB levels at specific interfaces, such as an analog FXS port connecting to a telephone, or an FXO port connecting to the PSTN. An analog voice router works best with a receive level of –3 dB. An analog telephone in North America and Europe works best with a receive level of –9 dB. Therefore, if the device connecting to that router provides a different level than the expected –3 dB, then input gain can be set to equalize it to –3 dB. If the output at the other end is a telephone that expects –9 dB, then the output voice port has to provide –6 dB output attenuation in addition to the –3 dB to send signaling to the telephone at the expected –9 dB levels. A systemwide loss plan looks at the dB levels of the initial input and the remote output ports and plans for the appropriate adjustments for end-to-end signal levels. You must consider other equipment (including PBXs) in the system when creating a loss plan.

Configuration Parameters

Parameters for configuring voice port voice-quality tuning are as follows:

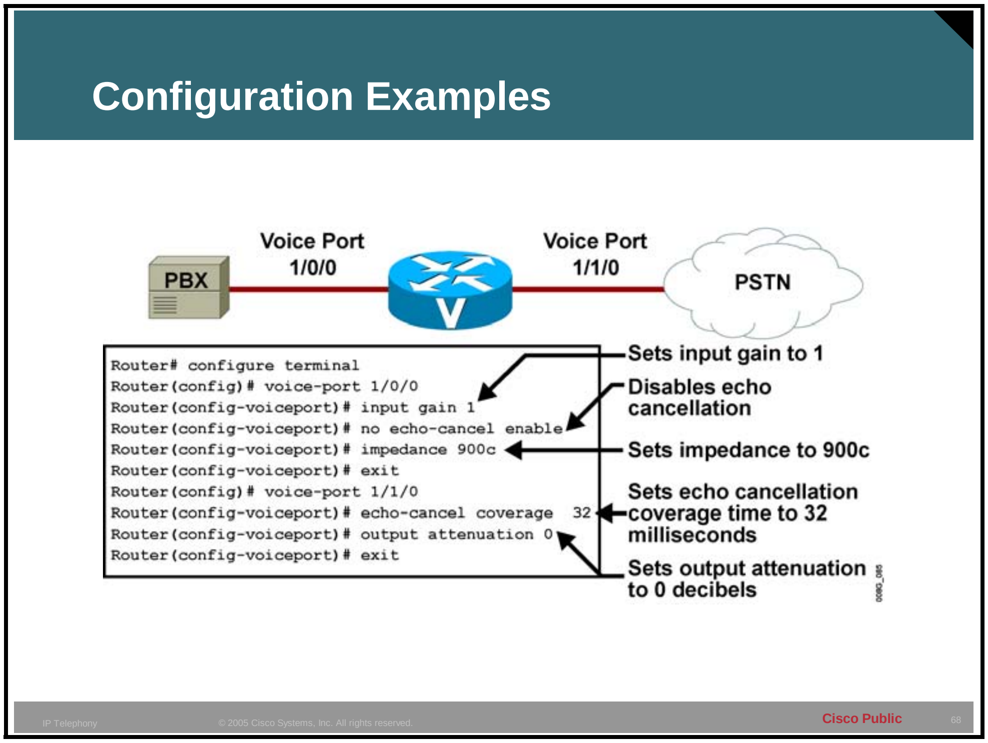
- **input-gain:** Configures a specific input gain, in decibels, to insert into the receiver side of the interface. The default value for this command assumes that a standard transmission loss plan is in effect, meaning that there must be an attenuation of –6 dB between telephones. The standard transmission plan defines country-specific dB levels and assumes that interfaces already provide the expected dB levels; for example, there must be attenuation of –6 dB between two telephones so that the input gain and output attenuation is 0, if the interfaces provide the required –6 dB attenuation.

  The gain of a signal to the PSTN can only be decreased. The gain of a signal coming into the router can be increased.
- **output-attenuation:** Configures the output attenuation value in decibels for the transmit side. The value represents the amount of loss to be inserted at the transmit side of the interface.
- **impedance:** Configures the terminating impedance of a voice port interface. The impedance value that is selected must match the setting from the specific telephony system to which it is connected. You must verify the impedance settings in the technical specifications document of the device. Impedance standards vary between countries. CO switches in the United States are predominantly 600 ohms real (600r). PBXs in the United States are normally 600r or 900 ohms complex (900c).

  Incorrect impedance settings or an impedance mismatch generates a significant amount of echo. You can mask the echo by enabling the **echo-cancel** command. In addition, gains often do not work correctly if there is an impedance mismatch.

Note The **input-gain** and **output-attenuation** commands accommodate network equipment and are not end-user volume controls for user comfort.

Example: Voice Port Tuning



This example shows voice port tuning parameters on the E&M and FXO ports of a Cisco voice-enabled router. In the example, the PBX output is –4 dB, whereas the voice router functions best at –3 dB. Therefore, the adjustment is made in the inbound path to the router using the **input-gain** command. The impedance setting on the router needs to be changed from the default of 600r to match the 900c impedance setting for the PBX. Because this is an E&M port, echo cancellation is disabled. The FXO port connecting to the PSTN has an adjustment for echo coverage that allows for longer-distance echo cancellation.

E&M voice port parameters include:

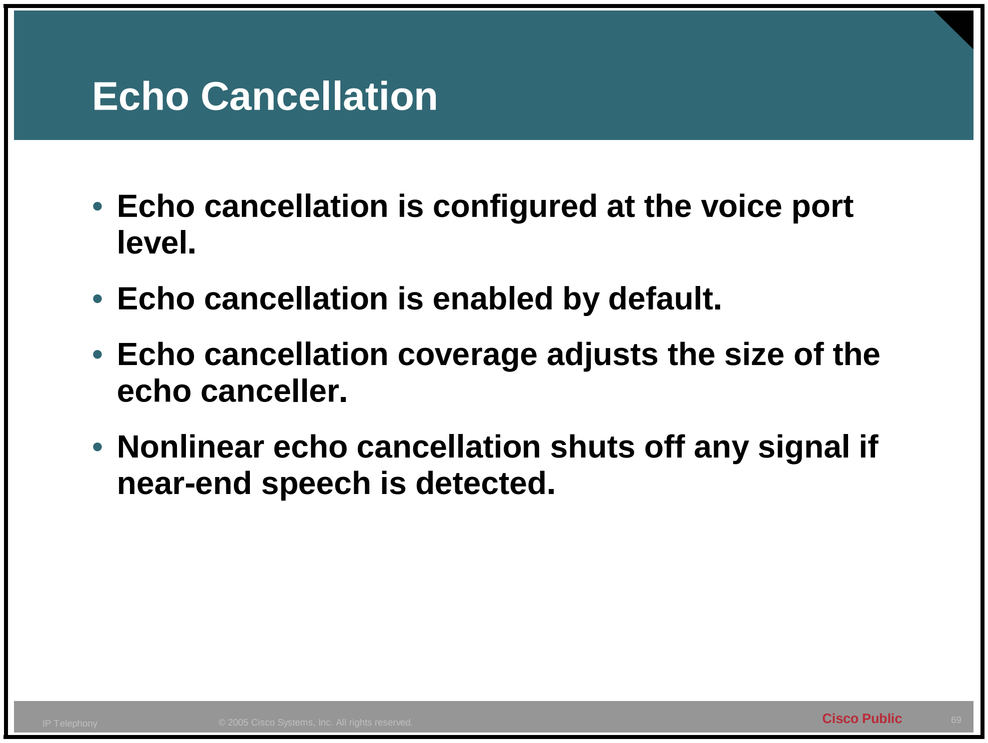
- **input-gain:** Increases the inbound voice level by 1 dB before the voice is transmitted across the network
- **no echo-cancel enable:** Disables echo cancellation
- **impedance:** Sets the impedance to match the connecting hardware

FXO voice port parameters include:

- **echo-cancel coverage:** Adjusts the cancellation coverage time to 32 ms. This allows for cancellation of echo that has greater delay.
- **output-attenuation:** Specifies that there is no attenuation as the signal is passed out of the interface to the PSTN.

Echo Cancellation Commands

This topic describes echo cancellation configuration parameters.



Echo cancellation is configured at the voice port level. It is enabled by default and its characteristics are configurable. Echo cancellation commands are as follows:

- **echo-cancel enable:** Enables cancellation of voice that is sent out through the interface and received back on the same interface. Sound that is received back in this manner is perceived by the listener as echo. Echo cancellation keeps a certain-sized sample of the outbound voice and calculates what that same signal looks like when it returns as an echo. Echo cancellation then attenuates the inbound signal by that amount to cancel the echo signal. If you disable echo cancellation, it will cause the remote side of a connection to hear echo. Because echo cancellation is an invasive process that can minimally degrade voice quality, you should disable this command if it is not needed. There is no echo path for a four-wire E&M interface. The echo canceller should be disabled for this interface type.

Note This command is valid only if the **echo-cancel coverage** command has been configured.

- **echo-cancel coverage:** Adjusts the coverage size of the echo canceller. This command enables cancellation of voice that is sent out through the interface and received back on the same interface within the configured amount of time. If the local loop (the distance from the interface to the connected equipment that is producing the echo) is longer, the configured value of this command should be extended.

  If you configure a longer value for this command, it takes the echo canceller longer to converge; in this case, the user may hear a slight echo when the connection is initially set up. If the configured value for this command is too short, the user may hear some echo for the duration of the call, because the echo canceller is not canceling the longer-delay echoes.

There is no echo or echo cancellation on the network side; for example, the non-POTS side of the connection.

Note This command is valid only if the echo cancel feature has been enabled.

- **non-linear:** The function enabled by the **non-linear** command is also known as residual echo suppression. This command effectively creates a half-duplex voice path. If voice is present on the inbound path, then there is no signal on the outbound path. This command is associated with the echo canceller operation. The **echo-cancel enable** command must be enabled for the **non-linear** command to take effect. Use the **non-linear** command to shut off any signal if near-end speech is not detected.

  Enabling the **non-linear** command normally improves performance; however, some users encounter truncation of consonants at the ends of sentences when this command is enabled. This occurs when one person is speaking and the other person starts to speak before the first person finishes. Because the nonlinear cancellation allows speech in one direction only, it must switch directions on the fly. This may clip the end of the sentence spoken by the first person or the beginning of the sentence spoken by the second person.

Caution Do not use the echo cancellation commands or adjust voice quality unless you are experienced in doing so. Arbitrarily adjusting these parameters could adversely affect voice quality.

ITU standard G.164 defines the performance of echo suppressors, which are the predecessors of echo cancellation technology. G.164 also defines the disabling of echo suppressors in the presence of 2100 Hz tones that precede low-bit-rate modems.

ITU standard G.165 defines echo cancellation and provides a number of objective tests that ensure a minimum level of performance. These tests check convergence speed of the echo canceller, stability of the echo canceller filter, performance of the non-linear processor, and a limited amount of double-talk testing. The signal used to perform these tests is white noise. Additionally, G.165 defines the disabling of echo cancellers in the presence of 2100 Hz signals with periodic phase reversals in order to support echo-canceling modem technology (for example, V.34), which do not work if line echo cancellation is performed in the connection.

ITU standard G.168 allows more rigorous testing and satisfies more testing requirements. White noise is replaced with a pseudo-speech signal for the convergence tests. Most echo cancellation algorithms use a least mean square algorithm to adapt the echo cancellation filter. This algorithm works best with random signals, and slows down with more correlated signals such as speech. Use of the pseudo-speech signal in testing provides a more realistic portrayal of the echo canceller's performance in real use.

Example: Echo Suppression Applied

If you speak into your telephone and hear your own voice a short time later, you are experiencing talker echo. Talker echo is caused by the remote telephony circuitry's two-wire to four-wire hybrid circuit. Enabling echo-cancellation on your voice port will eliminate the problem. Depending on the return time of the echoed voice, you can further adjust using the **echo-cancel coverage** command.

Echo Canceller Comparison

This table contains echo canceller comparison information.

Echo Canceller Comparison

	G.165 EC	G.168 EC
Tail Coverage	Up to 32 ms	Up to 64 ms
Minimum ERL	Greater than or equal to –6 dB	Configurable to greater than or equal to –0 dB, –3 dB, or –6 dB
Echo Suppression	Up to 10 seconds	Not required due to faster convergence
Minimum Cisco IOS Software Release	12.2(11)T, 12.2(8)T5, 12.2(12), and higher	12.2(13)T, 12.2(8)YN, 12.2(15)T, 12.3(4)T, 12.3(4)XD, and higher